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PROGRAM FOR NARROW-BAND ANALYSIS OF AIRCRAFT FLYOVER NOISE USING ENSEMBLE AVERAGING TECHNIQUES

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FIGURES AND TABLES

PROGRAM FOR NARROW-BAND ANALYSIS OF AIRCRAFT FLYOVER NOISE
USING ENSEMBLE AVERAGING TECHNIQUES

1.0 INTRODUCTION

Many studies, such as that by Chun, et.al. (Ref. 1), have been made to analyze jet engine noise. In general, these studies are conducted in a static mode, and predictions are made to reflect an actual flight situation. The result is usually expressed as a value of EPNdB (Effective Perceived Noise in decibels), where much of the detailed information is "lost" due to the requirements for computing EPNdB.

In the analysis package which follows, the intention is to convert flight data to equivalent static data using current prediction methods. Tones are clearly distinguishable from broadband noise since narrow-band analysis is employed. Eventually, such narrow-band analysis of flight data is expected to result in criteria for ground tests, which are easier and less costly to perform than flight tests.

The package presented is one which encompasses several problems associated with acoustical analysis of a moving source with respect to a stationary observer. The nonstationarity of the data causes difficulty in applying conventional time series analysis. Propagation effects influence all recorded data and must be accounted for. Also, the short integration time for each recording microphone requires some type of signal enhancement to increase accuracy of the data levels.

2.0 EXPERIMENTAL DESIGN

The development of this data reduction package was initiated to study jet engine flight noise where fan tones were radiating from the engine inlet (see Fig. 1). The aircraft's flight track was such that the noise source was flown directly over an array of microphones at constant altitude and velocity.

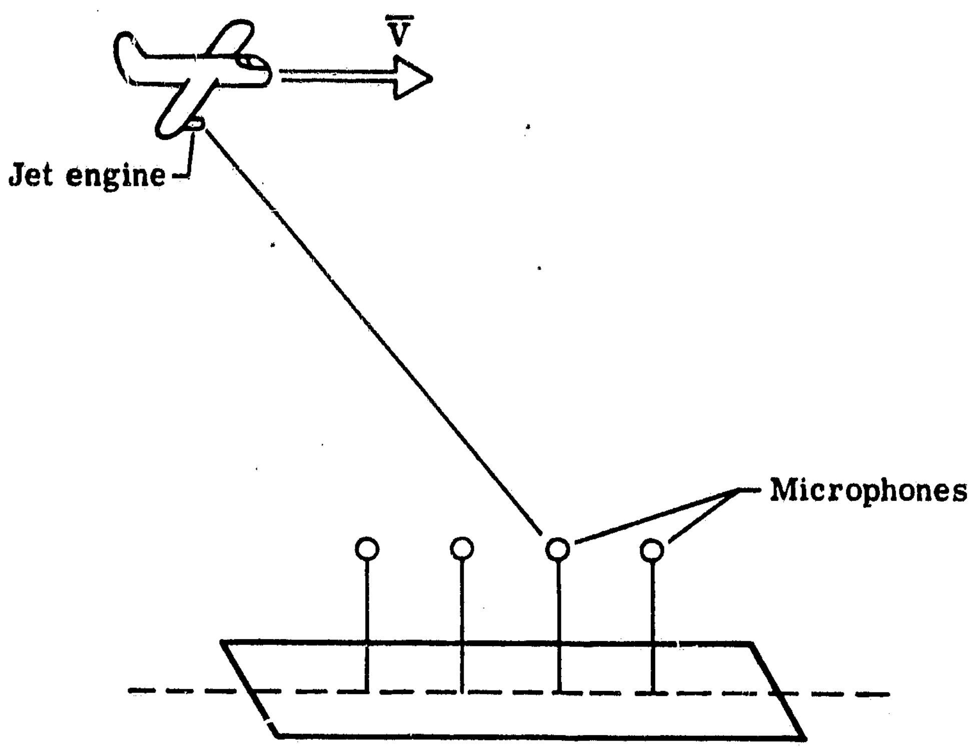

Figure 1. - General test design

The aircraft's position, in general, can vary from the intended flight path. Those variations were recorded by the use of a laser radar. The coordinates of the source (X', Y', Z') as shown in Figure 2, are defined by the displacements from an axis system (X, Y, and Z) whose origin is the first microphone in the array. These coordinates were obtained from a spherical to Cartesian transformation of the radar data followed by a translation of the origin to the first microphone.

Weather information was obtained with a specially implemented balloon stationed near the microphone array. Temperature, barometric pressure, relative humidity and wind speed were recorded at various altitudes from ground level to the altitude of the aircraft. All but wind speed are used in the propagation corrections.

3.0 STATIONARITY VERSUS NONSTATIONARITY

Many studies have been completed on the farfield noise levels of a stationary jet engine (i.e., via static testing) and the methodology for interpreting the noise characteristics are well understood. When the noise source is moving, however, the farfield noise is not as predictable as in the static mode since the forward speed effects (the motion effects) on the noise are not well understood (Ref. 1). In analyzing aircraft flyover data, it is desirable to obtain accurate spectra at various aircraft positions to determine noise levels generated by the source and to offer some understanding as to the forward speed effects.

A main area of concern in the analysis of flyover data is its nonstationarity. Data whose statistical properties vary with the passage of time (Ref. 2) are known as nonstationary. Frequency analysis, or time series analysis, was developed to handle random, stationary data. Hence, some difficulty is encountered in determining the frequency content of nonstationary data utilizing conventional data reduction techniques.

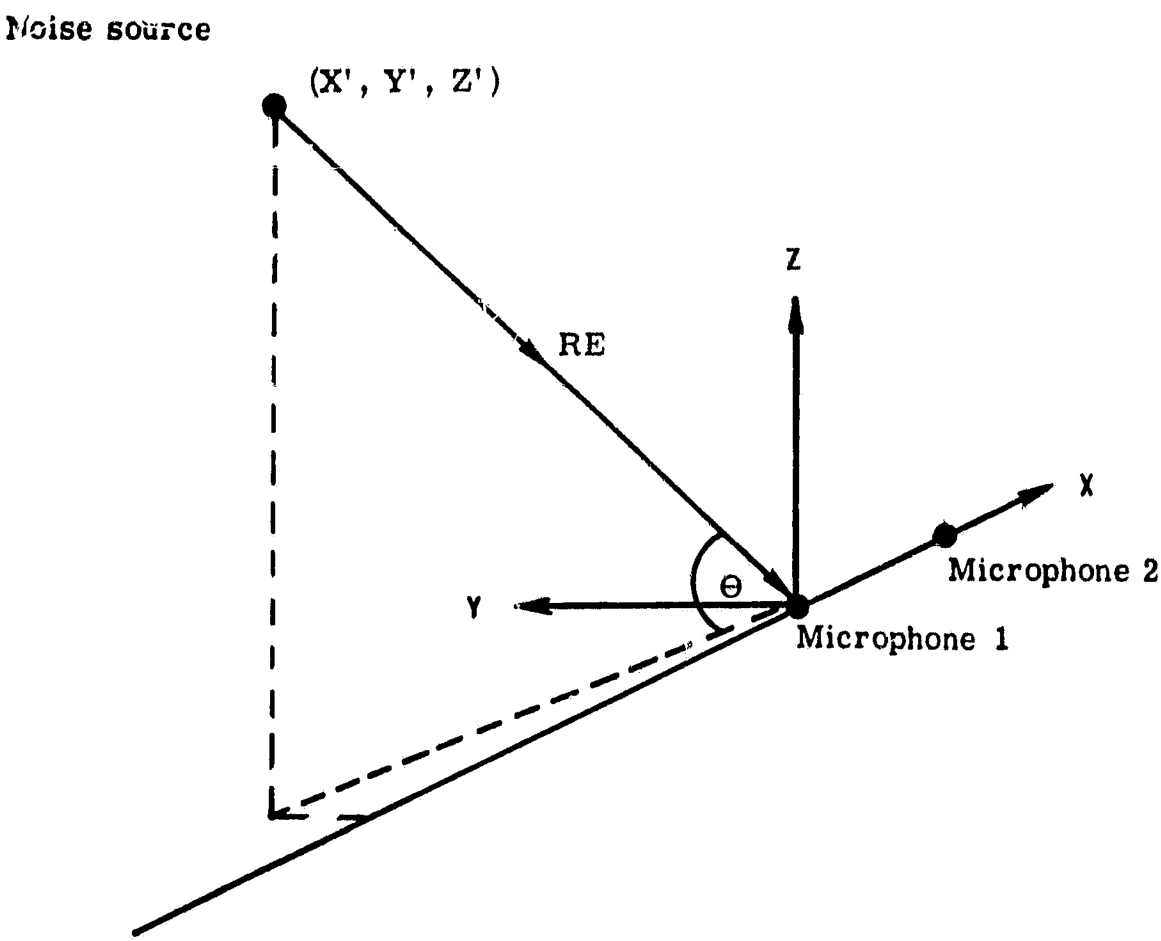

Figure 2. - Coordinate system

3.1 Implication of the Doppler Effect

The Doppler effect is of major concern as the apparent frequency changes throughout the movement of the source past the observer. For any microphone, the recorded frequency is equal to the actual generated frequency only when the aircraft is positioned directly over the microphone ($\theta=90°$). Thus, a method of analysis must be employed which assumes that a flyover noise data set meets the criteria for use of conventional time series analysis whose algorithms are generally based upon use of the Fast Fourier Transform (FFT).

The problem of nonstationarity in flyover analysis has been studied by such people as E.P. McDaid and L. Maestrello (Ref. 3) who found that if the directionality of the source is not taken into consideration, the effect of nonstationarity is negligible for most practical cases. In general, if the aircraft is considered to move discretely along its path from point to point, one may accept a small increment of time during which the data is relatively or locally stationary. According to J. S. Bendat and A. G. Piersol (Ref. 2) this assumption is acceptable providing the statistical properties within this increment do not change and hence time series analysis may be employed. Specifically, as has been shown with vibration data, e.g., vibration of a spacecraft during launch (Ref. 2), data may be considered locally stationary over those small increments if the data has normal and Chi-squared distributions.

3.2 Ensemble Averaging

Choosing a small increment of time over which to apply time series analysis inhibits the ability to ensemble average to obtain frequency content information. Yet, for most algorithms, it is desirable to average to obtain reasonable statistical accuracy. With a linear array of microphones placed along the flight path, each, theoretically, will record identical spectra at

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different recording times, barring any transient occurrences with the source production or any atmospheric disturbances. Referring to Figure 3, that time separation Δt_i is

$$\Delta t_i = d_i/\overline{V}$$

where d_i is the distance between each microphone and the reference microphone, and $\overline{V}$ is the velocity directly above the microphone array. Spectra at corresponding angles, θ, may be averaged to lead to a resulting ensemble averaged spectra. Thus, the total number of averages L for the resultant power spectra is the number M of FFT's averaged in the time series analysis times the number N of microphones averaged, or L=M·N. The final power spectral density PSD_R may be expressed as

$$PSD_R = \sum_{i=1}^{N} PSD_i$$

where PSD_i is the power spectra for microphone i calculated over M ensemble averages.

3.3 Determination of the Number of Averages

As was previously stated, one would like to acquire some specified level of accuracy for each spectral estimate. In general, the resulting power spectral density accuracy increases as the number of degrees of freedom ND increases. For PSD calcuation via the direct method (Ref. 4),

$$ND = 2L.$$

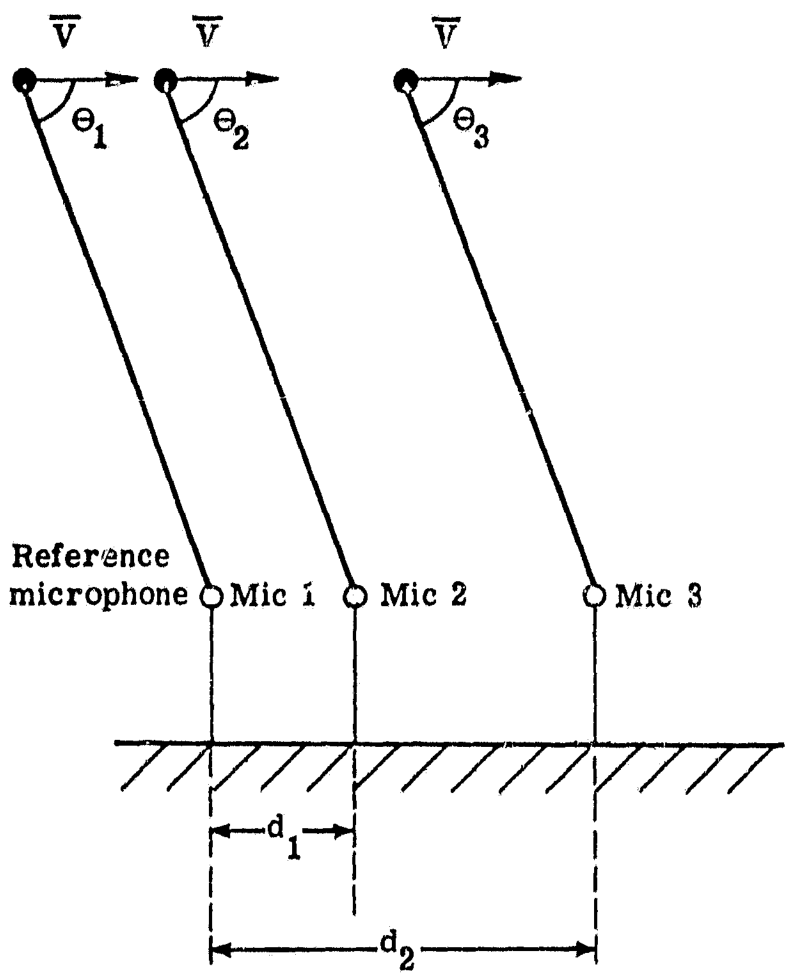

Figure 3. - Microphone position and directivity angle

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There are many ways to numerically define the error associated with power spectral density estimates generated by conventional time series analysis algorithms. (All those described below apply to PSD's calculated by the direct method (Ref. 4), not the Blackman-Tukey method.) The first expression for the error ε of estimation is

$$\varepsilon = \frac{1}{\sqrt{L}} = \sqrt{\frac{2}{ND}}$$

This states that the error ε decreases as ND increases. Once ND reaches 100, or the number of ensemble averages is 50, little accuracy is gained.

A second method results in defining confidence intervals based upon a Chi-squared distribution (Ref. 4). A percentage of confidence, or the percent probability that the measured mean square pressure spectrum accurately represents the true mean square pressure spectrum, may be chosen. Depending on the number of ensemble averages, a confidence interval at that probability is calculated. For example, at a 90% confidence level and 5 ensemble averages (10 degrees of freedom), the confidence interval is from -4.1 dB to 2.9 dB. To achieve accuracy within ±1 dB at 90% confidence, 40 averages would be needed (actual confidence interval is -1.2 dB to 1.0 dB).

