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

DOREEN GRIDLEY

KENTRON INTERNATIONAL
HAMPTON, VIRGINIA 23666

CONTRACT NAS1-16000

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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 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 \((X, Y, Z)\) 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, that 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, \( \alpha \), 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 \( L = MN \). The final power spectral density \( PSD_R \) may be expressed as

\[
PSD_R = \frac{N}{L} \sum_{i=1}^{N} PSD_i
\]

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),

\[ ND = 2L. \]
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 \( \varepsilon \) of estimation is

\[
\varepsilon = \frac{1}{\sqrt{L}} \sqrt{\frac{2}{N\Delta}}
\]

This states that the error \( \varepsilon \) 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 ±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 \( \varepsilon_r \) 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 Fourier Transform. $\epsilon_p$ is a measure of the convergence of mean spectra. If $\epsilon_p$ 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 $\bar{V}$, 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 ($M_c = \bar{V}/c$ 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.

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The expression \( 1 - \mu \cos \theta \) is the same as was defined in the Doppler effect. The difference in sound pressure level between static and flight cases \( \Delta SPL_1 \) is

\[
\Delta SPL_1 = 20(2n+2) \log_{10} (1 - \mu \cos \theta)
\]

### 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 \( \Delta SPL_2 \) may then be expressed as

\[
\Delta SPL_2 = -20(\log_{10} \left( \frac{r_1}{r_2} \right))
\]

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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 $\alpha_{\text{CL}}$, rotational relaxation $\alpha_{\text{rot}}$, and vibrational relaxation of nitrogen and oxygen, $\alpha_{\text{vib},N}$ and $\alpha_{\text{vib},O}$ respectively, or

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

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/m² by

\[
\log_{10}(P_{\text{sat}}/P_0) = 10.79586 [1 - (T_{01}/T)]
-5.02808 \log_{10}(T/T_{01}) + 1.50474 \times 10^{-4}(1-10^{-8.29692[(T/T_{01})-1]})
+0.42873 \times 10^{-3}(10^7 76958[1-(T_{01}/T)]-1) -2.2195983
\]

where \( P_0 \) = reference pressure of 1.013 x 10⁵ N m²

\( T_{01} = 273.16°K \)

2) Calculate the absolute humidity \( H \) in %

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

3) Calculate the relaxation frequency of oxygen and nitrogen, \( f_r,0 \) and \( f_r,N \) by

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

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

where \( T_0 \) = reference temperature of 293.15°K.

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

\[
a(f) = 8.686(T/T_0)^{1/2} \left[ \frac{f^2}{(P/P_0)} \right] \times
(1.84 \times 10^{-11} + 2.19 \times 10^{-4}(T/T_0)^{-1}(P/P_0)(2239/T)^2 x \exp \left( -2239/T \right) \left[ f_{r,0} + \left( f^2/f_{r,0} \right) \right]
\]
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 (k) the attenuation of the sound wave in terms of dB, \( \Delta \text{SPL}_3 \), is

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

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

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

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
\[ \Delta \Gamma \approx \sqrt{\text{RE} \cdot \text{RE} + 4Zh} - \text{RE} \]
if \( h \ll Z \) and \( h \ll \text{RE} \)

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

where \( \Delta k = \frac{2\pi}{c} \cdot \frac{\Delta f}{2} \) (where \( \Delta f \) 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 \( \Delta S P L_b \) is

\[ \Delta S P L_b = -10 \log (2 + 2 e^{-(\Delta k \Delta r)^2} \cos(k_c \Delta r) \frac{\sin(\Delta k \Delta r)}{\Delta k \Delta r}) \]

### 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 \( f_k \),

\[
S P L_{f k c} = S P L_{f k m} + 10 \log_{10} \left( 1 - 10^{\frac{S P L_{f k B} - S P L_{f k m}}{10}} \right)
\]

where \( S P L_{f k c} \) is the background subtracted resultant SPL, \( S P L_{f k m} \) is the measured or data SPL value, and \( S P L_{f k B} \) is the background SPL value.

