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

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CONTRACT NAS1-16000

March 1982
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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 $E_{PNdB}$ (Effective Perceived Noise in decibels), where much of the detailed information is "lost" due to the requirements for computing $E_{PNdB}$.

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 (θ=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
different recording times, barring any transient occurrences with the source production or any atmospheric disturbances. Referring to Figure 3, the time separation $\Delta t_i$ is

$$\Delta t_i = \frac{d_i}{V}$$

where $d_i$ is the distance between each microphone and the reference microphone, and $V$ is the velocity directly above the microphone array. Spectra at corresponding angles, $\theta$, may be averaged to lead to a resulting ensemble averaged spectra. Thus, the total number of averages $N$ 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 $N = M \times N$. The final power spectral density $PSD_R$ may be expressed as

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

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 calculation via the direct method (Ref. 4),

$$ND = 2M.$$
Figure 3. - Microphone position and directivity angle
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 $c$ of estimation is

$$c = \frac{1}{\sqrt{L}} \tau \sqrt{\frac{2}{N\delta}}$$

This states that the error $c$ decreases as $N\delta$ increases. Once $N\delta$ 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 $\pm 1$ 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 $e_r$ in percent,

$$L = (e_r)^2 W$$
where $W$ is the weighting factor for the data window applied to the time domain data when calculating the Discrete Fourier Transform. $c_r$ is a measure of the convergence of mean spectra. If $c_r$ is 10, which corresponds to a 90% confidence, and the Hann data window is applied ($W = 3/8$),

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

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 $\mathbf{\bar{V}}$, 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 ($M_c = \mathbf{\bar{V}}/c$ where $c$ is the speed of sound in the medium through which the wave travels) and $\theta$ 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.

\[
\begin{align*}
0 & \quad \text{Monopole} \\
1 & \quad \text{Dipole} \\
2 & \quad \text{Quadrapole}
\end{align*}
\]

The expression \( 1 - M_c \cos \phi \) is the same as was defined in the Doppler effect. The difference in sound pressure level between static and flight cases \( \Delta \text{ASPL}_1 \) is

\[
\Delta \text{ASPL}_1 = 20(2n+2) \log_{10} (1 - M_c \cos \phi)
\]

### 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 \( \Delta \text{ASPL}_2 \) may then be expressed as

\[
\Delta \text{ASPL}_2 = -20 \log_{10} \left( \frac{r_1}{r_2} \right)
\]
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 $\alpha$ 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 $\alpha$. 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 $\alpha_{CL}$, rotational relaxation $\alpha_{rot}$, 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

\[ \log_{10}(P_{\text{sat}}/P_0) = 10.79586 [1 - (T_{01}/T)] \]

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

\[+0.42873 \times 10^{-3}(10^{4.76955[1-(T_{01}/T)]} - 1) - 2.2195983\]

where $P_0$ = reference pressure of $1.013 \times 10^5$ N m$^2$

$T_{01} = 273.16^\circ$K

2) Calculate the absolute humidity $H$ in %

\[ H = RH \left(\frac{P_{\text{sat}}}{P_0}\right)/\left(\frac{P}{P_0}\right) \]

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

\[ f_{r,0} = \left(\frac{P}{P_0}\right) \left[24 + 4.41 \times 10^4 H(0.05 + H)/(0.391 + H)\right] \]

\[ f_{r,N} = \left(\frac{P}{P_0}\right)(T/T_0)^{-1/2}[9 + 350H \exp (-6.142[(T/T_0)]^{-1/3} - 1)] \]

where $T_0 = \text{reference temperature of } 293.15^\circ$K.

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

\[ a(f) = 8.686(T/T_0)^{1/2}[f^2/(P/P_0)] \times \]

\[(1.84 \times 10^{-11} + 2.19 \times 10^{-4}(T/T_0)^{-1}(P/P_0)(2239/T)^2 \times \]

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

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\[ + 8.16 \times 10^{-4} \left( \frac{T}{T_0} \right)^{-1} \left( \frac{P}{P_0} \right) \left( \frac{3352}{T} \right)^2 \times \]

\[ \frac{\exp \left( -\frac{3352}{T} \right)}{[\frac{\left( f_{r,N} \right)}{N} + \left( \frac{f^2}{f_{r,N}} \right) r_0]} \]

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 (\( \Delta \)) the attenuation of the sound wave in terms of dB, \( \Delta \text{ASPL}_3 \), is

\[ \Delta \text{ASPL}_3 = \sum_{i=1}^{N} a_i(f) \cdot r_i \]

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

\[ G = 2 + 2^{-} (\frac{a k \Delta r}{c})^2 \cos(k \Delta r) \]

where \( k \) is the wave number \( \left( \frac{2 \pi f}{c} \right) \), \( a = 0.01 \), and \( \Delta r \) 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
\[ \Delta \Gamma \approx \sqrt{RE^2 + 4Zh} - RE \]
if \( h \ll Z \) and \( h \ll RE \)

Figure 5. - Ground impedance model
\[ G = 2 + 2 e^{-\left(\frac{ak\Delta r}{c}\right)^2} \cos\left(k_c\Delta r\right) \frac{\sin\left(\frac{ak\Delta r}{c}\right)}{\Delta k\Delta r} \]

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

\[ \Delta SPL_b = -10 \log \left(2 + 2 e^{-\left(\frac{ak\Delta r}{c}\right)^2} \cos\left(k_c\Delta r\right) \frac{\sin\left(\frac{ak\Delta r}{c}\right)}{\Delta k\Delta r}\right) \]

### 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 \( \theta \) 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 \( f_k \),

\[ SPL_{f_kc} = SPL_{f_km} + 10 \log_{10} \left(1 - 10^{-\frac{\left(SPL_{f_km} - SPL_{f_kB}\right)}{10}}\right) \]

where \( SPL_{f_kc} \) is the background subtracted resultant SPL, \( SPL_{f_km} \) is the measured or data SPL value, and \( SPL_{f_kB} \) is the background SPL value.

For a signal to noise ratio \( \epsilon(f_k) \), where \( \epsilon(f_k) = SPL_{f_km} - SPL_{f_kB} \), 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/m², 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

Analog tape $\rightarrow$ Digital tape
  $\downarrow$
  1 Engineering units tape
  $\downarrow$
  2 OVIBSH
  $\downarrow$
  Time-shifted tape

Radar file $\rightarrow$ AAP $\rightarrow$ 3
  $\downarrow$
  Raw SPL tape
  $\downarrow$
  4 MANDATA
  $\downarrow$
  Averaged raw SPL

Weather $\rightarrow$ 5 FLYOVER
  $\downarrow$
  Corrected-average SPL
  $\downarrow$
  6 SPLTHTC
  $\downarrow$
  Directivity
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

$$t_r = \frac{(2 \cdot M \cdot \text{NPTS})}{SR}$$

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 OV1BSSH (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, $\text{NSHIFT}_i$, where

$$\text{NSHIFT}_i = \frac{\text{SR} \cdot d_i}{\overline{V}}$$

and $d_i$ is the distance from microphone $i$ to the reference microphone or reference position.
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 R) 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 \( \theta \). 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 \(( \text{pressure})^2\), \(P^2\), or power. More specifically, the average power \(P_a^2(f)\) at frequency \(f\) is

\[
P_a^2(f) = \frac{\sum_{i=1}^{N} \left( \text{SPL}_i(f) \times 10^{-3.0} \right)}{N}
\]

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

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., \(c(f_k) > 10 \text{ dB}\) 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 SPLHTC (Appendix E) determines peak value at a selected frequency band \( f \pm \Delta f \) where \( f \) is the frequency of interest and \( \Delta f \) 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 \( \theta \).