Still another method was studied by K. Rao and J. Preisser (Ref. 6). The estimated and asymptotic variances were compared to determine the number of averages necessary to produce an adequate spectra. To achieve a reasonable normalized random error ε_r in percent,

$$L \approx (\varepsilon_r)^2 W$$

where W is the weighting factor for the data window applied to the time domain data when calculating the Discrete Fouri r Transform. ε_r is a measure of the convergence of mean spectra. If ε_r is 10, which corresponds to a 90% confidence, and the Hann data window is applied ($W = 3/8$),

$$L \approx (10^2) \cdot 3/8 \approx 40 \text{ averages}$$

From the error measurements presented above, a total of 40 averages appears to be sufficient to result in a satisfactory spectral representation for most flyover noise data. One must remember, however, that the data must be relatively stationary over the time interval corresponding to M averages or blocks. A block is the time segment over which the Fourier Transform is applied. Hence, a case where 5 Fourier Transforms (M=5) per microphone and 8 microphones (N=8) are averaged could be utilized if the stated criteria are met.

4.0 PROPAGATION EFFECTS AND BACKGROUND

To obtain an accurate spectral representation of noise data, it is important to account for all physical phenomenon present. In this section, a brief overview of the propagation effects and background is given. The methods employed in the flyover analysis package are discussed and the equations for their calculation are given.

4.1 The Doppler Effect

The Doppler effect, or the apparent change in frequency due to the relative motion of the source to the observer, is perhaps the most well known of the propagation effects. In the case of narrow-band flyover analysis, it cannot be ignored.

As the aircraft passes over a microphone at some average velocity $\overline{V}$, the observed frequency f_o is related to the actual source frequency f_s by

$$f_s = f_o (1 - M_c \cos\theta)$$

where M_c is the Mach number ($M_c = \overline{V}/c$ where c is the speed of sound in the medium through which the wave travels) and θ is the angle previously defined.

4.2 Convective Amplification

It has been well established that an acoustic signal is amplified due to the motion of the source. Various mathematical expressions for this effect exist depending on the type of source in motion.

A simple model which has been used up until very recently incorporates a small pulsating sphere represented by a convective monopole. A. Dowling (Ref. 7) states that this model is not accurate as motion introduces additional coupled monopoles whose effects lead to convective features previously not predicted. Convective amplification does depend upon the geometry of the source as well.

At this point, no representation which encompasses all that is discussed by Dowling exists. A frequently used expression relating the source pressure P_s to the observed pressure P_o is

$$P_o = \frac{P_s}{(1 - M_c \cos\theta)^{2n+2}}$$

where n indicates the type of noise source.

$$n = \begin{matrix} 0 & \text{Monopole} \\ 1 & \text{Dipole} \\ 2 & \text{Quadrapole} \end{matrix}$$

The expression $(1 - M_c \cos\theta)$ is the same as was defined in the Doppler effect. The difference in sound pressure level between static and flight cases ΔSPL_1 is

$$\Delta SPL_1 = 20(2n+2) \log_{10} (1 - M_c \cos\theta)$$

4.3 Inverse Square Law

The inverse square law describes the effect of the intensity of a signal falling off as $1/r^2$ where r is the radial distance from the source to the observer. In other words, if r_1 and r_2 correspond to two points on a ray emanating from a source, the respective acoustic pressures are related by

$$P_2^2 = P_1^2 \left(\frac{r_1}{r_2}\right)^2$$

The difference in sound pressure level ΔSPL_2 may then be expressed as

$$\Delta SPL_2 = -20(\log_{10} (\frac{r_1}{r_2}))$$

4.4 Atmospheric Absorption

Sound absorption in still air leads to an attenuation of the wave as it passes through the atmospheric medium. Atmospheric absorption has been studied quite extensively. For example, C. M. Harris (Ref. 8) defined the coefficient of absorption α under controlled conditions for various values of relative humidity, temperature, and frequency; M. Greenspan (Ref. 9) studied the rotational relaxation of nitrogen, oxygen, and air; K. S. Chun, et.al. (Ref. 1) offer a simplified calculation of α. All are attempts to quantize the total atmospheric absorption into thermal and viscous effects (called classical absorption) and rotational and vibrational relaxation effects. Vibrational relaxation is primarily due to both nitrogen and oxygen relaxation.

The absorption coefficient is a composite of classical absorption α_{CL}, rotational relaxation α_{rot}, and vibrational relaxation of nitrogen and oxygen, $\alpha_{vib,N}$ and $\alpha_{vib,O}$ respectively, or

$$\alpha = \alpha_{CL} + \alpha_{rot} + \alpha_{vib,N} + \alpha_{vib,O}$$

F. D. Shields and H. F. Bass (Ref. 10) have combined these coefficients to provide a thorough method of calculating the absorption coefficient in terms of dB/meter which can easily be applied to sound pressure level data. The following is an outline of their development.

Given the barometric pressure P, temperature T, and relative humidity RH, at any frequency f, the following procedure may be employed.

1) Calculate the partial pressure of saturated water vapor in N/m^2 by

$$\log_{10}(P_{sat}/P_o) = 10.79586\,[1 - (T_{01}/T)]$$

$$-5.02808\,\log_{10}(T/T_{01}) + 1.50474 \times 10^{-4}(1-10^{-8.29692[(T/T_{01})-1]})$$

$$+0.42873 \times 10^{-3}(10^{4\ 76955[1-(T_{01}/T)]}-1)\ -2.2195983$$

where P_o = reference pressure of 1.013×10^5 N m^2

T_{01} = 273.16°K

2) Calculate the absolute humidity H in %

$$H = RH\,(P_{sat}/P_o)/(P/P_o)$$

3) Calculate the relaxation frequency of oxygen and nitrogen, $f_{r,0}$ and $f_{r,N}$ by

$$f_{r,0} = (P/P_o\ [24 + 4.41 \times 10^{+4}\ H[(0.05 + H)/(0.391 + H)]]$$

$$f_{r,N} = (P/P_o)(T/T_o)^{-1/2}[9 + 350H\ \exp\ (-6.142[(T/T_o)^{-1/3}-1])]$$

where T_o = reference temperature of 293.15°K.

4) Calculate the absorption coefficient a(f) in dB/m

$$a(f) = 8.686(T/T_o)^{1/2}[f^2/(P/P_o)] \times$$

$$(1.84 \times 10^{-11} + 2.19 \times 10^{-4}(T/T_o)^{-1}(P/P_o)(2239/T)^2 \times$$

$$[\exp\ (-2239/T)]/[f_{r,0} + (f^2/f_{r,0})]$$

$$+ 8.16 \times 10^{-4}(T/T_0)^{-1}(P/P_0)(3352/T)^2 \times$$

$$[\exp(-3352/T)]/[f_{r,N} + (f^2/f_{r,N})])$$

Once the atmospheric absorption coefficient is calculated, it may be applied to atmospheric layers, each with corresponding P, T, and RH value (see Fig. 4). Over N layers (ℓ) the attenuation of the sound wave in terms of dB, ΔSPL_3, is

$$\Delta SPL_3 = \sum_{i=1}^{N} a_i(f) \cdot r_i$$

at each frequency f, where $r_i = h/\sin\theta$.

4.5 Ground Impedance

The levels recorded by a microphone above the ground include energy which has been reflected by the surface. This additional intensity must be subtracted from the observed sound pressure level values to obtain a free field value.

Consider Figure 5 in which both a reflected wave and a direct wave from the source are recorded by the microphone. As explained by Pao et.al. (Ref. 11), the ground factor or ratio of the free field mean square pressure to the mean square pressure with ground effect, when the surface is considered to be acoustically hard is

$$G = 2 + 2^{-(+ak\Delta r)^2} \cos(k\Delta r)$$

where k is the wave number $(\frac{2\pi f}{c})$, a = 0.01, and Δr is the difference between the reflected path and the direct path. When the noise is averaged over finite frequency bandwidths,

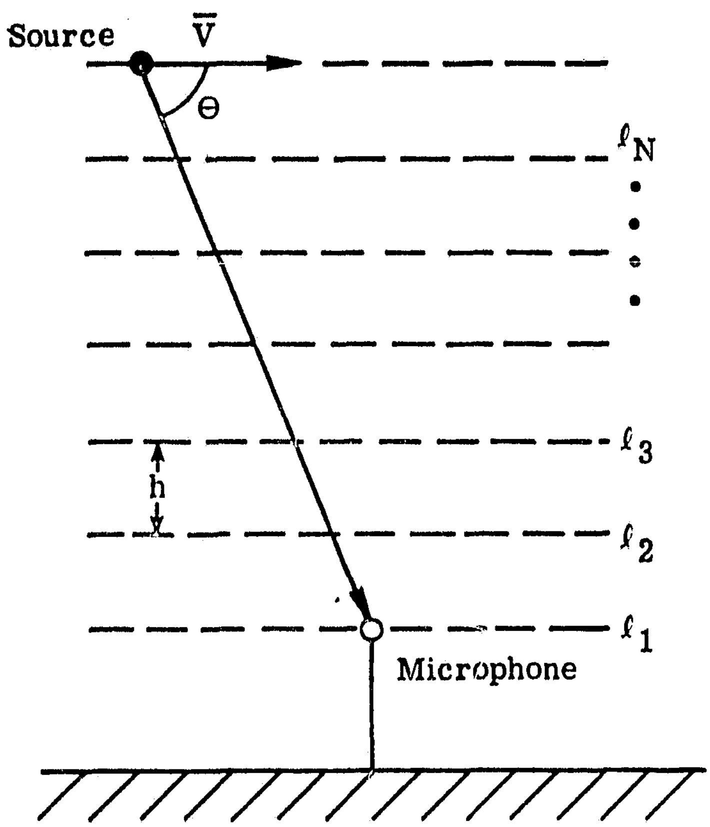
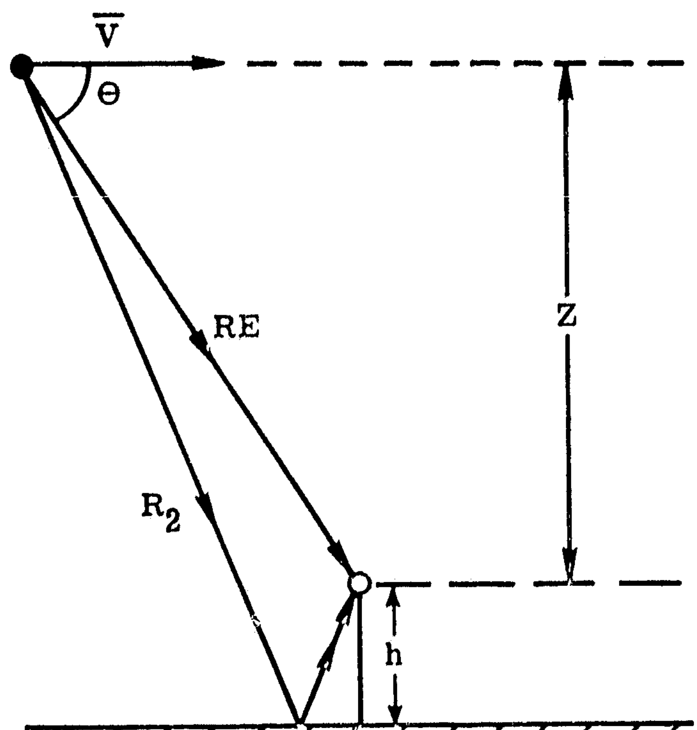


Figure 4. - Atmospheric absorption layered model



$$\Delta\Gamma \approx \sqrt{RE \circ RE + 4Zh} - RE$$

if $h \ll Z$ and $h \ll RE$

Figure 5. - Ground impedance model

$$G = 2 + 2\, e^{-(ak\Delta r)^2} \cos(k_c r) \frac{\sin(\Delta k \Delta r)}{\Delta k \Delta r}$$

where $\Delta k = \frac{2\pi}{c} \cdot \frac{\Delta f}{2}$ (where Δf is the bandwidth of interest), and k_c is $\frac{2\pi f_c}{c}$ (where f_c is the central frequency about the band.). The resulting difference in sound pressure level ΔSPL_4 is

$$\Delta SPL_4 = -10 \log \left(2 + 2\, e^{-(ak\Delta r)^2} \cos(k_c \Delta r) \frac{\sin(\Delta k \Delta r)}{\Delta k \Delta r}\right)$$

4.6 Background Subtraction

It may be desirable to subtract background noise from a calculated sound pressure level. This is easily accomplished by comparing a background spectrum at some value of θ to a data spectrum at the same angle. The spectral values must be compared at each value of frequency.

In general, to subtract sound pressure level values at some frequency f_k,

$$SPL_{fkc} = SPL_{fkm} + 10 \log_{10} \left(1 - 10^{\frac{(SPL_{fkB} - SPL_{fkm})}{10}}\right)$$

where SPL_{fkc} is the background subtracted resultant SPL, SPL_{fkm} is the measured or data SPL value, and SPL_{fkB} is the background SPL value.

For a signal to noise ratio $\varepsilon(f_k)$, where $\varepsilon(f_k) = SPL_{fkm} - SPL_{fkB}$, greater than 10 dB, a negligible correction is required as the background level is considerably smaller than the level of interest. A signal to noise ratio less than or equal to 3 dB implies that the background and data levels are very close, i.e., background only. If the data consistently shows a small signal to noise ratio throughout all frequencies, some question may be raised as to the validity of either the background spectra or the data spectra.

5.0 FLYOVER ANALYSIS DATA REDUCTION PROCEDURES

As was mentioned earlier, for flyover analysis narrow-band spectra and directivities are the desired output. The directivity is often used as a characteristic measurement of a jet engine, and sound pressure levels are used for more extensive analysis. The procedures employed to yield these results are six-fold.

5.1 Analog-to-Digital to Engineering Units Tape

The first procedure results in an engineering units tape, for example, a tape whose data channel units are N/m^2, which places the information in a readable form for the performance of all succeeding functions (see Fig. 6). The first step involves analog-to-digital conversion by available transcription methods. Care must be taken in this step. Before digitizing, consideration must be given to the maximum frequency of interest and the frequency resolution (bandwidth) to be desired for spectral analysis. Once the maximum frequency of interest is known, it is standard practice to low-pass filter at that frequency to avoid aliasing, or folding, in the time series analysis results. For most hardware systems, a rate of digitization, or sample rate SR, is required to be 2.5 times the selected cutoff frequency or greater to avoid biasing when filtering the data. All time series analyses can yield results up to 1/2 • SR which is known as the Nyquist frequency.