For a signal to noise ratio \( \epsilon(f_k) \), where \( \epsilon(f_k) = S P L_{f k m} - S P L_{f k B} \), 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

\[ BW = \frac{SR}{2 \times NPTS} \]

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

1. Engineering units tape
2. OVIBSH
   Time-shifted tape

Radar file → AAP

3. Raw SPL tape

4. MANDATA
   Averaged raw SPL

Weather → FLYOVER

5. Corrected-average SPL

6. SPLTHTC
   Directivity

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

\[
t_r = \frac{(2 \cdot M \cdot NPTS)}{SR}
\]

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 OVIBSH (Appendix A) by matrix manipulation (Step 2 of Figure 6). The velocity \( \bar{V} \) is extracted from radar information and used to calculate the number of points each microphone is to be shifted, \( N_{SHIFT_i} \), where

\[
N_{SHIFT_i} = \frac{SR \cdot d_i}{\bar{V}}
\]

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

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

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
\[ \text{SPL}_a(f) = 10 \log_{10} \left( \frac{P_a(f)}{\text{REF}^2} \right) \]

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., \( \epsilon(f_k) > 10 \text{ dB} \) 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 SPLTHTC (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 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 50,000 samples per second. This sample rate was chosen to meet the criteria presented in Section 5.1. The maximum frequency of interest was 20,000 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,

\[ BW = \frac{50,000}{2(256)} = 97.656 \text{ Hertz} \]

and

\[ t_r = \frac{2 \times 5 \times 256}{50,000} = 0.0512 \text{ sec.} \]

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

\[ NSHIFT_i = \frac{(i-1)(30 \text{ ft}) \cdot 50,000/\text{sec}}{200 \text{ ft/sec}} = 7500 (i-1) \text{ pts.} \]

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
Figure 8. - Continued
Figure 8. - Continued

(c) Microphone 3
Figure 8. - Continued

(d) Microphone 4
Figure 8. - Continued

(e) Microphone 5
(f) Microphone 6

Figure 8. - Continued
(g) Microphone 7

Figure 8. - Continued
Figure 8. - Continued

(h) Microphone 8
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>
</tbody>
</table>
# Table 2

Example of weather data necessary for propagation correction inputs

<table>
<thead>
<tr>
<th>Altitude (m)</th>
<th>Temperature (°C)</th>
<th>Relative Humidity (%)</th>
<th>Pressure (mbars)</th>
</tr>
</thead>
<tbody>
<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>49.8</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
- $V$: 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 $i$
- $N_D$: number of degrees of freedom
- $\epsilon$: error of estimation
- $\epsilon_r$: normalized random error
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

\( \Delta SPL_1 \)  difference in sound pressure level due to convective amplification

\( RE \)  distance between acoustic source and observer

\( \Delta SPL_2 \)  difference in sound pressure level due to the inverse square law

\( \alpha \)  absorption coefficient

\( \alpha_{\text{CL}} \)  classical absorption coefficient

\( \alpha_{\text{rot}} \)  rotational relaxation coefficient

\( \alpha_{\text{vib,N}} \)  vibrational relaxation of nitrogen coefficient

\( \alpha_{\text{vib,O}} \)  vibrational relaxation of oxygen coefficient

\( T \)  temperature

42
RH  relative humidity
f  frequency
Psat  pressure of saturation
H  absolute humidity in percent
fr,O  relaxation frequency of oxygen
fr,N  relaxation frequency of nitrogen
a(f)  absorption frequency at frequency f
ΔSPL₃  difference in sound pressure level due to atmospheric absorption
rᵢ  depth of layer i
G  ground factor
k  wave number
Δr  difference between direct and reflected path
f_c  central frequency
ΔSPL₄  difference in sound pressure level due to ground impedance
SPLₙₖ₉  corrected SPL at frequency fₙₖ₉
SPLₙₖₘ  measured SPL at frequency fₙₖₘ
SPLₙₖ₉  background SPL at frequency fₙₖ₉
\( c(f_k) \)  \hspace{1cm} \text{signal-to-noise ratio at } f_k \\
SR \hspace{1cm} \text{sample rate} \\
BW \hspace{1cm} \text{bandwidth or frequency resolution} \\
NPTS \hspace{1cm} \text{number of spectral output points} \\
t_r \hspace{1cm} \text{time increment of assumed local stationarity} \\
NSHIFT_i \hspace{1cm} \text{number of points corresponding to } \Delta t_i \\
P_a^2(f) \hspace{1cm} \text{average power at frequency } f \\
REF \hspace{1cm} \text{reference for } \text{dB conversion} \\
SPL_a(f) \hspace{1cm} \text{average sound pressure level at frequency } f
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,(D/N-1)*VEL10 POINTS WHERE D IS THE MICROPHONE
SPACING, VEL IS THE AIRCRAFT'S SPEED AND SR IS THE SAMPLE RATE
OF DIGITIZATION.