Program SPLHTC 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 SPLHTC 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,

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

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

\[ \text{NSHIFT}_i = \frac{(i-1)(30 \text{ ft}) \cdot 50000/\text{sec}}{200 \text{ ft/sec}} = 7500 (i-1) \text{ pts.} \]

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.
Figure 8. - Raw sound pressure levels

(a) Microphone 1
Figure 8. - Continued

(b) Microphone 2
Figure 8. - Continued

(c) Microphone 3
Figure 8. - Continued
Figure 8. - Continued

(e) Microphone 5
Figure 8. - Continued

(f) Microphone 6
Figure 8. - Continued
Figure 8. - Continued

(h) Microphone 8

Figure 8. - Continued
Figure 9. - Averaged sound pressure level
Figure 10. - Averaged and corrected sound pressure level
Figure 11. - Directivity pattern
**TABLE 1**

EXAMPLE OF INSTRUMENTATION CORRECTIONS ADDED TO MEASURED SOUND PRESSURE LEVELS

<table>
<thead>
<tr>
<th>FREQUENCY (kHz)</th>
<th>PRESSURE RESPONSE (dB)</th>
<th>DIFFRACTION (dB)</th>
<th>WINDSCREEN (dB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.0</td>
<td>0.0</td>
<td>+.1</td>
<td>0.</td>
</tr>
<tr>
<td>1.1</td>
<td>0.0</td>
<td>+.1</td>
<td>0.</td>
</tr>
<tr>
<td>1.2</td>
<td>0.0</td>
<td>+.1</td>
<td>0.</td>
</tr>
<tr>
<td>1.3</td>
<td>0.0</td>
<td>+.1</td>
<td>0.</td>
</tr>
<tr>
<td>1.4</td>
<td>0.0</td>
<td>+.1</td>
<td>0.</td>
</tr>
<tr>
<td>1.5</td>
<td>0.0</td>
<td>+.1</td>
<td>0.</td>
</tr>
<tr>
<td>1.6</td>
<td>0.0</td>
<td>+.1</td>
<td>0.</td>
</tr>
<tr>
<td>1.7</td>
<td>0.0</td>
<td>+.1</td>
<td>0.</td>
</tr>
<tr>
<td>1.8</td>
<td>0.0</td>
<td>+.1</td>
<td>0.</td>
</tr>
<tr>
<td>1.9</td>
<td>0.0</td>
<td>+.1</td>
<td>0.</td>
</tr>
<tr>
<td>2.0</td>
<td>0.0</td>
<td>+.1</td>
<td>0.</td>
</tr>
<tr>
<td>2.1</td>
<td>0.0</td>
<td>+.2</td>
<td>-0.3</td>
</tr>
<tr>
<td>2.2</td>
<td>0.0</td>
<td>+.2</td>
<td>-0.3</td>
</tr>
<tr>
<td>2.3</td>
<td>0.0</td>
<td>+.2</td>
<td>-0.3</td>
</tr>
<tr>
<td>2.4</td>
<td>0.0</td>
<td>+.2</td>
<td>0.0</td>
</tr>
<tr>
<td>2.5</td>
<td>0.0</td>
<td>+.2</td>
<td>0.0</td>
</tr>
<tr>
<td>2.6</td>
<td>0.0</td>
<td>+.2</td>
<td>-0.5</td>
</tr>
<tr>
<td>2.7</td>
<td>0.0</td>
<td>+.2</td>
<td>-0.5</td>
</tr>
<tr>
<td>2.8</td>
<td>0.0</td>
<td>+.2</td>
<td>-0.5</td>
</tr>
<tr>
<td>2.9</td>
<td>0.0</td>
<td>+.2</td>
<td>-0.5</td>
</tr>
<tr>
<td>3.0</td>
<td>0.0</td>
<td>+.2</td>
<td>-0.5</td>
</tr>
<tr>
<td>ALTITUDE (m)</td>
<td>TEMPERATURE (°C)</td>
<td>RELATIVE HUMIDITY (%)</td>
<td>PRESSURE (mbars)</td>
</tr>
<tr>
<td>-------------</td>
<td>------------------</td>
<td>-----------------------</td>
<td>-----------------</td>
</tr>
<tr>
<td>0</td>
<td>12.4</td>
<td>50.0</td>
<td>1013.0</td>
</tr>
<tr>
<td>10</td>
<td>12.2</td>
<td>49.9</td>
<td>1011.9</td>
</tr>
<tr>
<td>20</td>
<td>11.8</td>
<td>48.8</td>
<td>1010.7</td>
</tr>
<tr>
<td>30</td>
<td>11.5</td>
<td>49.0</td>
<td>1009.6</td>
</tr>
<tr>
<td>40</td>
<td>11.0</td>
<td>51.1</td>
<td>1008.4</td>
</tr>
<tr>
<td>50</td>
<td>10.6</td>
<td>52.0</td>
<td>1007.4</td>
</tr>
<tr>
<td>60</td>
<td>10.6</td>
<td>53.0</td>
<td>1006.2</td>
</tr>
<tr>
<td>70</td>
<td>10.6</td>
<td>53.0</td>
<td>1005.1</td>
</tr>
<tr>
<td>80</td>
<td>10.7</td>
<td>53.0</td>
<td>1003.9</td>
</tr>
<tr>
<td>90</td>
<td>10.7</td>
<td>54.0</td>
<td>1002.9</td>
</tr>
<tr>
<td>100</td>
<td>10.0</td>
<td>53.0</td>
<td>1001.7</td>
</tr>
</tbody>
</table>
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.
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
- $\Delta t_i$: time difference between microphone positions
- $d_i$: distance between microphone $i$ and reference microphone
- $\bar{V}$: average velocity of the aircraft
- $\theta$: directivity angle of acoustic source
- $M$: number of Fourier transforms averaged
- $N$: number of microphones
- $L$: $M \cdot N$ (total number of averages)
- $\text{PSD}_R$: resultant power spectral density
- $\text{PSD}_i$: power spectral density of microphone $i$
- $ND$: number of degrees of freedom
- $\varepsilon$: error of estimation
- $\varepsilon_r$: normalized random error
<table>
<thead>
<tr>
<th>Symbol</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>W</td>
<td>data window weighting factor</td>
</tr>
<tr>
<td>$f_s$</td>
<td>frequency of the source</td>
</tr>
<tr>
<td>$f_0$</td>
<td>frequency at the observer</td>
</tr>
<tr>
<td>$M_c$</td>
<td>Mach number of the source</td>
</tr>
<tr>
<td>$P_s$</td>
<td>pressure of the source</td>
</tr>
<tr>
<td>$P_0$</td>
<td>pressure at the observer</td>
</tr>
<tr>
<td>$n$</td>
<td>exponent in convective amplification equation</td>
</tr>
<tr>
<td>$\Delta SPL_1$</td>
<td>difference in sound pressure level due to convective amplification</td>
</tr>
<tr>
<td>$RE$</td>
<td>distance between acoustic source and observer</td>
</tr>
<tr>
<td>$\Delta SPL_2$</td>
<td>difference in sound pressure level due to the inverse square law</td>
</tr>
<tr>
<td>$\alpha$</td>
<td>absorption coefficient</td>
</tr>
<tr>
<td>$\alpha_{CL}$</td>
<td>classical absorption coefficient</td>
</tr>
<tr>
<td>$\alpha_{rot}$</td>
<td>rotational relaxation coefficient</td>
</tr>
<tr>
<td>$\alpha_{vib,N}$</td>
<td>vibrational relaxation of nitrogen coefficient</td>
</tr>
<tr>
<td>$\alpha_{vib,O}$</td>
<td>vibrational relaxation of oxygen coefficient</td>
</tr>
<tr>
<td>$T$</td>
<td>temperature</td>
</tr>
</tbody>
</table>
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 $f$