In conjunction with the selection of the maximum frequency of interest and hence the sample rate, a frequency resolution must be chosen. The bandwidth BW is

$$BW = \frac{SR}{2 \cdot NPTS}$$

where NPTS is the number of output points from a time series analysis program. In the direct method of power spectral density computation

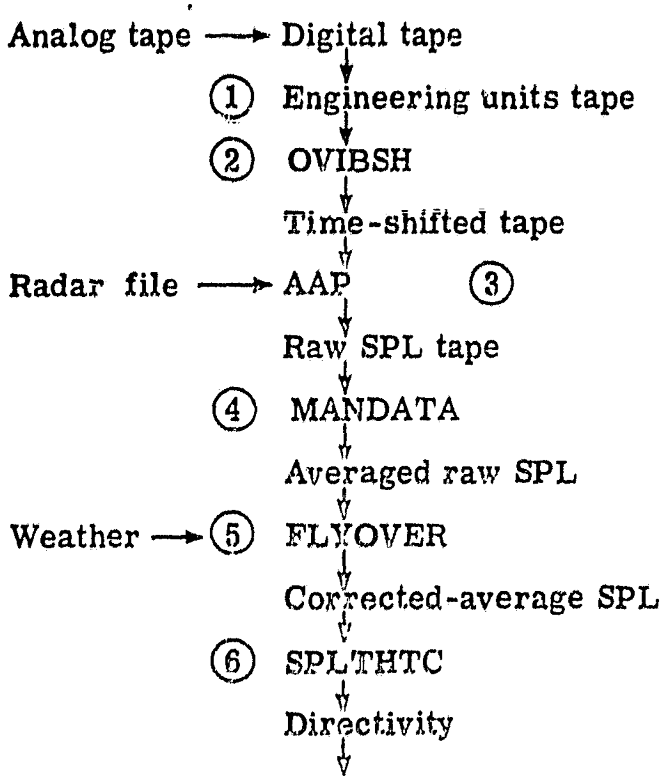


Figure 6. - Data reduction flowchart

NPTS is one-half the number of points over which the Fourier Transform is applied, i.e., 1/2 the number of points chosen per block. One must remember that over the time interval corresponding to M blocks, the data should be relatively stationary. In other words, the time increment of assumed local stationarity t_r is

$$t_r = (2 \cdot M \cdot NPTS)/SR$$

Hence, maximum frequency, bandwidth, and time of local stationarity must all be examined prior to the selection of SR.

The second step in generating the engineering units tape is to apply the proper gains and sensitivities to each recorded data channel. The sensitivities are found by recording and digitizing known calibration signals through each data channel which yields a linear relationship between counts and engineering units. Gains are tabulated for each microphone and for each flyover which is made. Once these are applied to the digitized data, flyover noise analysis may begin.

5.2 Time Shifting

To be able to average microphone sound pressure levels, each microphone must be shifted in time to appear to be located at the same reference position. This is accomplished by the program called OV1BSH (Appendix A) by matrix manipulation (Step 2 of Figure 6). The velocity $\overline{V}$ is extracted from radar information and used to calculate the number of points each microphone is to be shifted, $NSHIFT_i$, where

$$NSHIFT_i = \frac{SR \cdot d_i}{\overline{V}}$$

and d_i is the distance from microphone i to the reference microphone or reference position.

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5.3 Raw Sound Pressure Level

It is at this point, Step 3 of Figure 6, that the determination of the noise levels begins. A time series analysis program called the Acoustics Analysis Program (Ref. 5 and Appendix B) is utilized to determine the raw sound pressure levels, or the sound pressure levels of time-shifted engineering units data for each microphone and each selected value of θ. Averaged and corrected SPL's and the directivities are calculated from the raw SPL's.

The Acoustics Analysis Program employs the direct method of computation for the power spectral densities, however, that is not a requirement of the flyover analysis package. Many time series analysis programs exist which utilize the Blackman-Tukey method, i.e., the power spectral density is calculated from the data set's autocorrelation. Some differences between the methods (Ref. 12) should be considered to generate comparable output.

5.4 Averaged SPL

To obtain corrected sound pressure levels, it is first necessary to average the raw sound pressure levels of the geometrically similar microphones at each selected angle as discussed in Section 3. This average must be accomplished by dealing with units of (pressure)^2, P^2, or power. More specifically, the average power $P_a^2(f)$ at frequency f is

$$P_a^2(f) = \sum_{i=1}^{N} \left(\frac{REF^2 * 10^{\frac{SPL_i(f)}{10}}}{N} \right)$$

where N is the number of microphones to be averaged and REF is a reference for dB conversion, ($REF = 2 \times 10^{-5} N/m^2$). This average power may be converted back to an average SPL value, $SPL_a(f)$ by

$$SPL_a(f) = 10 \log_{10}\left(\frac{P_a^2(f)}{REF^2}\right)$$

This is done by program MANDATA (Appendix C) which is Step 4 of Figure 6.

5.5 Corrected Average SPL

Step 5 of Figure 6 is the application of program FLYOVER (Appendix D) to the averaged sound pressure levels. It corrects the spectra for instrumentation effects and propagation effects. The output is then a realistic picture of the source generated noise levels, if background is considered to be negligible, i.e., $\varepsilon(f_k) \geq 10$ dB for all values of f_k.

The first correction to be applied is that of instrumentation. It is composed of 1) pressure response, 2) diffraction, and 3) windscreen corrections. In general, these corrections are frequency dependent and are to be added to the observed sound pressure level. They are functions of the type of microphone and the angles of acoustic incidence.

The propagation effects are applied in the following order:

1) Convective Amplification
2) Inverse Square Law
3) Atmospheric Absorption
4) Ground Impedance
5) Doppler Frequency Shift

The order of application is not important with the exception of the Doppler effect. Some of the earlier corrections, i.e., atmospheric absorption and ground impedance, are frequency dependent and utilize the observed frequency for calculation.

5.6 Directivity

The final procedure in Figure 6, Step 6 involves the calculation of the directivity. Program SPLTHTC (Appendix E) determines peak value at a selected frequency band $f \pm \Delta f$ where f is the frequency of interest and Δf is a factor which allows for small variations in f from one angle to another as implementation of the Doppler shift does not yield identical values of frequency for each θ.

Program SPLTHTC also has the capability to subtract the background directivity by the method discussed in Section 4.6. This results in a true representation of the source's directivity.

The final option to SPLTHTC is to sum the two largest values within the band $f \pm \Delta f$. This is necessary to account for a spreading of the peak value to two frequency values. The phenomenon is caused by reflection and the fact that the Fourier transform is applied to a discrete interval.

6.0 APPLICATION

Up to this point, no sample data have been presented. It is the intent of this section to aid in understanding the data reduction techniques by presenting an example of flyover analysis (Ref. 13).

6.1 Design

For this test, a monotone source of 4000 Hertz was mounted on the wing of an aircraft. The source was flown over a microphone array consisting of 10 microphones located 30 feet apart and placed 30 feet above the runway (see Fig. 7). Atmospheric data were recorded by a weather balloon during each flight. The aircraft's position was recorded by radar and, in general, was

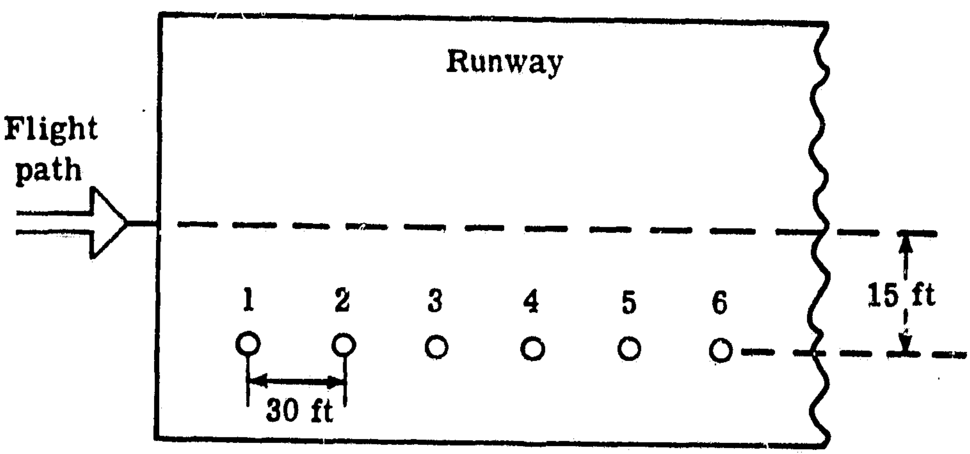

Figure 7. - Microphone array for experimental design

300 feet above the runway at a velocity of 200 ft/second. The analog tape which recorded the pressures of all ten microphones was digitized at 50000 samples per second. This sample rate was chosen to meet the criteria presented in Section 5.1. The maximum frequency of interest was 20000 Hertz and a bandwidth of 100 Hertz or less was requested. To be able to average 8 microphones and 5 transforms for a random error of 10 (see Section 3.3), a block size (the number of points over which to compute the Fast Fourier transform) of 512 points was chosen. This results in 256 output points, and,

$$BW = \frac{50000}{2(256)} = 97.656 \text{ Hertz}$$

and

$$t_r = \frac{2 \cdot 5 \cdot 256}{50000} = 0.0512 \text{ sec.}$$

which corresponds to an aircraft displacement of 10.24 feet along its flight path. This is a relatively small distance compared to the total recorded X displacement which is approximately 4000 ft. Therefore, it is considered a discrete increment for time series analysis purposes.

6.2 Results

Given the information above, the microphones must be shifted by a number of points equal to

$$NSHIFT_i = \frac{(i-1)(30 \text{ ft}) \cdot 50000/\text{sec}}{200 \text{ ft/sec}} = 7500 \ (i-1) \text{ pts.}$$

where i is the microphone number and microphone 1 is the reference microphone. (The shifting procedure results in 67,500 fewer digital points and should be taken into consideration when determining the time interval for digitization.).

At this point, raw sound pressure levels are generated by the Acoustics Analysis Program. Figures 8 (A through H) show raw sound pressure levels of microphones 1 through 8 at an angle of 80°. Note that radar information has been incorporated and is displayed on the plots. Raw sound pressure levels of this type were generated for all microphones over 5 blocks of 512 points at angles of 20° through 110° at 5° increments.

The next step is to average the microphones at each selected angle. Figure 9 is the averaged sound pressure level resulting from the spectra seen in Figure 8. The averaging process is done in terms of pressure squared. Note the averaged spectra's relative smoothness when compared to the spectra seen in Figure 8, which is due to the employment of ensemble averaging.

Program FLYOVER (Step 5 of Figure 6) is now applied to each averaged sound pressure level. Input required includes the radar information present on each average spectra file, and instrumentation corrections, and weather data. Samples of the latter two can be seen in Tables 1 and 2. The averaged and corrected sound pressure level results and an example can be seen in Figure 10. This is the same data shown in Figures 8 and 9. Note that the strongest signal occurs at 4000 Hertz which is the frequency generated by the source.

Once the averaged and corrected sound pressure levels for all selected angles have been computed, the directivity may be determined. Choosing 4000 Hertz as a frequency of interest, Figure 11 results. A value of 200 Hertz was given as the band over which to determine the sum of the two highest values in program SPLTHTC. The summing is done as the peak is spread over two frequency values (see Figure 10). Note that the directivity of the monotone source is a constant which is the expected result.

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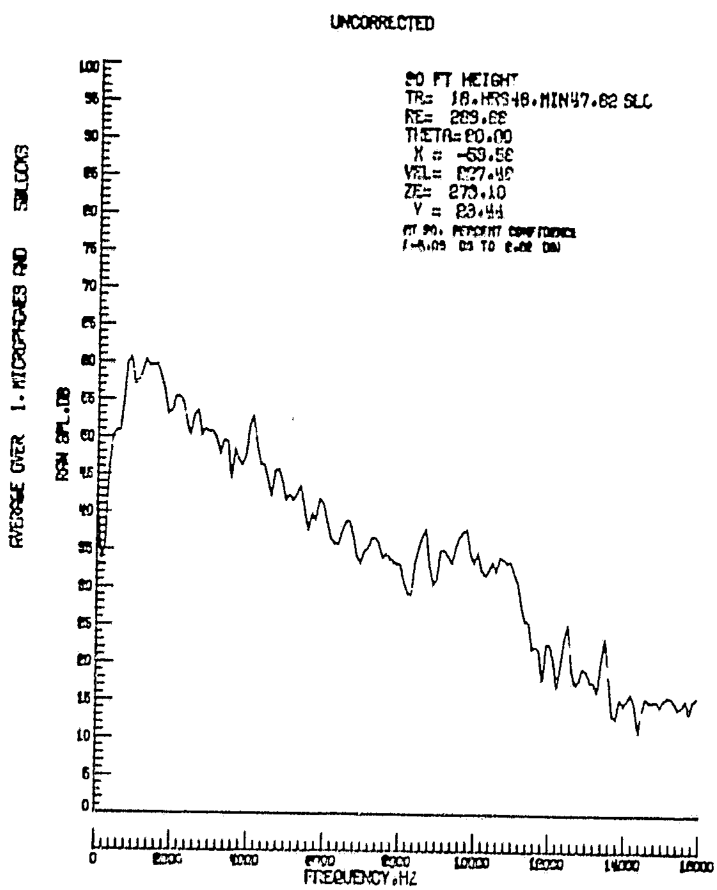

(a) Microphone 1

Figure 8. - Raw sound pressure levels

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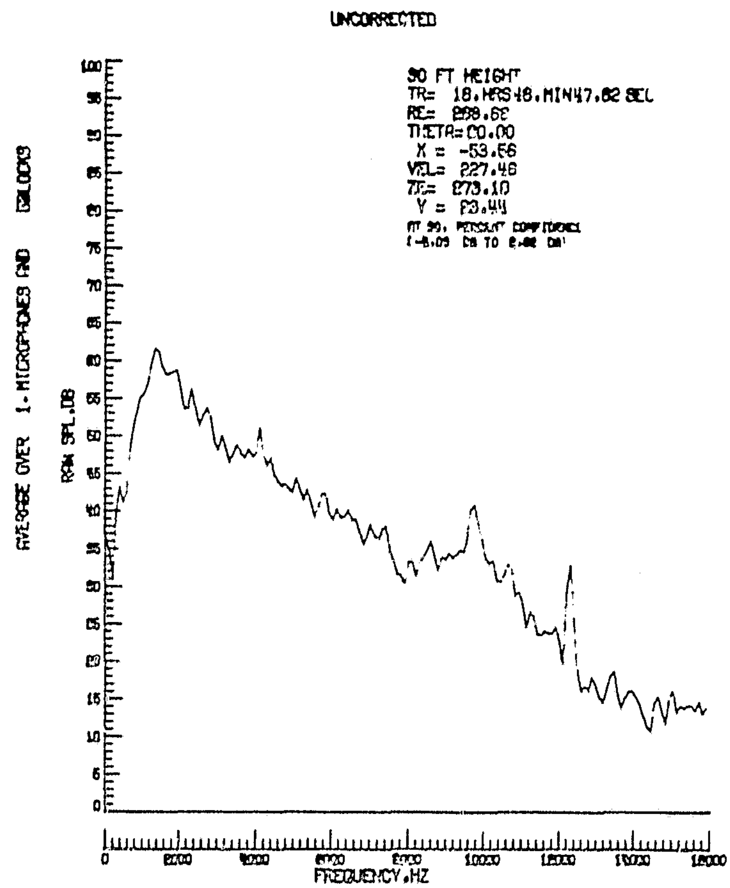