FOR RUN#22, TAU IS IN INCREMENTS OF 875.)

D. "GRIDLEY
2/22/80

OVISBH MAINTAINED BY SYSTEM DEVELOPMENT CORPORATION

DIMENSION DATA(1875,12),NAMES(12),IUNITS(12),IHDR(B),IST(IO)
DIMENSION JHDR(A)
NAMELIST/INPUT/IST,NPTSS
DATA IST:5,10,15,20,25,30,35,40,45,50/,
JHDR/4110H

201 CONTINUE
C
READ SKIP FACTORS FOR CHANNELS
READ (14,INPUT)
IF (EOF(14)) 202,203
202 STOP 'NO NAMELIST FOUND'
203 CONTINUE
READ (14,5010) JHDR
5010 FORMAT(10)
IF (NPTSS .GT. 1875) STOP 'NPTSS CAN NOT BE GT 1875'
C
C 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) IST,NCHAN,NAMES(I),IUNITS(I),IHDR(I),IST(IO)
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 S
10 LI=0
11 DO 10 K=1,NCHAN
READ (1) (DATA(K,I),I=1,IST(IO))
IF (EOF(1)) 6,10
10 LI=I+1
L=LSAV+K-I
IF (L.EQ.0) GO TO 21
1 IF (I.EQ.0) GO TO 11
11 CONTINUE
L=NPTSS
12 WRITE (E) (DATA(K,1),K=1,L)
WRITE (E) (DATA(K,L),K=1,L)
13 J=J+1
DO 20 J=1,NCHAN
IF (IST(J-2).GE.IJ) GO TO 20
WRITE (E) (DATA(K,J),K=1,L)
IF (J.EQ.0) EJ=J+1
20 CONTINUE
```

D. "GRIDLEY
2/22/80

OVISBH MAINTAINED BY SYSTEM DEVELOPMENT CORPORATION

DIMENSION DATA(1875,12),NAMES(12),IUNITS(12),IHDR(B),IST(IO)
DIMENSION JHDR(A)
NAMELIST/INPUT/IST,NPTSS
DATA IST:5,10,15,20,25,30,35,40,45,50/,
JHDR/4110H

201 CONTINUE
C
READ SKIP FACTORS FOR CHANNELS
READ (14,INPUT)
IF (EOF(14)) 202,203
202 STOP 'NO NAMELIST FOUND'
203 CONTINUE
READ (14,5010) JHDR
5010 FORMAT(10)
IF (NPTSS .GT. 1875) STOP 'NPTSS CAN NOT BE GT 1875'
C
C 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) IST,NCHAN,NAMES(I),IUNITS(I),IHDR(I),IST(IO)
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 S
10 LI=0
11 DO 10 K=1,NCHAN
READ (1) (DATA(K,I),I=1,IST(IO))
IF (EOF(1)) 6,10
10 LI=I+1
L=LSAV+K-I
IF (L.EQ.0) GO TO 21
1 IF (I.EQ.0) GO TO 11
11 CONTINUE
L=NPTSS
12 WRITE (E) (DATA(K,1),K=1,L)
WRITE (E) (DATA(K,L),K=1,L)
13 J=J+1
DO 20 J=1,NCHAN
IF (IST(J-2).GE.IJ) GO TO 20
WRITE (E) (DATA(K,J),K=1,L)
IF (J.EQ.0) EJ=J+1
20 CONTINUE
```
20 CONTINUE
IF (L1.EQ.0) GO TO 1
21 DO 20 J=2,NCHAN
          ENDFILE J
          ENDFILE J
          REWIND J
30 CONTINUE
CALL EVICT(1)
IF (JHDR(1) .LE. IBLNK) GO TO 31
          DO 30 INT=1,4
32 JHDR(INT)=JHDR(INT)
31 CONTINUE
          WRITE (12) ISH,NCHAN,(NAME(1),I=1,NCHAN),(UNIT(I),I=1,NCHAN),
          1 (JHDR(I),I=1,4)
          WRITE FILES 8 THROUGH 12 ON TAPE
40 L=1, K=1, MPTSS
        READ (2)(DATA(K,1),K=1,MPTSS)
        READ (2)(DATA(K,2),K=1,MPTSS)
        L=1, I=1, J=1, KL=LSAV
        DO 54 J=3,NCHAN
          READ (J) (DATA(K,J),K=1,KL)
          IF (L.EQ.LSAV) L=LSAV
          IF (L.EQ.LSAV) GO TO 51
          IF (EOF(J))45,50,51
50 CONTINUE
        L=LSAV
        GO TO 51
45 L=1, I=1, J=1
        GO TO 50
51 IF (L.EQ.0) GO TO 100
          DO 56 I=1,L
            WRITE (13) (DATA(I,J),J=1,NCHAN)
30 CONTINUE
100 IF (L.EQ.0) GO TO 48
            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.
APPENDIX C. - Program for Ensemble Averaging