$\Delta SPL_3$  difference in sound pressure level due to atmospheric absorption

$r_i$  depth of layer $i$

G  ground factor

k  wave number

$\Delta r$  difference between direct and reflected path

$f_c$  central frequency

$\Delta SPL_4$  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 $f_k$
<table>
<thead>
<tr>
<th>Symbol</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>$c(f_k)$</td>
<td>signal-to-noise ratio at $f_k$</td>
</tr>
<tr>
<td>SR</td>
<td>sample rate</td>
</tr>
<tr>
<td>BW</td>
<td>bandwidth or frequency resolution</td>
</tr>
<tr>
<td>NPTS</td>
<td>number of spectral output points</td>
</tr>
<tr>
<td>$t_r$</td>
<td>time increment of assumed local stationarity</td>
</tr>
<tr>
<td>$\text{NSHIFT}_i$</td>
<td>number of points corresponding to $\Delta t_i$</td>
</tr>
<tr>
<td>$P_a^2(f)$</td>
<td>average power at frequency $f$</td>
</tr>
<tr>
<td>REF</td>
<td>reference for dB conversion</td>
</tr>
<tr>
<td>$\text{SPL}_a(f)$</td>
<td>average sound pressure level at frequency $f$</td>
</tr>
</tbody>
</table>
APPENDIX A. - Program to Shift Microphone Data

```
PROGRAM OVISBH(INPUT,OUTPUT,TAPE1,TAPE2,TAPE3,TAPE4,TAPE5,TAPE6,
                TAPE7,TAPE8,TAPE9,TAPE10,TAPE11,TAPE12,
                TAPE13,TAPE14,INPUT)

        OVISBH SHIFTS MICROPHONE DATA FROM OVID DATA. IF N IS THE
        MICROPHONE NUMBER EACH CHANNEL GETS SHIFTED FORWARD BY
        TAU((IHD-1)/VEL)SAMPLES POINTS WHERE D IS THE MICROPHONE
        SPACING, VEL IS THE AIRCRAFT'S SPEED AND SR IS THE SAMPLE RATE
        OF DIGITIZATION.
        (FOR RUN #2, TAU IS IN INCREMENTS OF 2575.)

        D. R. RIDLEY
        2/22/80

        OVISBH MAINTAINED BY SYSTEM DEVELOPMENT CORPORATION

        DIMENSION DATA(1875,12),NAMES(12),IUNITS(12),IHDR(8),IST(10)
        DIMENSION JDHDR(4)
        NAMELIST/INPUT/IST,NPTSS
        DATA IST:5,10,15,20,25,30,35,40,45,50,NPTSS/1875/
        JDHDR/04108H

        READ (14,INPUT)
        IF (EOF(14)) 202,203
        203 CONTINUE
        READ (14,B10) JHDR
        5010 FORMAT(10I0)
        IF (NPTSS .GT. 1875) STOP 'NPTSS CAN NOT BE GT 1875'

        READ FROM TAPE1 ALL CHANNELS AND PLACE ON FILES 2 THROUGH 12
        NPTSS IS 1/5 OF THE NO. OF POINTS TO SKIP
        IST IS THE ARRAY OF MULTIPLES OF NPTSS FOR EACH CHANNEL
        READ (1) ICH,NCHAN,(NAMES(I),I=1,NCHAN),(IUNITS(I),I=1,NCHAN),
        (IHDR(I),I=1,8)
        IF (EOF(1)) 6,9
        5 STOP 'RECHECK - TAPE1 IS INPUT TAPE - SHOULD HAVE 10 DATA CH'S'
        9 IF (NCHAN.GT.12) GO TO 5
        I=0 $1=0
        DO 10 K=1,NCHAN
             READ (1) (DATA(K,I),I=1,NCHAN)
             IF (EOF(1)) 6,10
             LS=K-1
             IF (LS .LT. 0) GO TO 21
             LI=1
             10 CONTINUE
             L=NPTSS
             IF (L.LT.0) GO TO 21
             LI=1
             10 CONTINUE
             L=NPTSS
             IF (L.LT.0) GO TO 21
             LI=1
             10 CONTINUE
             11 WRITE (2) (DATA(K,I),I=1,L)
             WRITE (2) (DATA(K,I),I=1,L)
             IF (I.EQ.NCHAN) 13,8
             12 CONTINUE
             13 WRITE (10,3) (DATA(K,I),I=1,L)
             WRITE (10,3) (DATA(K,I),I=1,L)
             IF (I.EQ.NCHAN) 16,8
             14 CONTINUE
             16 CONTINUE
```

45
CONTINUE
IF (L1.EQ.0) GO TO 1
DO 30 J=1,MCNAV
ENDFILE J
ENDFILE J
REWIND J
30 CONTINUE
CALL EVICT(I)
IF (JHDR(I).EQ.0) GO TO 31
DO 32 INT=1,4
JHDR(INT)=JHDR(INT)
31 CONTINUE
WRITE (12) ISH,MCNAV,(NAME(S,I),I=1,MCNAV), (UNIT(S,I),I=1,MCNAV),
1 ;(JHDR(I),I=1,8)
WRITE FILES 2 THROUGH 12 ON TAPE
WRITE (13) (MCHAM(I),I=1,MCNAV)
DO 108 I=1,L
IF (I.EQ.0) GO TO 100
WRITE (13) (DATA(I,J),J=1,MCNAV)
100 CONTINUE
END FILE 13 SENDFILE 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.
APPENDIX C. - Program for Ensemble Averaging

PROGRAM MANDATA(INPUT, OUTPUT, TAPE1, TAPE2, TAPE3, TAPE4, TAPE5, TAPE6, TAPE7, TAPE8, TAPE9, TAPE10)

PROGRAM MANDATA is a package for manipulating data from a TIFT file. It consists of basically two routines: one to stack files and one to perform the basic mathematical functions. Its creation was necessary for manipulating files created by the Acoustics Analysis Program (AAP) and the Signal Analysis Program (SAP). It may be used on any TIFT file, the stacking routine, however, is one more applicable to AAP and SAP created files.