(b) Microphone 2

Figure 8. - Continued

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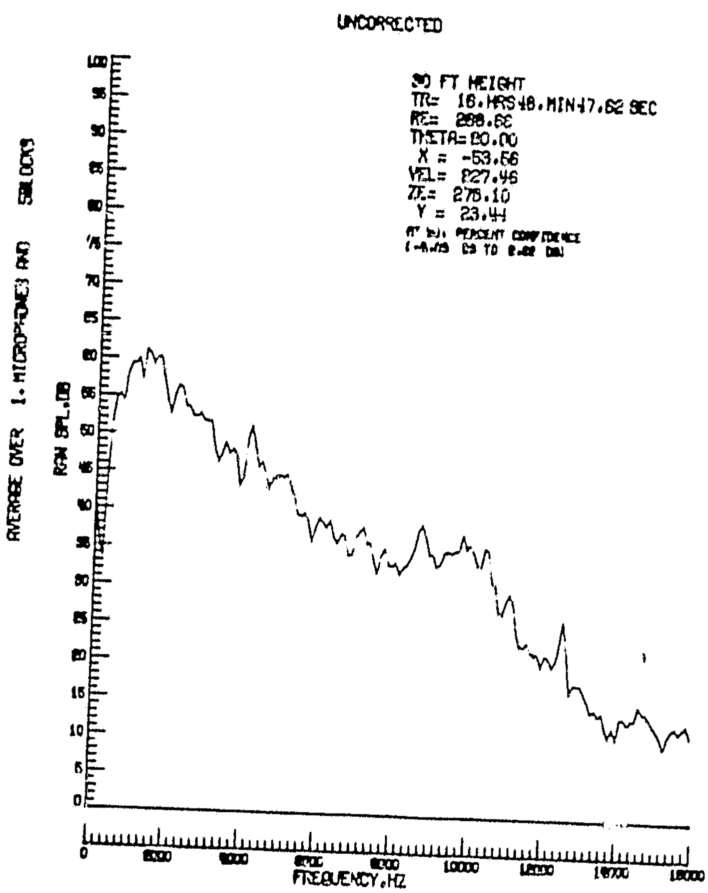

(c) Microphone 3

Figure 8. - Continued

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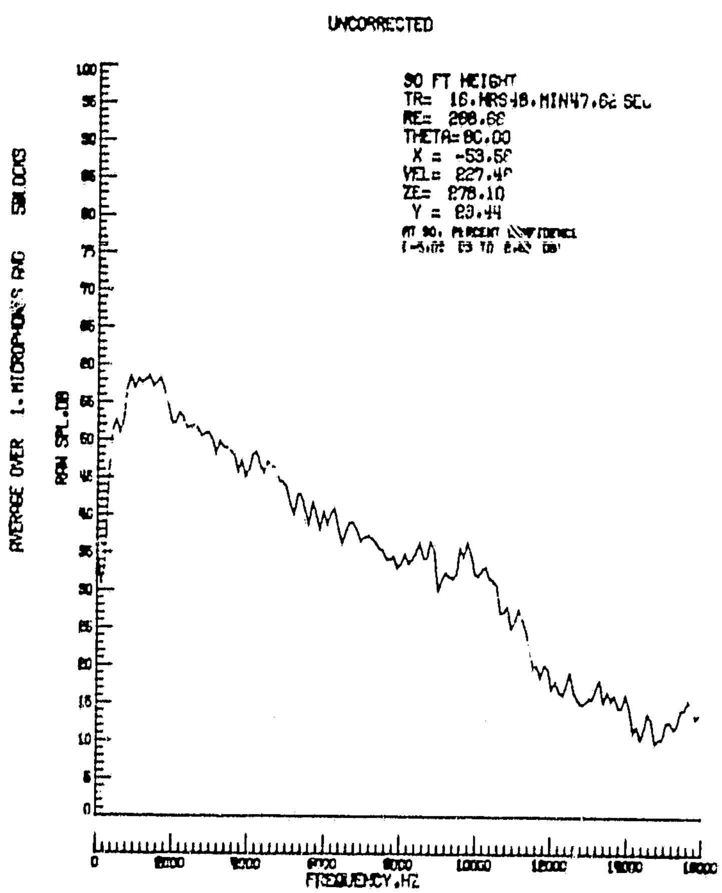

(d) Microphone 4

Figure 8. - Continued

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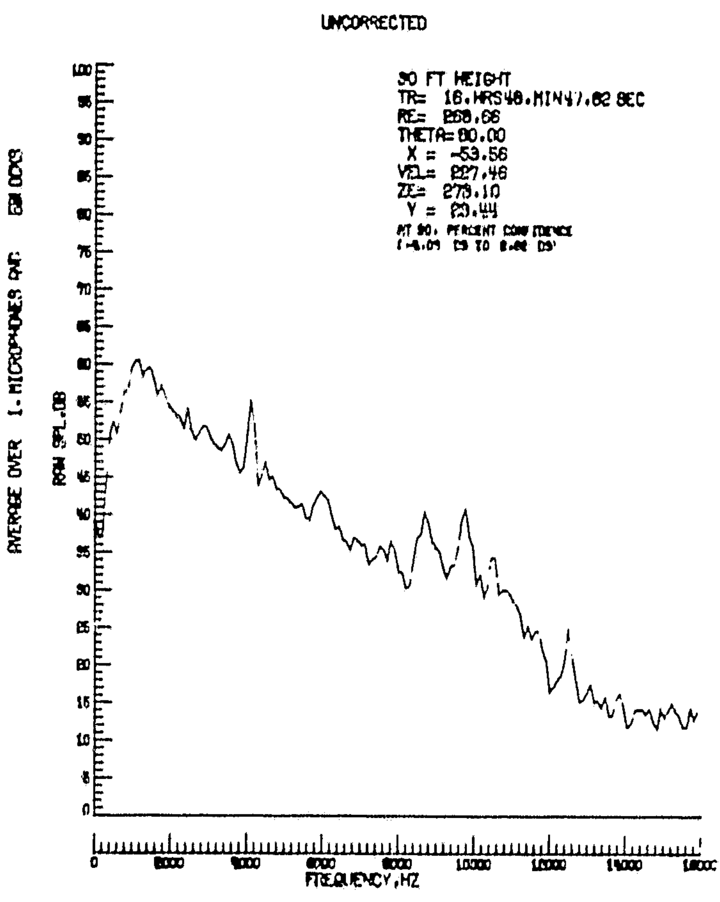

(e) Microphone 5

Figure 8. - Continued

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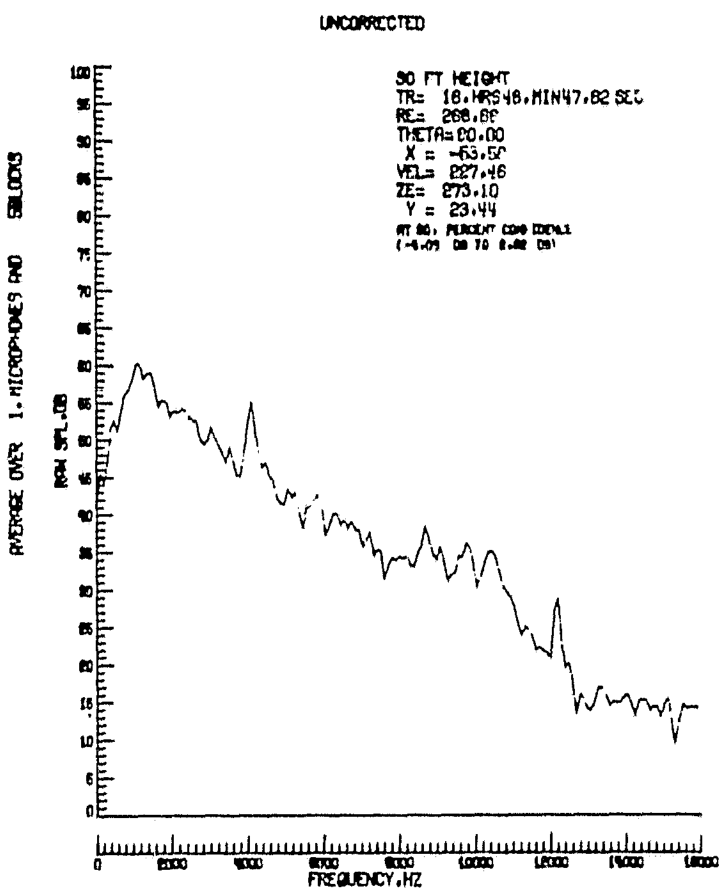

(f) Microphone 6

Figure 8. - Continued

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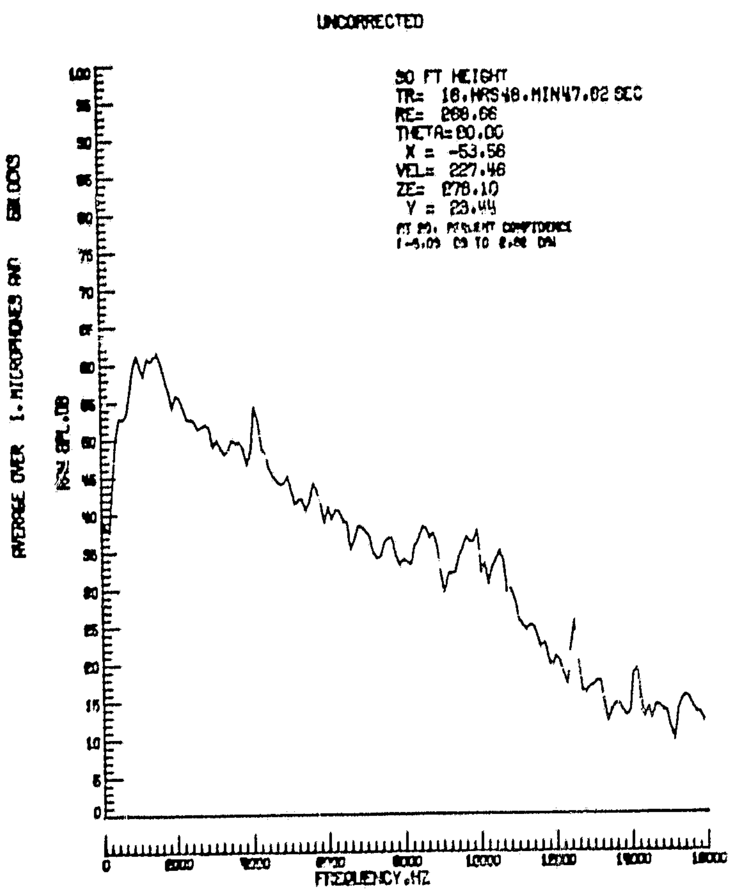

(g) Microphone 7

Figure 8. - Continued

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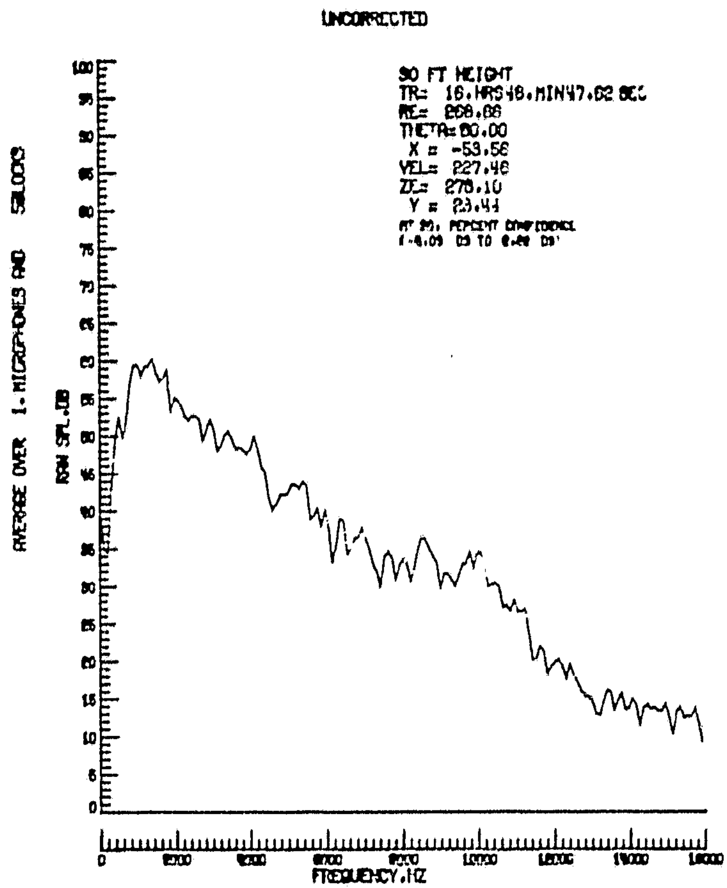

(h) Microphone 8

Figure 8. - Continued

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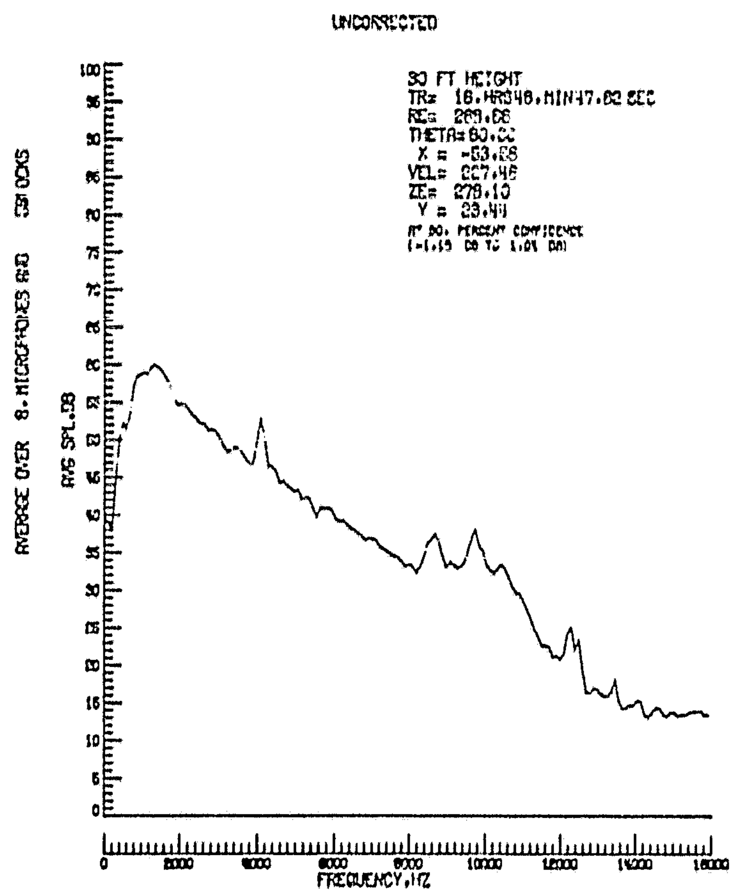