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

PROGRAM MANDATA is a package for manipulating data from a TITF 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 running on BASEX and the signal analysis program running on UNIVAC. It may be used on any TITF file, the stacking routine, however, is one that is applicable to AAP and NEUSAP created files.

DOOREN 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>
</table>
| JOPT     | INTEGER | 0 - do not stack files
|          |         | 1 - stack all serial numbers
|          |         | 2 - stack selected serial numbers
| JSN      | ARRAY   | SERIAL NUMBERS TO BE STACKED/USED WHEN JOPT=2
|          | 10010   | EXAMPLE: JSN=1,3,4 implies that serial numbers 1,3, and 4 will be stacked. If JSN=000, a constant will be written to channel 1.
| CONST    | REAL    | CONSTANT TO BE EMPLOYED WHEN JSN=000
| IFUNC    | INTEGER | 0 - no action
|          |         | 1 - add channels
|          |         | 2 - subtract channels
|          |         | 3 - multiply channels
|          |         | 4 - divide channels
|          |         | 5 - average channels
|          |         | 6 - Chi-squared equivalence test
| KSN      | INTEGER | NEW SERIAL NUMBER OF FIRST FILE RETAINED (SUBSEQUENT FILES, IF ANY, WILL BE IN SEQUENTIAL ORDER FROM KSN)
| ISPL     | INTEGER | 0 - input is not in db
|          |         | 1 - input is in db
| OSPL     | INTEGER | 0 - output not to be in db
|          |         | 1 - output to be in db
REF REAL
0.0002

IWAR INTEGER
0

CONVERSION FACTOR FOR ISPL AND OSPL

FLAG TO CALCULATE ASSYMPOTIC AND
ESTIMATED VARIANCES WHEN AVERAGING
0=DO NO CALCULATE
1=CALCULATE

DIMENSION JNAMES(101),JUNITS(101),JSN(100),DATA(101),NAME(101),
UNITS(101),HDR(8)
REAL OSPL,JHDR(8)

DATA IOPT,JSN,CONV,IFUNC,KSN,ISPL,OSPL,REF/1,100*0,2*0,0,2*0,
0.0002,JWAR/8.

NAME IS PL/INPUT/IOPT,JSN,CONV,IFUNC,KSN,ISPL,OSPL,REF,IWAR

READ NAMELIST INPUT

READ INPUT.

IF (EOF(E)) 9999,2

THIS SECTION IS FOR STACKING SERIAL NUMBERS

THAT CONTAIN ONE CHANNEL OTHER THAN TIME OR FREQUENCY.

1

BEGINNING OF OPERATIONS

IF (EOF(E)) 100,130

READ (1) ISN,NCHAN,(NAME(I),I=1,NCHAN),(UNITS(I),I=1,NCHAN),

IF (EOF(E)) 900,110

CONTINUE

IF (EOF(E)) 100,130

READ (1) ISN

READ (1) (DATA(I),I=1,2)