DOREEN GRIDLEY
JANUARY 1980

MANDATA MAINTAINED BY SYSTEM DEVELOPMENT CORPORATION

### NAMELIST INPUTS ###

<table>
<thead>
<tr>
<th>VARIABLE</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>IOPT</td>
<td>INTEGER</td>
<td>0 - do not stack file 1 - stack all serial numbers 2 - stack selected serial numbers</td>
</tr>
<tr>
<td>JSN</td>
<td>ARRAY</td>
<td>SERIAL NUMBERS TO BE STACKED/USED WHEN IOPT=2, EXAMPLE: JSN=1,2,4, WILL BE STACKED IF JSN=5, A CONSTANT WILL BE WRitten TO CHANNEL 1.</td>
</tr>
<tr>
<td>CONST</td>
<td>REAL</td>
<td>CONSTANT TO BE EMPLOYED WHEN JSN=99</td>
</tr>
<tr>
<td>IFUNC</td>
<td>INTEGER</td>
<td>0 - no action 1 - add channels 2 - subtract channels 3 - multiply channels 4 - divide channels 5 - average channels 6 - chi-squared equivalence test EXAMPLE: IOPT=2, JSN=1,2,4, IFUNC=6, SERIALS 1, 2, AND 4 WILL BE STACKED AS CHANNELS 1, 2, AND 3 RESPECTIVELY, CHANNEL 3 WILL BE SUBTRACTED FROM CHANNEL 1 AS WILL CHANNEL 4, THEREFORE, THE RESULT=1-2-3</td>
</tr>
<tr>
<td>KSN</td>
<td>INTEGER</td>
<td>NEW SERIAL NUMBER OF FIRST FILE RETAINED (SUBSEQUENT FILES, IF ANY, WILL BE IN SEQUENTIAL ORDER FROM KSN)</td>
</tr>
<tr>
<td>ISPL</td>
<td>INTEGER</td>
<td>0 - input is not in db 1 - input IS in db</td>
</tr>
<tr>
<td>OSPL</td>
<td>INTEGER</td>
<td>0 - output not to be in db 1 - output to be in db</td>
</tr>
</tbody>
</table>
REF REAL 0.0002
IVAR INTEGER 0

CONVERSION FACTOR FOR ISPL AND OSPL

DIMENSION JAMES(101),JUNIT(101),I@N(101),DATA(101),NAMES(101),
REAL OSPL,JHDR(8),
DATA IOPT,JSN,CONF,IFUNC,KSN,ISPL,OSPL,REF/1,100//0,200,0,200,
0.00007,IVAR/0.

NAME ISPL,INPUT,IOPT,JSN,CONF,IFUNC,KSN,ISPL,OSPL,REF,IVAR

REAL JSN(100),IUNIT(101),HDR(8),
DATA IOPT,JSN,CONF,IFUNC,KSN,ISPL,OSPL,REF/1,100//0,200,0,200,
0.00007,IVAR/0.

NAME ISPL,INPUT,IOPT,JSN,CONF,IFUNC,KSN,ISPL,OSPL,REF,IVAR

READ NAMELIST INPUT

READ INPUT.

IF (EOF(5)) 9999,2

C THIS SECTION IS FOR STACKING SERIAL NUMBERS

C THAT CONTAIN ONE CHANNEL OTHER THAN TIME(OR FREQUENCY)

K=1

WRITE (6,input)

IF (IOPT,EO.0) GO TO 1000
N=1

100 READ (1) JSN,NCHAN,(NAMES(I),I=1,NCHAN),(IUNIT(1),I=1,NCHAN),

IF (EOF(1)) 900,110

110 CONTINUE

IJK=1

IF (IOPT,EO.1) GO TO 200

KHK=1

IF (KN,E0.0) GO TO 150

DO 120 I=1,KN

120 IF (JSN(EQ,JSN(I))) GO TO 120

GO TO 150

C IF NEEDED, CHECK TO SEE IF ISN IS A DESIRED SERIAL NUMBER

130 READ (1)

140 IF (EOF(1)) 160,150

150 IF (JSN(EQ,JSN(I))) GO TO 200

IF (JSN-EQ,JSN(I)) 120,200

200 K=K+1

C K IS THE NUMBER OF ISNS FOUND+1

IF (Kn,NE.2) GO TO 450

IF (JSN(EQ,JSN(I))) GO TO 300

DO 210 I=1,8

210 JAMES(I)-NAMES(I)

JUNIT(I)-IUNIT(I)

NAMES(I+1,0)