Figure 9. - Averaged sound pressure level

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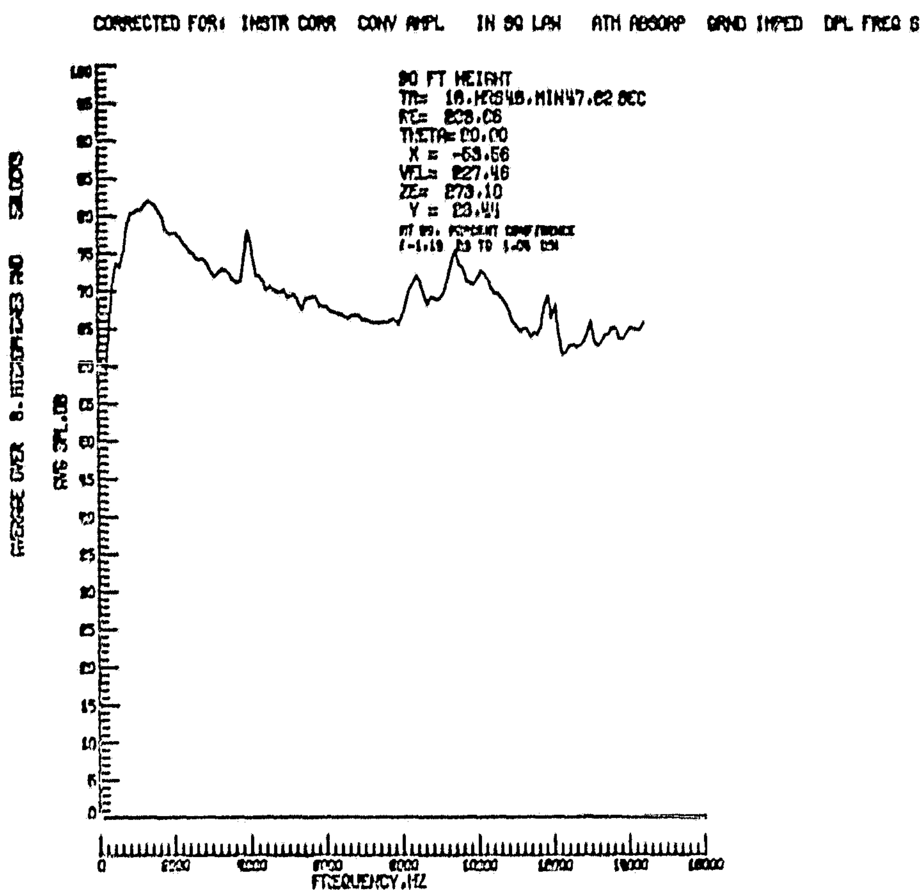

Figure 10. - Averaged and corrected sound pressure level

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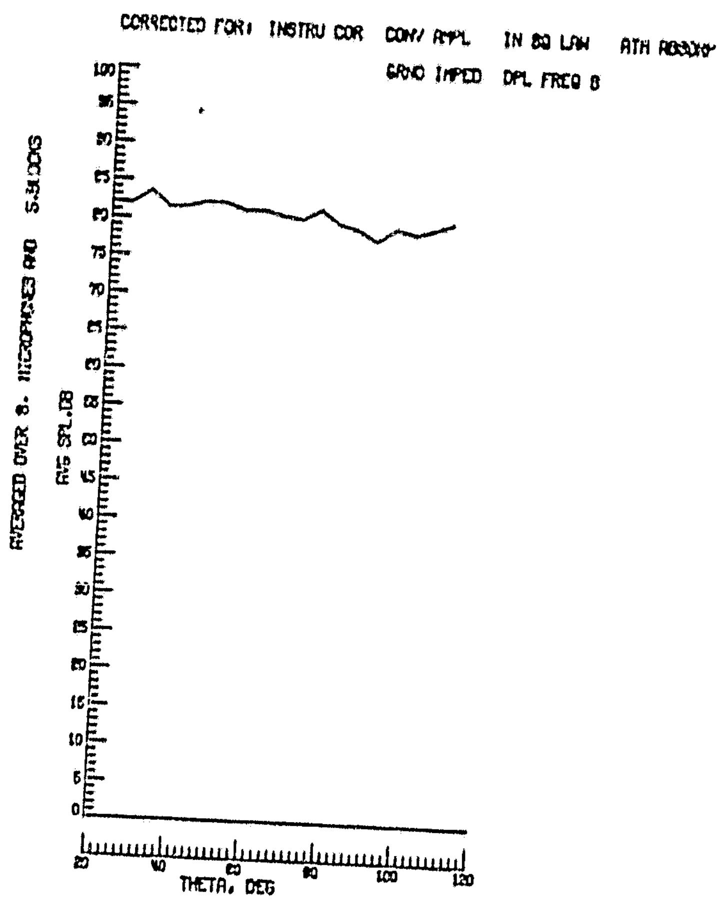

Figure 11. - Directivity pattern

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TABLE 1

EXAMPLE OF INSTRUMENTATION CORRECTIONS ADDED TO
MEASURED SOUND PRESSURE LEVELS

FRQUENCY (kHz)	PRESSURE RESPONSE (dB)	DIFFRACTION (dB)	WINDSCREEN (dB)
1.0	0.0	+.1	0.
1.1	0.0	+.1	0.
1.2	0.0	+.1	0.
1.3	0.0	+.1	0.
1.4	0.0	+.1	0.
1.5	0.0	+.1	0.
1.6	0.0	+.1	0.
1.7	0.0	+.1	0.
1.8	0.0	+.1	0.
1.9	0.0	+.1	0.
2.0	0.0	+.1	0.
2.1	0.0	+.2	-0.3
2.2	0.0	+.2	-0.3
2.3	0.0	+.2	-0.3
2.4	0.0	+.2	0.0
2.5	0.0	+.2	0.0
2.6	0.0	+.2	-0.5
2.7	0.0	+.2	-0.5
2.8	0.0	+.2	-0.5
2.9	0.0	+.2	-0.5
3.0	0.0	+.2	-0.5

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TABLE 2

EXAMPLE OF WEATHER DATA NECESSARY FOR PROPAGATION CORRECTION INPUTS

ALTITUDE (m)	TEMPERATURE (°C)	RELATIVE HUMIDITY (%)	PRESSURE (mbars)
0	12.4	50.0	1013.0
10	12.2	49.9	1011.9
20	11.8	48.8	1010.7
30	11.5	49.0	1009.6
40	11.0	51.1	1008.4
50	10.6	52.0	1007.4
60	10.6	53.0	1006.2
70	10.6	53.0	1005.1
80	10.7	53.0	1003.9
90	10.7	54.0	1002.9
100	10.0	53.0	1001.7

7.0 CONCLUSIONS

Development of an analysis package for the determination of noise generated by a moving source with respect to a stationary observer has been accomplished. The procedures outlined in this document when applied to flyover data yield a static equivalent noise field with a high degree of statistical accuracy. Its utilization for a static/flight comparisons will aid in an understanding of forward speed effects on aircraft flyover noise.

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8.0 SYMBOLS

X'	x position of the aircraft relative to the microphone
Y'	y position of the aircraft relative to the microphone
Z'	z position of the aircraft relative to the microphone
Δt_i	time difference between microphone positions
d_i	distance between microphone i and reference microphone
$\bar{V}$	average velocity of the aircraft
θ	directivity angle of acoustic source
M	number of Fourier transforms averaged
N	number of microphones
L	M•N (total number of averages)
PSD_R	resultant power spectral density
PSD_i	power spectral density of microphone i
ND	number of degrees of freedom
ε	error of estimation
ε_r	normalized random error

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W	data window weighting factor
f_s	frequency of the source
f_o	frequency at the observer
M_c	Mach number of the source
P_s	pressure of the source
P_o	pressure at the observer
n	exponent in convective amplification equation
ΔSPL_1	difference in sound pressure level due to convective amplification
RE	distance between acoustic source and observer
ΔSPL_2	difference in sound pressure level due to the inverse square law
α	absorption coefficient
α_{CL}	classical absorption coefficient
α_{rot}	rotational relaxation coefficient
$\alpha_{vib,N}$	vibrational relaxation of nitrogen coefficient
$\alpha_{vib,O}$	vibrational relaxation of oxygen coefficient
T	temperature

RH	relative humidity
f	frequency
P_{sat}	pressure of saturation
H	absolute humidity in percent
$f_{r,O}$	relaxation frequency of oxygen
$f_{r,N}$	relaxation frequency of nitrogen
$a(f)$	absorption frequency at frequency f
ΔSPL_3	difference in sound pressure level due to atmospheric absorption
r_i	depth of layer i
G	ground factor
k	wave number
Δr	difference between direct and reflected path
f_c	central frequency
ΔSPL_4	difference in sound pressure level due to ground impedance
SPL_{fkc}	corrected SPL at frequency f_k
SPL_{fkm}	measured SPL at frequency f_k
SPL_{fkB}	background SPL at frequency f_k

$\varepsilon(f_k)$	signal-to-noise ratio at f_k
SR	sample rate
BW	bandwidth or frequency resolution
NPTS	number of spectral output points
t_r	time increment of assumed local stationarity
$NSHIFT_i$	number of points corresponding to Δt_i
$P_a^2(f)$	average power at frequency f
REF	reference for dB conversion
$SPL_a(f)$	average sound pressure level at frequency f

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9.0 APPENDICES

APPENDIX A. - Program to Shift Microphone Data

```
      PROGRAM OV1BSH(INPUT,OUTPUT,TAPE1,TAPE2,TAPE3,TAPE4,TAPE5,TAPE6,
     1              TAPE7,TAPE8,TAPE9,TAPE10,TAPE11,TAPE12,
     2              TAPE13,TAPE14=INPUT)
C
C         OV1BSH SHIFTS MICROPHONE DATA FROM OV1B DATA, IF N IS THE
C     MICROPHONE NUMBER EACH CHANNEL GETS SHIFTED FORWARD BY
C           TAU=(D(N-1)/VEL)*SR POINTS WHERE D IS THE MICROPHONE
C     SPACING,VEL IS THE AIRCRAFT'S SPEED AND SR IS THE SAMPLE RATE
C     OF DIGITIZATION.
C         (FOR RUN422, TAU IS IN INCREMENTS OF 9375.)
C
C                                 D. GRIDLEY
C                                 2/22/80
C
C     OV1BSH MAINTAINED BY SYSTEM DEVELOPMENT CORPORATION
C
C
      DIMENSION DATA(1875,12),NAMES(12),IUNITS(12),IHDR(8),IST(10)
      DIMENSION JHDR(4)
      NAMELIST/INPUT/IST,NPTS5
      DATA IST/5,10,15,20,25,30,35,40,45,50/,NPTS5/1875/,
     1     JHDR/4*10H          /
      IBLNK=10H
 201  CONTINUE
C
C      READ SKIP FACTORS FOR CHANNELS
C
      READ (14,INPUT)
      IF (EOF(14)) 202,203
 202  STOP 'NO NAMELIST FOUND'
 203  CONTINUE
      READ (14,5010) JHDR
 5010 FORMAT(4A10)
      IF (NPTS5 .GT. 1875) STOP 'NPTS5 CAN NOT BE GT 1875'
C
C          READ  FROM TAPE1 ALL CHANNELS AND PLACE ON FILES 2 THROUGH 12
C          NPTS5 IS 1/5 OF THE NO. OF POINTS TO SKIP
C          IST IS THE ARRAY OF MULTIPLES OF NPTS5 FOR EACH CHANNEL
C
      READ (1) ISN,NCHAN,(NAMES(I),I=1,NCHAN),(IUNITS(I),I=1,NCHAN),
     1         (IHDR(I),I=1,8)
      IF (EOF(1)) 5,9
  5   STOP ' RECHECK -TAPE1 IS INPUT TAPE- SHOULD HAVE 10 DATA CH'S'
  9   IF (NCHAN.GT.12) GO TO 5
      L1=0 $I1=0
      IJ=0     $IJK=0
  1   DO 10 K=1,NPTS5
          READ (1) (DATA(K,I),I=1,NCHAN)
          IF (EOF(1)) 6,10
  6       LSAV=K-1
      L=LSAV
      IF(L.EQ.0) GO TO 21
          L1=1
          GO TO 11
 10   CONTINUE
      L=NPTS5
 11   WRITE (2)  (DATA(K,1),K=1,L)
      WRITE (2)  (DATA(K,2),K=1,L)
      IJ=IJ+1
      DO 20 J=3,NCHAN
         IF (IST(J-2).GE.IJ) GO TO 20
         WRITE (J) (DATA(K,J),K=1,L)
         IF (J.EQ.NCHAN) IJK=IJK+1
```

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```
 20   CONTINUE
      IF (L1.EQ.0) GO TO 1
 21   DO 30 J=2,NCHAN
          ENDFILE J
          ENDFILE J
          REWIND J
 30   CONTINUE
      CALL EVICT(1)
      IF (JHDR(1) .EQ. IBLNK) GO TO 31
      DO 32 INT=1,4
 32       IHDR(INT)=JHDR(INT)
 31   CONTINUE
      WRITE (13) ISN,NCHAN,(NAMES(I),I=1,NCHAN),(IUNITS(I),I=1,NCHAN),
     1    (IHDR(I),I=1,8)
C
C         WRITE FILES  2 THROUGH 12 ON TAPE1
C
      IJ1=0  $KL=NPTS5
 40   READ (2)(DATA(K,1),K=1,NPTS5)
      READ (2) (DATA(K,2),K=1,NPTS5)
      IJ1=IJ1+1
      IF(IJ1.EQ.IJK) KL=LSAV
      DO 50 J=3,NCHAN
          READ  (J) (DATA(K,J),K=1,KL)
      IF (KL.EQ.LSAV) L=KL
      IF(KL.EQ.LSAV) I1=1
      IF(KL.EQ.LSAV) GO TO 51
          IF (EOF(J))45,50
 45       I1=1
          L=LSAV
          GO TO 51
 50   CONTINUE
      L=NPTS5
 51   IF (L.EQ.0) GO TO 100
      DO 60 I=1,L
          WRITE (13) (DATA(I,J),J=1,NCHAN)
 60   CONTINUE
 100  IF(I1.EQ.0) GO TO 40
      ENDFILE 13  $ENDFILE 13
      STOP ' SUCCESSFUL RUN '
      END
```

APPENDIX B. - Acoustic Analysis Program

The Acoustics Analysis Program is a time series analysis program maintained by

System Development Corporation
3217 North Armistead Avenue
Hampton, Virginia 23666

It utilizes the Cooley-Tukey algorithm for the Fourier Transform which converts time domain data to frequency domain data. Power spectral densities are calculated by the direct method, i.e., the power spectral density is proportional to the Fourier Transform of the data squared. The sound pressure level is simply the power spectral density converted to decibels.

The program also has the capability to calculate, print, and plot one-third octave spectra, auto correlations, cross spectral densities, cross correlations, coherence functions, and transfer functions. It can also retain most of these functions for further calculations. It is a lengthy program, as is its input parameter list. It also is system dependent and will not be presented here.

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APPENDIX C. - Program for Ensemble Averaging