ENCODE (I,6000,JUNIT(I)) ISN

600 FORMAT (10)

DO 220 I=1,8

220 JHDR(I)-HDR(I)

250 READ (1,DATA(1),I=1,8)

IF (EOF(1)) 600,270

270 IJK=IJK+1

49
IF (IJK, EQ. 2) TIME = DATA(1)
IF (JSH(N), NE. 999) GO TO 400
300 DATA(1) = 0,
DATA(2) = CONST
400 JTO = 1
IF (MOD(K, 2), EQ. 0) JTO = 9
WRITE (JTO) (DATA(I), I=1, 2)
GO TO 250
450 IF (JSH(N), NE. 999) GO TO 500
JNAMES(K) = 10H 999
DATA = CONST
GO TO 511
500 ENCODE (10, 6001, JUNITS(K)) ISN
600 FORMAT (10)
JNAMES(K) = NAMES(2)
501 READ (1) (DATA(I), L=1, 2)
IF (EOF(1)) 600, 513
510 IJK = 1
DATA = DATA(I) + DATA(1) + DATA(1)
IF (JSH(N), NE. 2) GO TO 511
IF (DATA(I), EQ. TIME) GO TO 511
IF (JSH(N), NE. 999) GO TO 501
PRINT, 'SERIAL NUMBER ', ISN, ' TIMES DO NOT AGREE WITH PREVIOUS SERIAL'
STOP 'STACKING NON-COMPATABLE SERIALS'
511 JK = 1
512 IF (JSH(N), NE. 999) AND. IJK, EQ. 2) TIME = DATA(1)
JTO = 7
IF (MOD(K, 2), EQ. 0) JTO = 8
JTO = 7
READ (JTO) (DATA(I), L=1, JK)
IF (EOF(JTO)) 600, 520
DATA = DATA(I) + DATA(I) + DATA(I) + DATA(I)
WRITE (JTO) (DATA(I), L=1, JK)
IF (JSH(N), NE. 999) GO TO 501
DATA = CONST
GO TO 518
ENDFILE ITO
ENDFILE ITO
IF (K, NE. 2) REWIND JTO
REWIND ITO
H = 1
IF (IOP(1), EQ. 1) GO TO 100
IF (JSH(N), NE. 0) GO TO 100
670 DO 680 I = 1, K
JUNITS(I) = JUNITS(I) + 1
NAMES(I) = NAMES(I)
680 DO 675 I = 1, K
JHDR(I) = JHDR(I)
675 DO 680 I = 1, K
KSH = KSH + 1
NCH = X
KTO = 4
WRITE (KTO) KSH, NCHNAM, (NAMES(I), I=1, NCHNAM), (JUNITS(I), I=1, NCHNAM),
(NHDR(I), I=1, 2)
700 IF (EOF(KTO), 710, 720
710 ENDFILE KTO
PRINT, 'VALUES ARE STACKED ON TAPE'
ACQUIRE INTO
REWIND INPUT
200 IF (EOF(S)) GOTO 201
100 IF (EOF(M)) STOP 'TAPE 4 IS EMPTY'
300 REWIND M
400 IF (IJKL.EQ.1) CALL FDBCON(DATA,NCH,REF) #IJ=IJ+1
500 STOP 'IFUNC CODE NOT DEFINED PROPERLY'
600 ADDING CHANNELS A=B+C+D+...
700 DO 1125 I=1,NCH
800 NDATA=DATA(1)
900 CONTINUE
1000 SUBTRACTING CHANNELS A-B-C-D-E-...
1100 DO 1225 I=1,NCH
1200 NDATA=DATA(I)
1300 CONTINUE
1400 MULTIPLYING CHANNELS A*B*C*D*E*...
1500 DO 1325 I=1,NCH
1600 NDATA=DATA(I)
1700 CONTINUE
GO TO 1600

DIVIDING CHANNELS

1400 NAMES(2)=10H4UID CHS
ADATA+DATA(2)
DO 1425 I=1,NCH
ADATA+DATA(/DATA(I))
CONTINUE
GO TO 1600

AVERAGING CHANNELS

1500 NAMES(2)=10H4UID CHS
SUM=0
DO I=1,2,NCH
SUM=SUM+DATA(I)
CONTINUE
ADATA=SUM/(NCH-1)

CHI-SQUARED EQUIVALENCE TEST

1600 IF (I.EQ.1) SDATA=0
SDATA=SDATA+ALOG10(ABS(/DATA(2))/DATA(3)))**2
GO TO 1070

CHI-SQUARE

CHISQ-IJISDATA/1.CF-FLOAT(IJ)
PRINTS, 'DF-',DF,'°',DATA, 'CHISO-',CHISO
CALL MDCH(CHISO,DF,P,IER)
IF (IER.EQ.32 .OR. IER.EQ.34) PRINTS, * ERROR IN USING CHI'
P(+(+)=I1300.
WRITE (6,5000) E14.6
5000 CONTINUE
5000 P=1.
5001 FORMAT(1X,15X, 'THE PROBABILITY THAT CHANNEL $, AND CHANNEL $,
5002 END
.A18..EX.. ARE STATISTICALLY EQUIVALENT IF .F10..5.. PERCENT.)
END
SUBROUTINE FDBCON(DATA, NCH, REF)
DIMENSION DATA(101)
DO 1 I=1,NCH
  DATA(I)=(10.**(A(I)/10.))*REF*REF
1 CONTINUE
RETURN
END
SUBROUTINE TDBCON(DAT, REF)
IF (DAT).LT.1
  1 CONTINUE
RETURN
END
APPENDIX D. - Program for Correcting Sound Pressure Levels

PROGRAM FLYOVER: INPUT, OUTPUT, TAPE1 INPUT, TAPE6 OUTPUT,
                  TAPE1, TAPE2, TAPE10

PROGRAM FLYOVER IS TO CORRECT NOISE FLYOVER SOUND PRESSURE
LEVEL DATA GENERATED BY THE ACOUSTICS ANALYSIS PROGRAM (AAP).
THE PROPAGATION PATH CORRECTIONS ARE:

1) INSTRUMENTATION CORRECTIONS--PRESSURE RESPONSE,
   DIFFRACTION, AND WINDSCREEN CORRECTIONS
2) CONVECTIVE AMPLIFICATION
3) INVERSE SQUARE LAW
4) ATMOSPHERIC ABSORPTION
5) GROUND IMPEDANCE
6) DOPPLER FREQUENCY SHIFT

FILES:
1) INPUT FILE IS TAPE1
2) OUTPUT FILE FOR CORRECTED SPL’S IS TAPE10

FLYOUER MAINTAINED BY SYSTEM DEVELOPMENT CORPORATION

THE FIRST INPUT EXPECTED IS A SIX (6) DIGIT CODE BEGINNING
IN COLUMN ONE (1). THE ORDER OF THE CODE IS THE ORDER OF
THE ABOVE CORRECTIONS.
0 IMPLIES DO NOT CORRECT
1 IMPLIES CORRECT

EX.1: 000000 NO CORRECTIONS
     101000 CORRECT FOR INSTRUMENTATION AND INV. SQ.

A NAMELIST #INPUTS IS EXPECTED NEXT.

NAMELIST INPUTS

VARIABLE Type AND Description

* * * * * * *  *  *  * * * * *

ICODE INTEGER   FLAG FOR WHICH THETAS ON TAPE1
0=ALL THETAS ON TAPE1
1=SELECTED THETAS ACCORDING
   TO ARRAY THTA

THTA REAL ARRAY 10x0.0   ARRAY OF THETAS TO BE CONSIDERED WHEN
                    ICODE=1.
                    NOTE: THETA ON FILE TAPE1 WILL BE
                    ACCEPTED IF IT IS WITHIN +.1
                    OR -1.1 OF THTA

N INTEGER 0   FLAG FOR TYPE OF ACOUSTIC SOURCE
DISTRIBUTION (FOR CONV. AMP.,)
0=M0NPOLE
1=DIPOLE
2=QUADROPOLE

STCR REAL ARRAY 80x0.995  ARRAY OF VALUES FOR INSTRUMENTATION
                      CORRECTIONS
                      STCR(1,1) ARE THE FREQUENCIES
                      STCR(1,2) ARE DTSPL FOR P. RESPONSE
                      STCR(1,3) ARE DTSPL FOR DIFFRACTION
                      STCR(1,4) ARE DTSPL FOR WINDSCREEN