```
      PROGRAM MANDATA(INPUT,OUTPUT,TAPE1,TAPE7,TAPE10,TAPE5=INPUT,
     *                TAPE6=OUTPUT,TAPE8,TAPE4)
C         PROGRAM MANDATA IS A PACKAGE FOR MANIPULATING DATA FROM A
C     TIFT FILE. IT CONSISTS OF BASICALLY TWO ROUTINES: ONE TO STACK
C     FILES  AND ONE TO PEFORM THE BASIC MATHEMATICAL FUNCTIONS.  ITS
C     CREATION WAS NECESSARY FOR MANIPULATING FILES CREATED BY THE
C     ACOUSTICS ANALYSIS PROGRAM(AAP/UN=3458720) AND THE SIGNAL ANALYSIS
C     PROGRAM(NEUSAP UN=349450 ), BUT IT MAY BE USED ON ANY TIFT FILE.
C     THE STACKING ROUTINE,HOWEVER, IS ONE MORE APPLICABLE TO AAP AND
C     NEUSAP CREATED FILES.
C
C                                             DOREEN GRIDLEY
C                                             JANUARY 1980
C
C     MANDATA MAINTAINED BY SYSTEM DEVELOPMENT CORPORATION
C
C                        **** NAMELIST $INPUTS ****
C
C     VARIABLE        DEFAULT          DESCRIPTION
C    **********      *********        ************
C      IOPT           INTEGER          0= DO NOT STACK FILES
C                        1             1= STACK ALL SERIAL NUMBERS
C                                      2= STACK SELECTED SERIAL NUMBERS
C
C      JSN             ARRAY           SERIAL NUMBERS TO BE STACKED(USED
C                      100*0           WHEN IOPT=2)
C                                        EXAMPLE: JSN=1,3,4  IMPLIES
C                                        THAT SERIAL NUMBERS 1,3,AND 4
C                                        WILL BE STACKED
C                                      IF JSN=999, A CONSTANT WILL BE
C                                      WRITTEN TO CHANNEL I.
C
C      CONST          REAL             CONSTANT TO BE EMPLOYED WHEN
C                        1             JSN=999
C
C      IFUNC          INTEGER          0=NO ACTION
C                        0             1=ADD CHANNELS
C                                      2=SUBTRACT CHANNELS
C                                      3=MULTIPLY CHANNELS
C                                      4=DIVIDE CHANNELS
C                                      5=AVERAGE CHANNELS
C                                      6=CHI-SQUARED EQUIVALENCE TEST
C                                         EXAMPLE: IOPT=2,JSN=1,2,4,IFUNC=2
C                                            SERIALS 1,2,AND 4 WILL BE
C                                            STACKED AS CHANNELS 1,2,AND 3
C                                            RESPECTIVELY. CHANNEL 3 WILL
C                                            BE SUBTRACTED FROM CHANNEL 1
C                                            AS WILL CHANNEL 2. THEREFORE,
C                                            THE RESULT=1-2-3
C
C
C      KSN            INTEGER          NEW SERIAL NUMBER OF FIRST FILE
C                        0             RETAINED (SUBSEQUENT FILES,IF ANY,
C                                      WILL BE IN SEQUENTIAL ORDER FROM
C                                      KSN)
C
C      ISPL           INTEGER          0=INPUT IS NOT IN DB
C                        0             1=INPUT IS IN DB
C
C      OSPL           INTEGER          0=OUTPUT NOT TO BE IN DB
C                        0             1=OUTPUT TO BE IN DB
C
```

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```
C       REF            REAL            CONVERSION FACTOR FOR ISPL AND OSPL
C                      0.0002
C
C       IVAR        INTEGER            FLAG TO CALCULATE ASSYMPTOTIC AND
C                      0               ESTIMATED VARIANCES WHEN AVERAGING
C                                        0=DO NO CALCULATE
C                                        1=CALCULATE
C
C                           * * * * * * * * * * * * *
C
      DIMENSION JNAMES(101),JUNITS(101),JSN(100),DATA(101),NAMES(101),
     *          IUNITS(101),HDR(8)
      REAL OSPL,JHDR(8)
      DATA IOPT,JSN,CONST,IFUNC,KSN,ISPL,OSPL,REF/1,100*0,2*0,0,2*0,
     *          0.0002/,IVAR/0/
      NAME IST/INPUT/IOPT,JSN,CONST,IFUNC,KSN,ISPL,OSPL,REF,IVAR
C ***
C ***          READ NAMELIST INPUT
C ***
    1 READ INPUT
      IF (EOF(5)) 9999,2
C ***
C ***          THIS SECTION IS FOR STACKING SERIAL NUMBERS
C ***          THAT CONTAIN ONE CHANNEL OTHER THAN TIME(OR FREQUENCY)
C ***
    2 K=1
      WRITE (6,INPUT)
      IF(IOPT.EQ.0) GO TO 1000
      N=1
      REWIND 1
  100 READ (1) ISN,NCHAN,(NAMES(I),I=1,NCHAN),(IUNITS(I),I=1,NCHAN),
     *          (HDR(I),I=1,8)
      IF (EOF(1)) 900,110
  110 CONTINUE
      IJK=0
      IF (IOPT.EQ.1) GO TO 200
      KN=N-1
      IF (KN.EQ.0) GO TO 150
      DO 120 I=1,KN
  120 IF (ISN.EQ.JSN(I)) GO TO 130
      GO TO 150
C ***          IF NEEDED, CHECK TO SEE IF ISN IS A DESIRED SERIAL NUMBER
  130 READ (1)
  140 IF (EOF(1)) 100,130
  150 IF (JSN(N).EQ.999) GO TO 200
      IF (ISN-JSN(N)) 130,200
  200 K=K+1
C ***          K IS THE NUMBER OF ISNS FOUND+1
      IF (K.NE.2) GO TO 450
      IF (JSN(N).EQ.999) GO TO 300
      DO 210 I=1,2
  210     JNAMES(I)=NAMES(I)
      JUNITS(1)=IUNITS(1)
      ENCODE (10,6000,JUNITS(2)) ISN
 6000 FORMAT (I10)
      DO 220 I=1,8
  220     JHDR(I)=HDR(I)
  250 READ (1) (DATA(I),I=1,2)
      IF (EOF(1)) 600,270
  270 IJK=IJK+1
```

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```
      IF (IJK.EQ.2) TIME=DATA(1)
      IF (JSN(N).NE.999) GO TO 400
300   DATA(1)=0.
      DATA(2)=CONST
400   ITO=7
      IF (MOD(K,2).EQ.0)ITO=8
      WRITE (ITO) (DATA(I),I=1,2)
      GO TO 250
450   IF (JSN(N).NE.999) GO TO 500
      JNAMES(K)=10HCONSTANT
      JUNITS(K)=10H   999
      SDATA=CONST
      GO TO 511
500   ENCODE(10,6001,JUNITS(K)) ISN
6001  FORMAT(I10)
      JNAMES(K)=NAMES(2)
501   READ (1) (DATA(L),L=1,2)
      IF (EOF(1))600,510
510   IJK=IJK+1  $SDATA=DATA(2) $FDATA=DATA(1)
      IF (IJK.NE.2) GO TO 511
      IF (DATA(1).EQ. TIME) GO TO 511
      IF (JSN(N-1).EQ.999) GO TO 511
      PRINT*, 'SERIAL NUMBER ',ISN,' TIMES DO NOT AGREE WITH PREVIOUS SE
     .RIAL'
      STOP 'STACKING NON-COMPATABLE SERIALS'
511   JK=K-1
512   IF (JSN(N).NE.999 .AND. IJK .EQ. 2) TIME=DATA(1)
      ITO=7
      IF (MOD(K,2).EQ.0)ITO=8
      JTO=8
      IF (ITO.EQ.8) JTO=7
      READ (JTO) (DATA(L),L=1,JK)
      IF (EOF(JTO)) 600,520
520   DATA(K)=SDATA
      IF (JSN(N).NE.999) DATA(1)=FDATA
      WRITE(ITO) (DATA(L),L=1,K)
      IF (JSN(N).NE.999)GO TO 501
      SDATA=CONST
      GO TO 512
600   ENDFILE ITO
      ENDFILE ITO
      IF (K.NE.2) REWIND JTO
      REWIND ITO
      N=N+1
      IF (IOPT.EQ.1) GO TO 100
      IF (JSN(N).NE.0) GO TO 100
670   DO 675 I=1,K
         IUNITS(I)=JUNITS(I)
675      NAMES(I)=JNAMES(I)
      DO 680 I=1,K
680      HDR(I)=JHDR(I)
      KSN=KSN+1
      NCHAN=K
      KTO=4
      WRITE (KTO) KSN,NCHAN,(NAMES(I),I=1,NCHAN),(IUNITS(I),I=1,NCHAN),
     .            (HDR(I),I=1,8)
700   READ  (ITO) (DATA(I),I=1,NCHAN)
      IF (EOF(ITO)) 710,720
710   ENDFILE KTO
      PRINT*, ' VALUES ARE STACKED ON TAPE4'
```

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```
      REWIND ITO
      REWIND JTO
      GO TO 1000
 720  WRITE (KTO) (DATA(I),I=1,NCHAN)
      GO TO 700
 900  IF (IOPT.EQ.1) GO TO 670
      STOP 'RECHECK SERIALS--ONE IS NOT FOUND'
 1000 IF (IFUNC.NE.0) GO TO 1001
C
C    TO CONTINUE TO STACK SERIALS ON TAPE4
C
      READ INPUT
      IF (EOF(5)) 1002,2
 1002 ENDFILE KTO
      STOP 'REQUESTED CHANNELS STACKED ON TAPE4'
 1001 IJ=0
      IF (IVAR.NE.0) WRITE (6,5020)
 5020  FORMAT (/ /,10X,*FREQUENCY *,5X,*AVG POWER *,5X,*   VARA   *,
     1  5X,*   VARE*,//)
      M=4
      REWIND M
      READ (M) ISN,NCHAN,(NAMES(I),I=1,NCHAN),(IUNITS(I),I=1,NCHAN),
     .         (HDR(I),I=1,8)
      IF (EOF(M)) 1050,1060
 1050 STOP 'TAPE4 IS EMPTY'
 1060 NCH=NCHAN
      NCHAN=2
      IJKL=0
 1070 READ (M) (DATA(I),I=1,NCH)
      IF (EOF(M)) 2000, 1080
 1080 IJKL=IJKL+1
       IF(ISPL.EQ.1.AND.IJKL.GT.7) CALL FDBCON(DATA,NCH,REF)   $IJ=IJ+1
      GO TO (1100,1200,1300,1400,1500,1700) IFUNC
      STOP 'IFUNC CODE NOT DEFINED PROPERLY'
C ***
C ***            ADDING CHANNELS  A+B+C+D+...
C ***
 1100 NAMES(2)=10HADDED CHS
      ADATA=0
      DO 1125 I=2,NCH
          ADATA=ADATA+DATA(I)
 1125 CONTINUE
      GO TO 1600
C ***
C ***            SUBTRACTING CHANNELS  A-B-C-D-E-F-...
C ***
 1200 NAMES(2)=10HSUBTR CHS
      ADATA=DATA(2)
      DO 1225 I=3,NCH
          ADATA=ADATA-DATA(I)
 1225 CONTINUE
      GO TO 1600
C ***
C ***            MULTIPLYING CHANNELS  A*B*C*D*...
C ***
 1300 NAMES(2)=10HMULTI CHS
      ADATA=1
      DO 1325 I=2,NCH
          ADATA=ADATA*DATA(I)
 1325 CONTINUE
```

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```
      GO TO 1600
C ***
C ***          DIVIDING CHANNELS     A/B/C/D/...
C ***
 1400 NAMES(2)=10HDIVID CHS
      ADATA=DATA(2)
      DO 1425 I=3,NCH
         ADATA=ADATA/DATA(I)
 1425 CONTINUE
      GO TO 1600
C ***
C ***          AVERAGING CHANNELS    (A+B+C+D+...)/(NCH-1)
C ***
 1500 NAMES(2)=10HAVGED CHS
      SUM=0
      DO 1525 I=2,NCH
         SUM=SUM+DATA(I)
 1525 CONTINUE
      ADATA=SUM/(NCH-1)
      IF (IVAR .EQ. 0) GO TO 1600
      VARE=0.
      VARA=ADATA**2/(NCH-1)
      DO 1550 I=2,NCH
          VARE=VARE+((DATA(I)**2)-(ADATA**2))
 1550 CONTINUE
      VARE=VARE/((NCH-1)**2)
      WRITE (6,5010) DATA(1),ADATA,VARA,VARE
 5010 FORMAT(10X,3(E12.5,5X),E12.5)
      GO TO 1600
 1700 CONTINUE
C ***
C ***     CHI-SQUARED EQUIVALENCE TEST
C ***
      IF (IJ.EQ.1) SDATA=0
      SDATA=SDATA+(ALOG10(ABS(DATA(2)/DATA(3))))**2
      GO TO 1070
 1600 DATA(2)=ADATA    $IF(IJ.EQ.1) KSN=KSN+1
      IF (OSPL.EQ.1.AND.IJKL.GT.7) CALL TDBCON(DATA(2),REF)
      IF (IJ.EQ.1) WRITE (10) KSN,NCHAN,(NAMES(I),I=1,NCHAN),
     .                (IUNITS(I),I=1,NCHAN),(HDR(I),I=1,8)
      WRITE (10) (DATA(K),K=1,2)
      GO TO  1070
 2000 IF (IFUNC.EQ.6) GO TO 3000
      ENDFILE 10
      REWIND M
      GO TO 1
 9999 ENDFILE 10
      STOP 'END OF PROGRAM'
 3000 CONTINUE
C *** COMPUTE CHI-SQUARE
      CHISQ=IJ*SDATA/4.
      DF=FLOAT(IJ)
      PRINT*,'DF=',DF,'SDATA=',SDATA,'CHISQ=',CHISQ
      CALL MDCH(CHISQ,DF,P,IER)
      IF (IER.EQ.129 .OR. IER.EQ.34) PRINT*,' ERROR IN USING CHI'
      P=(1-P)*100.
      WRITE (6,5000)
 5000 FORMAT(//,15X,* CHI-SQUARE EQUIVALENCE TEST*,//)
      WRITE(6,5001) NAMES(2),NAMES(3),P
 5001 FORMAT(5X,* THE PROBABILITY THAT CHANNEL *,A10,* AND CHANNEL *,
```

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```
     .A10,/,5X,* ARE STATISTICALLY EQUIVALENT IS *,F10.5,* PERCENT*)
      END
      SUBROUTINE FDBCON(DATA,NCH,REF)
      DIMENSION DATA(101)
      DO 1 I=2,NCH
          DATA(I)=(10.**(DATA(I)/10.))*REF*REF
1     CONTINUE
      RETURN
      END
      SUBROUTINE TDBCON(DAT,REF)
          IF (DAT) 2,1
2         DAT=10.*ALOG10(ABS(DAT)/REF*REF)
1     CONTINUE
      RETURN
      END
```