54
HGT  REAL  30.  DISTANCE FROM GROUND TO MICROPHONE (FEET)
REF  REAL  15.  REFERENCE DISTANCE FOR INVERSE SQUARE LAW CORRECTION
P   REAL ARRAY  251-999.  MEASURED ATMOSPHERIC PRESSURE (MBARS) CORRESPONDING TO ARRAY ALT
T   REAL ARRAY  251-999.  MEASURED ATMOSPHERIC TEMPERATURE (DEG C) CORRESPONDING TO ARRAY ALT
    NOTE: THE TEMPERATURE ARRAY MUST BE INPUT WITH EACH SUCCESSIVE NAMELIST IF APPLYING CORRECTIONS 2, 4, 5, OR 6
ALT  REAL ARRAY  251-999.  MEASURED ALTITUDE ( METERS) INPUT IS TO BE IN DECREASING ORDER AND IN EQUAL INCREMENTS
RH  REAL ARRAY  251-999.  MEASURED RELATIVE HUMIDITY (PERCENT) CORRESPONDING TO ARRAY ALT
IUXR INTEGER  0  FLAG TO PRINT OUT RESULTS OF SPL AFTER EACH CORRECTION IS MADE
0-DO NOT PRINT
1-PRINT
F1  REAL  6.0  INITIAL FREQUENCY FOR CORRECTIONS TO BE APPLIED
F2  REAL  50000.  FINAL FREQUENCY FOR CORRECTIONS TO BE APPLIED
SR  REAL  50000.  SAMPLE RATE OF DIGITIZATION
NPTS INTEGER  512  NUMBER OF POINTS PER BLOCK (VARIABLE READ IN AAP)
NBLKS INTEGER  5  NUMBER OF BLOCKS FOR THE SPL USED IN AAP
NMICS INTEGER  8  NUMBER OF MICROPHONES AVERAGED IN MANDATE AFTER AAP
ITAPE INTEGER  0  FLAG FOR WRITING (AND PRINTING) ONLY THOSE FREQUENCIES BETWEEN F1 AND F2
0-WRITE ALL FREQUENCIES
1-WRITE FREQUENCIES BETWEEN F1 AND F2
IPRINT INTEGER  0  FLAG FOR PRINTING WHEN IUXR=1
0-PRINT DELTA SPL VALUES
1-PRINT CORRESPONDING SPL VALUES

FOR A. MUELLER
D. GRIDLEY

55
(This is a revised version of a program developed 5-80)

DIMENSION THTA(10), STRCR(200,4), P(25), T(25), ALT(25), RH(25), IFNC(16)

NAMELIST /INPUT/ I CODE, THTA, N,STRCR, MGT, REF, P, T, ALT, RH

DATA I CODE, THTA, N, STRCR/0, 10X0.0, 0, 8008-999./

NAMELIST /INPUT/ MGT, REF, P, T, ALT, RH

DATA MGT, REF, P, T, ALT, RH/100*-999./

DATA (2), HAMS/2,5, (IUNITS,R), IHD9(8)

NAMELIST /INPUT/ I CODE, THTA, N, STRCR, MGT, REF, P, T, ALT, RH,

1 IFNC(6), DATA(I), HAMS/2,5, (IUNITS,R), IHD9(8)

READ ARRAY IFNC FROM INPUT

READ (6, 5111) (IFNC(I), I=1,6)

IF (EOF(6)) 1000, 2

READ NAMELIST $INPUT

READ (5, INPUT)

IF (EOF(5)) 1000, 4

WRITE INPUT INFORMATION TO OUTPUT FILE

WRITE (6, 5222)

5222 FORMAT(IHI, //, SX, *IFNC = CODE FOR CORRECTIONS (0 MEANS NOT APPLIED:

1. A MEANS APPLIED), /, SX, *IN THIS ORDER: /, 10X,

2. INSTRUMENTATION CORRECTIONS, /, 10X,

3. CONVECTIVE AMPLIFICATIONS, /, 10X,

4. INVERSE SQUARE LAW, /, 10X, ATMOSPHERIC ABSORPTION, /, 10X,

5. GROUND IMPEDANCE, /, 10X, DOPPLER FREQUENCY SHIFT, /)

WRITE (5, 5333) (IFNC(I), I=1,6)

5333 FORMAT(IOX, *IFNC = 8, 611)

WRITE (6, INPUT)

READ HEADER RECORD, AND TR, RE, THETA, X, Y, Z, AND V (THE FIRST 7

DATA RECORDS TO DETERMINE IF THE FILE IS ONE REQUESTED BY I CODE

AND THTA

READ (1) ISM, M0WI, (NAMS(I), I=1,NCHAN), (IUNITS(I), I=1, NCHAN),

1 (IHDR(I), I=1,8)

IF (EOF(1)) 900, 20

DD READ (1) (DATA(I), I=1, NCHAN)

IF (EOF(1)) 900, 20

STOP 'END OF FILE BEFORE RADAR INFORMATION IS READ'

READ (1) I CODE, NCHAN, (NAMES(I), I=1, NCHAN), (IUNITS(I), I=1, NCHAN),

1 (IHDR(I), I=1,8)

IF (EOF(1)) 900, 20

DD READ (1) (DATA(I), I=1, NCHAN)

IF (EOF(1)) 30, 40

STOP 'END OF FILE BEFORE RADAR INFORMATION IS READ'

READ (1) I CODE, NCHAN, (NAMES(I), I=1, NCHAN), (IUNITS(I), I=1, NCHAN),

1 (IHDR(I), I=1,8)

IF (EOF(1)) 900, 20

DD READ (1) (DATA(I), I=1, NCHAN)

IF (EOF(1)) 30, 40

STOP 'END OF FILE BEFORE RADAR INFORMATION IS READ'

READ (1) I CODE, NCHAN, (NAMES(I), I=1, NCHAN), (IUNITS(I), I=1, NCHAN),

1 (IHDR(I), I=1,8)

IF (EOF(1)) 900, 20

DD READ (1) (DATA(I), I=1, NCHAN)

IF (EOF(1)) 30, 40

STOP 'END OF FILE BEFORE RADAR INFORMATION IS READ'

READ (1) I CODE, NCHAN, (NAMES(I), I=1, NCHAN), (IUNITS(I), I=1, NCHAN),

1 (IHDR(I), I=1,8)

IF (EOF(1)) 900, 20

DD READ (1) (DATA(I), I=1, NCHAN)

IF (EOF(1)) 30, 40

STOP 'END OF FILE BEFORE RADAR INFORMATION IS READ'

READ (1) I CODE, NCHAN, (NAMES(I), I=1, NCHAN), (IUNITS(I), I=1, NCHAN),

1 (IHDR(I), I=1,8)

IF (EOF(1)) 900, 20

DD READ (1) (DATA(I), I=1, NCHAN)