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APPENDIX D. - Program for Correcting Sound Pressure Levels

```
      PROGRAM FLYOVER(INPUT,OUTPUT,TAPE5=INPUT,TAPE6=OUTPUT,
     1                TAPE1,TAPE2,TAPE10)
C
C         PROGRAM FLYOVER IS TO CORRECT OV1B FLYOVER SOUND PRESSURE
C         LEVEL DATA GENERATED BY THE ACOUSTICS ANALYSIS PROGRAM(AAP).
C         THE PROPAGATION PATH CORRECTIONS ARE:
C
C              1) INSTRUMENTATION CORRECTIONS--PRESSURE RESPONSE,
C                 DIFFRACTION, AND WINDSCREEN CORRECTIONS
C              2) CONVECTIVE AMPLIFICATION
C              3) INVERSE SQUARE LAW
C              4) ATMOSPHERIC ABSORPTION
C              5) GROUND IMPEDANCE
C              6) DOPPLER FREQUENCY SHIFT
C
C         FILES:
C              1) INPUT FILE IS TAPE1
C              2) OUTPUT FILE FOR CORRECTED SPL'S IS TAPE10
C
C     FLYOVER MAINTAINED BY SYSTEM DEVELOPMENT CORPORATION
C
C         THE FIRST INPUT EXPECTED IS A SIX(6) DIGIT CODE BEGINNING
C         IN COLUMN ONE(1). THE ORDER OF THE CODE IS THE ORDER OF
C         THE ABOVE CORRECTIONS.
C              0 IMPLIES DO NOT CORRECT
C              1 IMPLIES CORRECT
C
C              EX.1: 000000    NO CORRECTIONS
C              EX.2: 101000    CORRECT FOR INSTRUMENTATION AND INV. SQ.
C
C         A NAMELIST $INPUTS IS EXPECTED NEXT.
C
C                             NAMELIST $INPUTS
C
C         VARIABLE * TYPE AND *     DESCRIPTION
C                  *  DEFAULT *
C       *  *  *  *  *  *  *  *  *  *  *  *  *  *  *  *  *  *  *  *  *  *
C
C         ICODE      INTEGER      FLAG FOR WHICH THETAS ON TAPE1
C                       0         TO CONSIDER
C                                    0=ALL THETAS ON TAPE1
C                                    1=SELECTED THETAS ACCORDING
C                                      TO ARRAY THTA
C
C         THTA      REAL ARRAY    ARRAY OF THETAS TO BE CONSIDERED WHEN
C                     10*0.0      ICODE=1.
C                                 NOTE: THETA ON FILE TAPE1 WILL BE
C                                       ACCEPTED IF IT IS WITHIN +.1
C                                       OR -.1 OF THTA
C
C         N          INTEGER      FLAG FOR TYPE OF ACOUSTIC SOURCE
C                       0         DISTRIBUTION(FOR CONV. AMP.)
C                                        0=MONOPOLE
C                                        1=DIPOLE
C                                        2=QUADRAPOLE
C
C         STRCR     REAL ARRAY    ARRAY OF VALUES FOR INSTRUMENTATION
C                   800*-999.    CORRECTIONS
C                                   STRCR(I,1) ARE THE FREQUENCIES
C                                   STRCR(I,2) ARE DTSPL FOR P. RESPONSE
C                                   STRCR(I,3) ARE DTSPL FOR DIFFRACTION
C                                   STRCR(I,4) ARE DTSPL FOR WINDSCREEN
```

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```
C
C     HGT         REAL         DISTANCE FROM GROUND TO MICROPHONE(FEET)
C                  30.
C
C     REF         REAL         REFERENCE DISTANCE FOR INVERSE SQUARE
C                  15.            LAW CORRECTION
C
C     P           REAL ARRAY   MEASURED ATMOSPHERIC PRESSUE  (MBARS)
C                 25*-999.        CORRESPONDING TO ARRAY ALT
C
C     T           REAL ARRAY   MEASURED ATMOSPHERIC TEMPERATURE (DEG C)
C                 25*-999.        CORRESPONDING TO ARRAY ALT
C                                 NOTE: THE TEMPERATURE ARRAY MUST
C                                       BE INPUT WITH EACH SUCCESSIVE
C                                       NAMELIST IF APPLYING CORREC-
C                                       TIONS 2,4,5, OR 6
C
C     ALT         REAL ARRAY   MEASURED ALTITUDE (METERS)
C                  25*-999.        INPUT IS TO BE IN DECREASING ORDER
C                                  AND IN EQUAL INCREMENTS
C
C     RH          REAL ARRAY   MEASURED RELATIVE HUMIDITY (PERCENT)
C                  25*-999.            CORRESPONDING TO ARRAY ALT
C
C     IVAR        INTEGER       FLAG TO PRINT OUT RESULTS OF SPL
C                  0            AFTER EACH CORRECTION IS MADE
C                                     0=DO NOT PRINT
C                                     1=PRINT
C
C     F1           REAL         INITIAL FREQUENCY FOR CORRECTIONS
C                  0.0          TO BE APPLIED
C
C     F2           REAL         FINAL FREQUENCY FOR CORRECTIONS TO
C                 50000.        BE APPLIED
C
C     SR           REAL          SAMPLE RATE OF DIGITIZATION
C                 50000.
C
C     NPTS         INTEGER       NUMBER OF POINTS PER BLOCK
C                  512           (VARIABLE NREAD IN AAP)
C
C     NBLKS       INTEGER       NUMBER OF BLOCKS FOR THE SPL USED
C                  5            IN AAP
C
C     NMICS       INTEGER       NUMBER OF MICROPHONES AVERAGED
C                  8            IN MANDATB AFTER AAP
C
C     ITAPE       INTEGER       FLAG FOR WRITING (AND PRINTING) ONLY
C                  0            THOSE FREQUENCIES BETWEEN F1 AND F2
C                                 0=WRITE ALL FREQUENCIES
C                                 1=WRITE FREQUENCIES BETWEEN F1 AND F2
C
C     IPRINT      INTEGER       FLAG FOR PRINTING WHEN IVAR=1
C                                  0=PRINT DELTA SPL VALUES
C                                  1=PRINT CORRESPONDING SPL VALUES
C
C                                 FOR A. MUELLER
C                                 1-12-81
C                                 D.GRIDLEY
C
```

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```
C              (THIS IS A REVISED VERSION OF A PROGRAM DEVELOPED 5-80)
C
      DIMENSION THTA(10),STRCR(200,4),P(25),T(25),ALT(25),RH(25),IFNC(6)
     1          ,DATA(2),NAMES(2),IUNITS(2),IHDR(8)
      NAMELIST /INPUT/ ICODE,THTA,N,STRCR,HGT,REF,P,T,ALT,RH,
     1                 F1,F2,IVAR,ITAPE,IPRINT,SR,NPTS,NBLKS,NMICS
      DATA  ICODE,THTA,N,STRCR/0,10*0.0,0,800*-999./,HGT/30./,
     1      P,T,ALT,RH/100*-999./,IVAR,F1,F2/0,0.,50000./,
     2      NBLKS,NMICS,ITAPE/5,8,0/,IPRINT/0/,SR,NPTS/50000.,512/
      DATA CORR,DTSPL,D2SPL,D3SPL,D4SPL,D5SPL/1.,5*0.0/,REF/15./
C
C     READ ARRAY IFNC FROM INPUT
C
    1 READ (5,5111) (IFNC(I),I=1,6)
 5111 FORMAT(6I1)
      IF (EOF(5)) 1000, 2
C
C     INITIALIZE VARIABLES
C
    2 PI=3.14159265
      P0=1013.25
      T0=293.15
      A=0.01
C
C     READ NAMELIST $INPUT $
C
    3 READ (5,INPUT)
      IF (EOF(5)) 1000,4
    4 CONTINUE
C
C         WRITE INPUT INFORMATION TO OUTPUT FILE
C
      WRITE (6,5222)
 5222 FORMAT(1H1,//,5X,*IFNC = CODE FOR CORRECTIONS(0 MEANS NOT APPLIED*
     1,*  1 MEANS APPLIED)*/,5X,*IN THIS ORDER:*,/,10X,
     2 *INSTRUMENTATION CORRECTIONS*,/,10X,
     3 *CONVECTIVE AMPLIFICATION*,/,10X,
     4 *INVERSE SQUARE LAW*,/,10X,*ATMOSPHERIC ABSORPTION*,/,10X,
     5 *GROUND IMPEDANCE*,/,10X,*DOPPLER FREQUENCY SHIFT*,/)
      WRITE (6,5333) (IFNC(I),I=1,6)
 5333 FORMAT(10X,*IFNC= *,6I1)
      WRITE (6,INPUT)
C
C     READ HEADER RECORD, AND TR,RE,THETA,X,VEL,Z, AND Y (THE FIRST 7
C     DATA RECORDS TO DETERMINE IF THE FILE IS ONE REQUESTED BY ICODE
C     AND THTA
C
      KCNT=0
   10 READ (1) ISN,NCHAN,(NAMES(I),I=1,NCHAN),(IUNITS(I),I=1,NCHAN),
     1          (IHDR(I),I=1,8)
      IF (EOF(1)) 900,20
   20 IJ=0
   25 READ (1) (DATA(I),I=1,NCHAN)
      IF (EOF(1)) 30,40
   30 STOP 'END OF FILE BEFORE RADAR INFORMATION IS READ'
   40 IJ=IJ+1
      IF (IJ .EQ. 1)   TR=DATA(2)
      IF (IJ .EQ. 2)   RE=DATA(2)
      IF (IJ .EQ. 3)THETA=DATA(2)
      IF (IJ .EQ. 4) XRAD=DATA(2)
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```
      IF (IJ .EQ. 5)  VEL=DATA(2)
      IF (IJ .EQ. 6)   ZE=DATA(2)
      IF (IJ .EQ. 7) YRAD=DATA(2)
      IF (IJ .NE. 7) GO TO 25
  100 IF (ICODE .EQ. 0) GO TO 120
      DO 110 I=1,10
          IF (THETA .GE.(THTA(I)-0.1).AND.THETA.LE.(THTA(I)+0.1))
     1    GO TO 120
  110 CONTINUE
C
C         OTHERWISE CALL SKIPFF
C
      CALL SKIPFF(SLTAPE1,1)
      GO TO 10
  120 KCNT=KCNT+1
C
C     WRITE HEADER RECORD AND RADAR INFORMATION TO TAPE10
C
      WRITE (10) ISN,NCHAN,(NAMES(I),I=1,NCHAN),(IUNITS(I),I=1,NCHAN),
     1          (IHDR(I),I=1,8)
      DUM=-999.
      WRITE (10) DUM,TR
      DUM=-998.
      WRITE (10) DUM,RE
      DUM=-997.
      WRITE (10) DUM,THETA
      DUM=-996.
      WRITE (10) DUM,XRAD
      DUM=-995.
      WRITE (10) DUM,VEL
      DUM=-994.
      WRITE (10) DUM,ZE
      DUM=-993.
      WRITE (10) DUM,YRAD
      IF (KCNT .GT. 1) GO TO 180
C
C     ADD 273.15 TO TEMP ARRAY
C
      DO 135 I=1,25
  135     IF (T(I) .NE. -999.)T(I)=T(I)+273.15
C
C     COMPUTE AVERAGE TEMPERATURE FOR AVERAGE C FOR DF,AND CA
C
      AT=0.
       ICNT=0
      I=0
  140 I=I+1
      IF (T(I) .EQ. -999. .OR. I  .GT. 25) GO TO 150
      IF((ALT(I) .LT. (HGT*.3048)) .OR. (ALT(I) .GT. (ZE*.3048)))
     1   GO TO 140
      AT=AT + T(I)
      ICNT=ICNT+1
      GO TO 140
  150 AT=AT/(FLOAT(ICNT))
      TA=AT/TO
      CAVG=(343.23*SQRT(TA))/0.3048
C
C       COMPUTE DELTA HEIGHT IN ALTITUDE
C
      DHGT=ALT(1)-ALT(2)
```

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```
C
C     COMPUTE C CLOSEST TO GROUND FOR GROUND IMPEDANCE
C
      ALTM=99999.
      DO 160 I=1,25
          IF (ALT(I) .EQ. -999.) GO TO 170
          IF (ALT(I) .GE. ALTM) GO TO 160
          ALTM=ALT(I)
          ID=I
  160 CONTINUE
  170 CGI=(343.23*SQRT(T(ID)/TO))/0.3048
C
C         COMPUTE VARIABLES FOR CORRECTIONS
C
C              CORR IS FOR THE DOPPLER FREQUENCY SHIFT
C              AMCH IS THE AVERAGE MACH NUMBER
C              DTSPL IS FOR THE INVERSE SQUARE LAW
C              D2SPL IS FOR CONVECTIVE AMPLIFICATION
C              DK IS A NUMBER BASED ON BANDWIDTH
C              DR IS THE PATH DIFFERENCE BETWEEN REFLECTED AND
C                 DIRECT PATH
C              DKDR IS THE PRODUCT OF DK AND DR
C
  180 AMCH=VEL/CAVG
      CORR=(1.-AMCH*COS(THETA*PI/180.))
      DTSPL=20.*ALOG10(REF/RE)
      IF (IFNC(3) .EQ. 0) DTSPL=0.0
      D2SPL=20.*(2*N+2)*ALOG10(CORR)
      IF (IFNC(2) .EQ. 0) D2SPL=0.0
      DK=(PI*SR)/(FLOAT(NPTS)*CGI)
C     DR=2.*HGT*SIN(THETA*PI/180.)
C     OLD METHOD OF CALUCULATING DR--INVALID DUE TO CHOICE OF ANGLE
C
      DR=SQRT(RE*RE+4.*ZE*HGT)-RE
      DKDR=DK*DR
      IF (DKDR .EQ. 0) W1=1.
      IF (DKDR .NE. 0) U1=SIN(DKDR)/DKDR
C
C     WRITE TABLE HEADINGS ON OUTPUT IF IVAR=1
C
      IF (IVAR .NE. 1) GO TO 130
      WRITE (6,5888) (IHDR(L),L=1,8)
 5888 FORMAT (1H1,///,8A10)
      WRITE (6,5444) TR,RE,THETA,XRAD,VEL,ZE,YRAD
     1                ,CAVG,CGI,AMCH,NMICS,NBLKS
 5444 FORMAT(//,5X,*T=*,E12.5,4X,*R=*,E12.5,4X,*THETA=*,E12.5,4X,
     1*X=*,E12.5,4X,*V=*,E12.5,4X,*Z=*,E12.5,4X,*Y=*,E12.5,//,
     A30X,*CAVG= *,E12.5,4X,*CGI= *,E12.5,4X,*M= *,E12.5,///,20X,
     2*AVERAGE OVER *,I4,* MICROPHONES, AND *,I4,* BLOCKS*,//,10X,
     3*CODE FOR CORRECTIONS MADE:*,/,15X,*IC=INSTRUMENTATION*,/,15X,
     4*CA=CONVECTIVE AMPLIFICATION*,/,15X,*IS=INVERSE SQUARE LAW*,/,15X,
     5*AA=ATMOSPHERIC ABSORPTION*,/,15X,*GI=GROUND IMPEDANCE*,/,15X,
     6*DF=DOPPLER FREQUENCY SHIFT*)
      IF (IPRINT .NE. 0) WRITE (6,5666)
 5666 FORMAT(//,4X,* FREQUENCY*,
     16(4X,*AVG SPL,DB*),4X,*FREQUENCY*,/,35X,*(IC)*,7X,*(IC,CA)*,7X,
     2*(IC,CA,IS)*,4X,*(IC,CA,IS,AA)*,1X,*(   ALL   )*,5X,*(DF)*,//)
      IF (IPRINT .EQ. 0) WRITE (6,5777)
 5777 FORMAT(//,4X,* FREQUENCY*,4X,*AVG SPL,DB*,
     1 5(4X,*DELTA SPL *),4X,*FREQUENCY*4X*AVG SPL,DB*,/,35X,*(IC)*,9X,
```

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```
    2 *(CA)*,11X,*(IS)*,12X,*(AA)*,8X,*(GI)*,7X,*(DF)*,
    3 10X,*FINAL*,//)
  130 CONTINUE
C
C     READ AND APPLY REQUESTED CORRECTIONS
C
  200 READ (1) F,SPL
      IF (EOF(1)) 500,210
  210 IF (F .GE. F1 .AND.F .LE. F2) GO TO 215
      IF (ITAPE .NE. 0) GO TO 200
      FFF=F
      SPL1=SPL2=SPL3=SPL4=SPL5=SPL
      IF (IVAR.NE. 1) GO TO 460
      GO TO 455
  215 IF (IFNC(1) .EQ. 0)  GO TO 300
C
C     INSTRUMENTATION CORRECTIONS
C
      FF=DB1=DB2=DB3=0.0
          IF (F.GT. STRCR(1,1))GO TO 220
          IND=0
          GO TO 270
  220 DO 260 I=1,200
          IF (STRCR(I,1) .EQ. -999.) GO TO 280
          IF (F.LT. STRCR(I,1)) GO TO 270
          IND=I
          FF=STRCR(I,1)
          DB1=STRCR(I,2)
          DB2=STRCR(I,3)
          DB3=STRCR(I,4)
  260 CONTINUE
  270 IND1=IND+1
      RATIO=(F-FF)/(STRCR(IND1,1)-FF)
      DB1= DB1 + RATIO*(STRCR(IND1,2)-DB1) $ DB1=INT(DB1*10.+.5)/10.
      DB2= DB2 + RATIO*(STRCR(IND1,3)-DB2) $ DB2=INT(DB2*10.+.5)/10.
      DB3= DB3 + RATIO*(STRCR(IND1,4)-DB3) $ DB3=INT(DB3*10.+.5)/10.
  280 D5SPL=DB1 +DB2 +DB3
      SPL=SPL+D5SPL
  300 IF (IFNC(2) .EQ. 0) GO TO 310
C
C          CONVECTIVE AMPLIFICATION
C
      SPL=SPL+D2SPL
  310 IF (IFNC(3) .EQ. 0) GO TO 320
C
C           INVERSE SQUARE LAW
C
      SPL=SPL-DTSPL
  320 IF (IFNC(4) .EQ. 0)  GO TO 400
C
C          ATMOSPHERIC ABSORPTION
C
      D4SPL=0.0
      DO 350 I=1,25
      IF (ALT(I) .EQ. -999.) GO TO 350
      IF ((ALT(I) .LT. (HGT*.3048)) .OR. (ALT(I) .GT. (ZE*.3048)))
     1    GO TO 350
          C=343.23*SQRT(T(I)/TO)*100./(2.54*12.)
      SATR=10.79586*(1.-(273.16/T(I))) -5.02808*ALOG10(T(I)/273.16)
     1    +1.50474*0.0001*(1.-10.**(-8.29692*((T(I)/273.16)-1.)))
```

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```
     2    +0.42873*0.001 *(10.**(4.76955*(1.-(273.15/T(I))))-1.)
     3    -2.2195983
      PR=10.**SATR
      HUM=RH(I)*PR*P(I)/PO
      FRLO=(P(I)/PO)*( 24. +44100*HUM*((0.05+HUM)/(0.391+HUM)))
      FRLN=(P(I)/PO)*SQRT(TO/T(I))*( 9. + 350.*HUM*
     1        EXP(-6.142*((T(I)/TO)**(-1./3.)  -1.)))
      ALPHA= 8.686*SQRT(T(I)/TO)*(F*F*PO/P(I))*
     1        (1.84*10.**(-11.) + 2.19*0.0001*(TO/T(I))*(P(I)/PO)*
     2        (2239./T(I))**2. * EXP(-2239./T(I)) /
     3        (FRLO +(F*F/FRLO))  + 8.16*0.0001*(TO/T(I))*
     4        (P(I)/PO)*(3352./T(I))*(3352./T(I))*
     5        EXP(-3352./T(I))/(FRLN+ (F*F/FRLN)))
      D4SPL=D4SPL + ALPHA*DHGT/(SIN(THETA*PI/180.))
  350 CONTINUE
      SPL=SPL+D4SPL
  360 CONTINUE
  400 IF (IFNC(5) .EQ. 0) GO TO 410
C
C              GROUND IMPEDANCE
C
          DUM=(-(A*2.*PI*F*DR/CGI)**2.)
          IF (DUM .LT. -675.84) W2=0.
          IF (DUM .GE. -675.84) W2=EXP(DUM)
          D3SPL=10.*ALOG10 (2. +2.*W2*COS(2.*PI*F*DR/CGI)*W1)
          SPL=SPL-D3SPL
  410 IF (IFNC(6) .EQ. 0) GO TO 420
C
C         DOPPLER FREQUENCY SHIFT
C
          F=F*CORR
  420 CONTINUE
C
C     SET UP AND PRINT VALUES IF IVAR=1
  450 IF (IVAR .NE. 1) GO TO 460
      IF (IFNC(6) .EQ. 0)  CORR=1.0
      FFF=F/CORR
      SPL1=SPL+D3SPL-D4SPL+DTSPL-D2SPL-D5SPL
      SPL2=SPL+D3SPL-D4SPL+DTSPL-D2SPL
      SPL3=SPL+D3SPL-D4SPL+DTSPL
      SPL4=SPL+D3SPL-D4SPL
      SPL5=SPL+D3SPL
  455 IF (IPRINT .NE. 0)
     1WRITE (6,5555) FFF,SPL1,SPL2,SPL3,SPL4,SPL5,SPL,F
      IF (IPRINT .NE. 0) GO TO 460
      D7SPL=-DTSPL
      D8SPL=-D3SPL
      WRITE(6,5555) FFF,SPL1,D5SPL,D2SPL,D7SPL,D4SPL,D8SPL,F,SPL
 5555 FORMAT(2X,F10.4,8(4X,F10.4))
 460  WRITE (10) F,SPL
      GO TO 200
  500 ENDFILE 10
      GO TO 10
  900 REWIND 1
      GO TO 1
 1000 ENDFILE 10
      STOP 'END OF PROGRAM'
      END
```

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APPENDIX E. - Program for Acoustic Directivities

```
      PROGRAM SPLTHTC(INPLT,OUTPUT,TAPE5=INPUT,TAPE6=OUTPUT,
     1                TAPE1,TAPE2,TAPE9,TAPE10)
C
C
C          PROGRAM SPLTHTC IS A PROGRAM WHICH GENERATES THE
C          RADIATION PATTERN (SPL VERSUS THETA) AT A DESIRED
C          FREQUENCY FOR ALL FILES ON TAPE1 WHICH ARE GENERATED
C          BY THE ACOUSTICS ANALYSIS PROGRAM(AAP) WITH MOD FILES
C          OVAAP30,OVAAP12, AND OVAAPGD APPLIED.
C
C                                        FOR OVIB FORWARD SPEED EFFECTS
C                                                   (A. MUELLER)
C
C                                        D. GRIDLEY (1981)
C
C     SPLTHTC MAINTAINED BY SYSTEM DEVELOPMENT CORPORATION
C
C          I.   FILES
C               A.   TAPE1 IS THE SPL FILE GENERATED BY AAP--THE
C                    FIRST 7 DATA RECORDS ARE RADAR INFORMATION
C               B.   TAPE2 IS THE RADIATION PATTERN FILE FOR THE
C                    BACKGROUND RUN (ONLY USED AS INPUT WHEN
C                    IBSUB=1)
C               C.   TAPE9 IS THE NEW RADIATION PATTERN FILE
C               D.   THPE10 IS THE NEW RADIATION PATTERN MINUS THE
C                    BACKGROUND (TAPE2) (ONLY OUTPUT WHEN IBSUB=1)
C
C         II.   NAMELIST $NAME$
C               A.FREQ IS THE DESIREC FREQUENCY AT WHICH TO GENERATE
C                 THE RADIATION PATTERN
C               B.ERR IS THE BANDWIDTH OVER WHICH TO FIND SPL VALUE
C                 (RANGE OF  FREQ-ERR TO FREQ+ERR IS USED)
C               C.IBSUB IS THE FLAG FOR SUBTRACTING BACKGROUND
C                 (0 = DO NOT SUBTRACT   1= SUBTRACT)
C               D.ISUMTYP IS THE FLAG FOR COMPUTATIONS
C                 (0=HIGHEST VALUE IN RANGE ONLY,1=SUM OF TWO
C                  HIGHEST SPL VALUES IN THE RANGE)
C
C
      DIMENSION NAMES(2),IUNITS(2),IHDR(8),DATA(2),JNAMES(2),JUNITS(2),
     1          THETAB(50),DBB(50)
      NAMELIST /NAME/FREQ,ERR,IBSUB,ISUMTYP
      DATA ERR/200/,FREQ/2000/,IBSUB/0/,ISUMTYP/0/,
     1     JNAMES/10H  THETA   ,10H    SPL   /,
     2     JUNITS/10H  DEGREES ,10H    DB    /
    1 READ (5,NAME)
      IF (EOF(5)) 1000,5
    5 WRITE (6,NAME)
      K=0
   10 READ (1) ISN,NCHAN,(NAMES(I),I=1,NCHAN),(IUNITS(I),I=1,NCHAN),
     1        (IHDR(I),I=1,8)
      IF (EOF(1)) 500,20
   20 K=K+1
      IF (K .EQ. 1) WRITE (9) ISN,NCHAN,(JNAMES(I),I=1,NCHAN),
     1              (JUNITS(I),I=1,NCHAN),(IHDR(I),I=1,8)
      J=0
      DBMAX=-99999.
      DB1MAX=0.0
   30 READ (1) (DATA(I),I=1,NCHAN)
      IF (EOF(1)) 35,40
   35 STOP 'FREQUENCY REQUESTED IS TOO LARGE'
   40 J=J+1
```

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```
      IF (J .GT. 7) GO TO 50
      IF (J .EQ. 3) THETA=DATA(2)
      GO TO 30
   50 IF (DATA(1) .GT.(FREQ +ERR)) GO TO 60
      IF (DATA(1) .LT.(FREQ -ERR)) GO TO 30
C
C          FIND MAXIMUM SPL VALUE IN RANGE
C
      IF (ISUMTYP .EQ. 0) GO TO 55
      IF (DATA(2) .LT. DB1MAX) GO TO 30
      IF (DATA(2) .LT. DBMAX) GO TO 52
      DB1MAX=DBMAX $DBMAX=DATA(2)
      GO TO 30
   52 DB1MAX=DATA(2)
      GO TO 30
   55 IF (DATA(2) .GT. DBMAX) DBMAX=DATA(2)
      GO TO 30
   60 IF (ISUMTYP .EQ. 0) GO TO 65
      DBMAX=(10.**(DBMAX/20.))*0.0002
      DB1MAX=(10.**(DB1MAX/20.))*0.0002
      DBMAX=10.*ALOG10((DBMAX*DBMAX+DB1MAX*DB1MAX)/(.0002*.0002))
   65 WRITE (9) THETA,DBMAX
      READ (1)
      IF (EOF(1)) 10,70
   70 CALL SKIPFF(SLTAPE1,1)
      GO TO 10
  500 ENDFILE 9
      REWIND 1
      IF (IBSUB .EQ. 0)  GO TO 1
      ENDFILE 9
      REWIND 9
      REWIND 2
C
C          PLACE BKG INTO ARRAY THETAB AND DBB
C
  510 READ (2) ISN
      IF (EOF(2)) 520,530
  520 STOP  'NO RADAR FILE INFO.'
  530 IJ=0
  540 IJ=IJ+1
      READ (2) THETAB(IJ),DBB(IJ)
      IF (EOF(2)) 550,540
  550 KNT=IJ-1
C
C          READ SPL VS THETA AND SUBTRACT BKG--ONLY THOSE THETAS
C          WITH A CORRESPONDING THETAB WILL BE WRITTEN TO TAPE10.
C
  600 READ (9) ISN,NCHAN,(NAMES(I),I=1,NCHAN),(IUNITS(I),I=1,NCHAN),
     1          (IHDR(I),I=1,8)
      IF (EOF(9)) 1000,610
  610 WRITE (10) ISN,NCHAN,(NAMES(I),I=1,NCHAN),(IUNITS(I),I=1,NCHAN),
     1            (IHDR(I),I=1,8)
  620 READ (9) THETA,DB
      IF (EOF(9)) 700,630
  630 L=0
  640 L=L+1
      IF (L .GT. KNT) GO TO 620
      IF (THETAB(L) .GT. (THETA+1.) .OR. THETAB(L) .LT. (THETA-1.))
     1    GO TO 640
  650 CONTINUE
```

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```
      IF((DB-DBB(L)) .GT. 3.) GO TO 660
      WRITE (6,5660) ISN,THETA
 5660 FORMAT(/,* FOR SERIAL *,I6,* S/N IS < OR = 3 FOR ANGLE *,F10.4)
      IF (DB-DBB(L) .LE. 0) GO TO 670
      GO TO 665
  660 IF ((DB-DBB(L)) .GE. 10.) GO TO 670
  665 DB=DB+10.*ALOG10(1.-10.**((DBB(L)-DB)/10.))
  670 WRITE (10) THETA,DB
      GO TO 620
  700 ENDFILE 10
      ENDFILE 10
      GO TO 1
 1000 STOP 'END OF PROGRAM'
      END
```

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REFERENCES

1. Chun, K. S.; Berman, C. H.; and Cowan, S. J.: "Effects of Motion on Jet Exhaust From Aircraft". NASA CR-2701, June 1976.

2. Bendat, J. S.; and Piersol, A. G.: Random Data: Analysis and Measurement Procedures. John Wiley and Sons, Inc., New York, 1971.

3. McDaid, E. P.; and Maestrello, L.: "Estimation of Spectra from Moving Sound Sources". AIAA Paper No. 72-667, June 1972.

4. Otnes, R. K.; and Enochson, L.: Digital Time Series Analysis. John Wiley and Sons, Inc., New York, 1972.

5. Brown, T. J.; Brown, C. G.; and Hardin, J. C.: "Program for the Analysis of Time Series". NASA TMX-2988, September 1974.

6. Rao, K. V.; and Preisser, J. S.: "Spectral Variance and Aeroacoustical Data". The Journal of the Acoustical Society of America, May 1981.

7. Dowling, A.: "Convective Amplification of Real Simple Sources". Journal of Fluid Mechanics, Vol. 74, Part 3, 1976.

8. Harris, C. M.: "Absorption of Sound in Air Versus Humidity and Temperature". The Journal of the Acoustical Society of America, Vol. 40, pp 148-162, 1966.

9. Greenspan, M.: "Rotational Relaxation in Nitrogen, Oxygen, and Air". The Journal of the Acoustical Society of America, Vol. 31, pp 155-160, 1959.

ORIGINAL PAGE IS
OF POOR QUALITY

10. Shields, F. D.; and Bass, H. E.: "Atmospheric Absorption of High Frequency Noise and Application to Fractional-Octave Band". NASA CR-2760, June 1977.

11. Pao, S. P.; Wenzel, A. R.; and Oncley, P. B.: "Prediction of Ground Effects on Aircraft Noise". NASA TP-1104, 1978.

12. Gridley, D.: "An Introduction to Time Series Analysis and Data Reduction Programming Capabilities". NASA CR-165732, June 1981.

13. Mueller, A. W.; and Preisser, J. S.: "Flight Test of a Pure Tone Acoustic Source". NASA TP-1898, October 1981.