IF (EOF(1)) 30, 40

STOP 'END OF FILE BEFORE RADAR INFORMATION IS READ'
IF (IJ .EQ. 5) UEL=DATA(2)
IF (IJ .EQ. 6) ZE=DATA(2)
IF (IJ .EQ. 7) VREL=DATA(2)
IF (IJ .NE. 7) GO TO 25
100 IF (ICODE .EQ. 0) GO TO 120
DO 110 1=1,10
   IF (THETA .GE. (THTA(I)-0.1).AND. THETA .LE. (THTA(I)+0.1))
      CALL SKIPFF(DUM,SLTAPE1,1)
   110 CONTINUE
   IF (I .NE. 7) GO TO 120
   110 CONTINUE
   IF (I .EQ. 7) YRAD=DATA(R)
   IF (I .NE. 7) GO TO 25
   120 IF (ICODE .EQ. 0) GO TO 120
   DO 135 I=1,25
      IF (T(I) .NE. -999.) T(I)=T(I)+273.15
   135 CONTINUE
   ADD 273.15 TO TEMP ARRAY
   WRITE HEADER RECORD AND RADAR INFORMATION TO TAPE10
   WRITE (10) ISN,NCHAN,(NAMES(I),I=1,NCHAN),(IUNITS(I),I=1,NCHAN),
             (IHDR(I),I=1,8)
   DUM=-993.
   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
   ADD 273.15 TO TEMP ARRAY
   DO 135 I=1,25
      IF (T(I) .NE. -999.) T(I)=T(I)+273.15
   135 CONTINUE
   COMPUTE AVERAGE TEMPERATURE FOR AVERAGE C FOR DF, AND CA
   AT=0.
   ICHT=0
   140 I=1
   IF (T(I) .EQ. -999.) OR. (I .GT. 25) GO TO 150
   IF (ALT(I) .LT. (HTS*3.3048)) OR. (ALT(I) .GT. (ZE*3.3048))
      IF (ALT(I) .LT. (HTS*3.3048)) OR. (ALT(I) .GT. (ZE*3.3048))
      I=1
      AT=AT + T(I)
      ICHT=ICTH+1
   150 AT=AT/ICH
   TA=AT/FLOAT(ICH)
   CM=(343.2*3048*(IHT))/(2.3048)
   COMPUTE DELTA HEIGHT IN ALTITUDE
   DMGT=ALT(I)-ALT(I-1)
   180 CONTINUE
ORIGINAL PAGE 13
OF POOR QUALITY

COMPUTE C CLOSEST TO GROUND FOR GROUND IMPEDANCE

ALTM=999999,
DO 180 I=1,25
IF (ALT(I) .EQ. -9999) GO TO 170
IF (ALT(I) .GE. ALTM) GO TO 160
ALTM=ALT(I)
ID=I
180 CONTINUE
170 CGI=(333.333*SORT(ID)/10)/0.3048

COMPUTE VARIABLES FOR CORRECTIONS

C ORR IS FOR THE DOPPLER FREQUENCY SHIFT
ANCH IS THE AVERAGE MACH NUMBER
DTSPL IS FOR THE INVERSE SQUARE LAW
DESPS IS FOR CONVOLUTION AMPLIFICATION
DK IS A NUMBER BASED ON RANGEWIDTH
DR IS THE PATH DIFFERENCE BETWEEN REFLECTED AND
DIRECT PATH
NDK IS THE PRODUCT OF DK AND DR

180 AMCH-VEL/CAVG
CORR=(1-AMCH*COS(THEAP1/180.))/
DTSPL-LOG10(REF/RE)
IF (IFNC(3) .EQ. 0) DESPL=0.0
IF (IFNC(2) .EQ. 0) DESPL=0.0
DXX=(PI45)/FLO(NTPS)*CG1
CH-1.09*(THEAP1/180.)

OLD METHOD OF CALCULATING DR-INVALID DUE TO CHOICE OF ANGLE

DR=SOR(T(RE*4.428)*HGT)-RE
DKDR-DR
IF (DKDR .EQ. 0) UI=1.0
IF (DKDR .EQ. 0) UI=1.0

WRITE TABLE HEADINGS ON OUTPUT IF IVAR=1
IF (IVAR NE. 1) GO TO 133
WRITE (6,5966) (IHDR(L),L-1,8)
6966 FORMAT (1H1,///,8010)
WRITE (6,5444) TR,RE,THETA,XRAD,VEL,ZRAD

5444 FORMAT(///,5X,T*,E12.5,4X,MR*,E12.5,4X,TETA*,E12.5,4X,
$
$


3CODE FOR CORRECTIONS IN PE: CARL, INS-ELEVATION OZ,.,15X,
4E20-C10NAL AMPLIFICATION,./,15X,DSOR-PSuez SQUARE LAWS./,15X,
5K5-AMTHTHERMIC ATTENTION,./,15X,NDI-GROUND IMPEDANCE,./,15X,
DGDF-DOPPLER FREQUENCY SHIFTS

IF (IPRINT NE. 0) WRITE (6,5666)
5666 FORMAT(///,4X,*FREQUENCY),
$15(4X,DAVG SPL DB),5*X, FREQUENCY,4X,25X.(IC)*,7X,(IC)*,7X,
$25*(IC)*,15(4X,*F,10,C15,15,R,15,A),1X,ALL 15,5*X,DF,10,///)

IF (IPRINT .EQ. 0) WRITE (6,5777)
5777 FORMAT(///,4X,*FREQUENCY),
$15(4X,DAVG SPL DB),5*X, FREQUENCY,4X,25X.(IC)*,7X,(IC)*,7X,
$25*(IC)*,15(4X,*F,10,C15,15,R,15,A),1X,ALL 15,5*X,DF,10,///)
READ AND APPLY REQUESTED CORRECTIONS

READ (1) F,SPL
IF (EOF(1)) G600,210
210 IF (F .LT. F1 .AND. F .LE. F2) GO TO 215
IF (ITAPE .NE. 0) GO TO 200
PPPL+SPL+2PL+SPL+5PL
IF (UNAV .NE. 1) GO TO 456
215 IF (IFNC(1) .EQ. 0) GO TO 300
INSTRUMENTATION CORRECTIONS

200 READ (1) .I., SP1
IF (F .GT. STRCR(I,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+1
FP abstraction(I)
DB1 abstraction(I,2)
DB2 abstraction(I,3)
DB3 abstraction(I,4)
260 CONTINUE
270 IND=IND+1
RATIO=(F/FF)/(STRCR(IND,1)-FF)
DB1 = DB1 + RATIO*(STRCR(IND,2)-DB1) * DB1+INT(DB1*10.+.5)/10.
DB2 = DB2 + RATIO*(STRCR(IND,3)-DB2) * DB2+INT(DB2*10.+.5)/10.
DB3 = DB3 + RATIO*(STRCR(IND,4)-DB3) * DB3+INT(DB3*10.+.5)/10.
280 DBSPL=DB1+DB2+DB3
SPL=SPL+DBSPL
300 IF (IFNC(2) .EQ. 0) GO TO 310
CONVECTIVE AMPLIFICATION
310 IF (IFNC(3) .EQ. 0) GO TO 320
INVERSE SQUARE LAW
320 IF (IFNC(4) .EQ. 0) GO TO 400
ATMOSPHERIC ABSORPTION

350 DO 350 I=1,200
IF (ALT(I) .EQ. -999.) GO TO 350
IF ((ALT(I) .LT. (HGT1*3.048)) .OR. (ALT(I) .GT. (2E1.3048))) GO TO 350
C-343.233*SQRT(T(I)/T0)**108./2.5418.*
SATX=10.7568/(1.-273.16/T(I))**-5.08969*ALC018(T(I)/273.16)
1 + 21.647*80.08181+10.884*8.29898*(T(I)/273.16)-1.)
59
original content of poor quality
APPENDIX E. - Program for Acoustic Directivities

PROGRAM SPLHTC(INPUT, OUTPUT, TAPE3-INPUT, TAPE4-OUTPUT,  
TAPE1, TAPE2, TAPE9, TAPE10)

PROGRAM SPLHTC is a program which generates the  
radiation pattern (SPL versus THETA) at a desired  
frequency for all files on TAPE1 which are generated  
by the Acoustic Analysis Program(AAP) with NO files  
OVARP3, OVARP4, and OVARP5 applied.

F.R.R. POLY FORWARD SPEED EFFECTS
(P. MUELLER)
D. GRIDLEY (1981):

SPLHTC MAINTAINED BY SYSTEM DEVELOPMENT CORPORATION

I. FILES
A. TAPE1 IS THE SFL FILE GENERATED BY AAP- THE  
FIRST 7 DATA RECORDS ARE RADAR INFORMATION
B. TAPE2 IS THE RADIATION PATTERN FILE FOR THE  
BACKGROUND RUN ONLY USED AS INPUT WHEN  
ISSUB=1
C. TAPE3 IS THE NEW RADIATION PATTERN FILE
D. TAPE4 IS THE NEW RADIATION PATTERN MINUS THE  
BACKGROUND (TAPE2) (ONLY OUTPUT WHEN ISSUB=1)

II. NAMELIST #NAMES
A. FREO IS THE DESIRED FREQUENCY AT WHICH TO GENERATE  
THE RADIATION PATTERN.
B. ERR IS THE BANDWIDTH (MHz) WHICH TO FIND SPL VALUE  
(RANGE OF FREO-ERR; FOR+ERR IS USED)
C. ISSUB IS THE FLAG FOR SUBTRACTING BACKGROUND  
(0=DO NOT SUBTRACT 1=SUBTRACT)
D. ISUMTPY IS THE FLAG FOR COMPUTATIONS  
(0=HIGHEST VALUE IN RANGE ONLY, 1+SUM OF TWO  
HIGHEST SPL VALUES IN THE RANGE)

DIMENSION NAMES(2), JNAMES(2), JUNITS(2),  
DATA(2), THETA(5), THETAB,  
NAMELIST /NAMES/FREO, ERR, ISSUB, ISUMTPY  
DATA ERR=<200, FREO=20000, ISSUB=0, ISUMTPY=0,  
JNAMES(1)=THETA, JNAMES(2)=SPL,  
JUNITS(1)=DEGREES, JUNITS(2)=SPL  
READ 5, NAME
10 READ (5, NAME)
      IF (EOF(5)) 1000, 5
      WRITE (6, NAME)
      IF (EOF(1)) 99999, 6
      READ (7) JNAME(1), JNAME(2), JUNITS(1), JUNITS(2),  
      THETA, THETAB, ...  
      20 READ (5, NAME)
      IF (EOF(5)) 99999, 6
      WRITE (6, NAME)

10 READ (1) ISN, NCHAN, (JNAMES(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN),  
       (JUNITS(1), I—1, NCHAN), (JNAMES(1), I—1, NCHAN),  
       (JNAMES(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN),  
       (JNAMES(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN),  
       (JNAMES(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN),  
       (JNAMES(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN),  
       (JNAMES(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN),  
       (JNAMES(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN),  
       (JNAMES(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN),  
       (JNAMES(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN),  
       (JNAMES(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN),  
       (JNAMES(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN),  
       (JNAMES(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN),  
       (JNAMES(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN),  
       (JNAMES(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN),  
       (JNAMES(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN), (JUNITS(1), I—1, NCHAN),  
61
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 39

FIND MAXIMUM SPL VALUE IN RANGE
IF (ISUMTYP .EQ. 0) GO TO 55
IF (DATA(2) .LT. DBMAX) GO TO 30
IF (DATA(2) .LT. DBMAX) GO TO 52
DBMAX=DATA(2)
GO TO 30
52 DBMAX=DATA(2)
GO TO 30
55 IF (DATA(2) .LT. DBMAX) GO TO 30
60 IF (ISUMTYP .EQ. 0) GO TO 65
DBMAX=10.**DBMAX/20.11.*.0002
DBMAX=10.**DBMAX/20.11.*.0002
DBMAX=10.**DBMAX/20.11.*.0002
DBMAX=10.**DBMAX/20.11.*.0002
65 WRITE (9) THETA, DBMAX
READ (1)
IF (EOF(1)) 10, 70
70 CALL SKIPF(SLTAPE1,1)
GO TO 10
500 ENDFILE 9
REUND 1
IF (IBSUB .EQ. 0) GO TO 1
ENDFILE 9
REUND 9
REUND 2

PLACE BKG INTO ARRAY THETA AND DBB
510 READ (2) ISN
IF (EOF(2)) 520, 530
520 STOP 'NO RADAR FILE INFO.'
530 J=0
540 IF (J=J+1) READ (2) THETA(1), DB(1)
IF (EOF(2)) 550, 540
550 KNT+J=1

READ SPL VS THETA AND SUBTRACT BKG--ONLY THOSE THETAS WITH A CORRESPONDING THETAD WILL BE WRITTEN TO TAPE1B.
600 READ (9) ISN, NCHAN, (NAMES(I), I=1, NCHAN), (IUNITS(I), I=1, NCHAN),
(INDR(I), I=1, NCHAN),
IF (EOF(9)) 1000, 610
610 WRITE (10) ISN, NCHAN, (NAMES(I), I=1, NCHAN), (IUNITS(I), I=1, NCHAN),
(INDR(I), I=1, NCHAN),
620 READ (9) THETA, DB
IF (EOF(9)) 700, 630
630 L=0
640 IF (L .GT. KNT) GO TO 620
IF (THETAD(L) .GT. (THETA+1.) .OR. THETAD(L) .LT. (THETA-1.))
1 GO TO 640
650 CONTINUE
IF((DB-DDB(L)) .GT. 3.) GO TO 660
WRITE (6,5660) ISH, THETA
5660 FORMAT(/S FOR SERIAL #,16,E5/N 16 ( OR = 3 FOR ANGLE #,F10.4)
IF (DB-DDB(L) .LE. 0) GO TO 665
GO TO 660
660 IF (DB-DDB(L) .GE. 10.) GO TO 670
665 DB=DB+10. PLOG10(1-10 EXP ((DB(L)-DB)/10.)
670 WRITE (10) THETA, DB
GO TO 680
700 ENDFILE 10
ENDFILE 10
GO TO 1
1000 STOP "END OF PROGRAM"
END
REFERENCES