IF (EOF(E)) 600,270

49
IF (IJK.EQ.2) TIME=DATA(1)
300  DATA(I)=0.
   DATA(J)=CONST
400  JTO=9
   IF (MOD(JK,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(I)=JNAME(1)
   N(H)=999
   DATA=CONST
   GO TO 511
500  ENCODE (10,600,1,JUNTS(K)) ISN
   6001 FORMAT (L)
   JNAMES(I)=NAMES(2)
   501  READ (1) (DATA(L),L=1,2)
   IF (EOF(1).EQ.0) GO TO 511
510  IJK=JNAMES(I)+BDATA+DATA(1)+DATA(1)
   IF (IJK.NE.2) GO TO 511
   IF (DATA(I).EQ.TIME) GO TO 511
   IF (JSH(N).NE.999) DATA(I) GO TO 511
   PRINT, "SERIAL NUMBER ", ISN, " TIMES DO NOT AGREE WITH PREVIOUS SERIAL"
   STOP 'STACKING NON-COMPATIBLE SERIALS'
511  JK=K
512  IF (JSH(N).NE.999) AND. (IJK.EQ.3) TIME=DATA(1)
   JTO=7
   IF (MOD(JK,2).EQ.0) JTO=8
   IF (JTO.EQ.8) JTO=7
   READ (JTO) (DATA(L),L=1,JK)
   IF (EOF(JTO)) GO TO 520
   DATA(I)=DATA(I)+DATA(1)+DATA(1)
   WRITE (JTO) (DATA(L),L=1,K)
   IF (JSH(N).NE.999) GO TO 501
   DATA=CONST
   GO TO 512
600  ENDFILE JTO
   ENDFILE ITO
   IF (X.NE.2) REWIND JTO
   REWIND ITO
   IF (X.EQ.1) GO TO 100
   IF (X.NE.0) GO TO 100
670  DO 700 1=1,K
   JUNITES(I)+JUNITES(I)
   700  DO 875 1=1,L
   IF (I.EQ.1) NAME(I)=NAME(I)
   875  IF (I.EQ.1) NAME(I)+NAME(I)
   680  HDR(I)=HDR(I)
   700  READ (I) (DATA(I),I=1,L)
   IF (EOF(I)) 710,720
   710  ENDFILE KTO
   PRINT, " VALUES ARE STACKED ON TAPE4"
ACOIND ITO
REWIND ITO
GO TO 1000
720 WRITE (ITO) (DATA(I),I=1,NCHAN)
GO TO 700
900 IF (ILOPT.EQ.0) GO TO 670
STOP 'RECHECK SERIALS--ONE IS NOT FOUND'
1000 IF (IFUNC.NE.0) GO TO 1001
GO TO 1000
TO CONTINUE TO STACK SERIALS ON TAPE4
READ INPUT
IF (EOF(I)) 1000,2
900 ENDFILE ITO
STOP 'RECHECK REQUESTED CHANNELS STACKED ON TAPE4'
1001 IF (I) WRITE (6,500)
GO TO 700
REWIND M
READ (M) IG,NCWAN,(NAME(I),I=1,NCHAN),(LIMITS(I),I=1,NCHAN),
IF (EOF(M)) 1000,1060
1050 STOP 'TAPE 4 IS EMPTY'
1060 NCHAN=NCHAN+2
1070 READ (M) (DATA(I),I=1,NCH)
IF (EOF(M)) 2000, 1080
1080 IJKL=IJKL+4
IF (IJKL.EQ.16) CALL FDOCON(DATA,NCH,REF) $12.141
GO TO (1100,1200,1300,1400,1500,1700) IFUNC
STOP 'IFUNC CODE NOT DEFINED PROPERLY'
C C C
C C C
ACOIND ITO
REWIND ITO
GO TO 1000
720 WRITE (ITO) (DATA(I),I=1,NCHAN)
GO TO 700
900 IF (ILOPT.EQ.1) GO TO 670
STOP 'RECHECK SERIALS--ONE IS NOT FOUND'
1000 IF (IFUNC.NE.0) GO TO 1001
GO TO 1000
TO CONTINUE TO STACK SERIALS ON TAPE4
READ INPUT
IF (EOF(I)) 1000,2
900 ENDFILE ITO
STOP 'RECHECK REQUESTED CHANNELS STACKED ON TAPE4'
1001 IF (I) WRITE (6,500)
GO TO 700
REWIND M
READ (M) IG,NCWAN,(NAME(I),I=1,NCHAN),(LIMITS(I),I=1,NCHAN),
IF (EOF(M)) 1000,1060
1050 STOP 'TAPE 4 IS EMPTY'
1060 NCHAN=NCHAN+2
1070 READ (M) (DATA(I),I=1,NCH)
IF (EOF(M)) 2000, 1080
1080 IJKL=IJKL+4
IF (IJKL.EQ.16) CALL FDOCON(DATA,NCH,REF) $12.141
GO TO (1100,1200,1300,1400,1500,1700) IFUNC
STOP 'IFUNC CODE NOT DEFINED PROPERLY'
C C C
C C C
ACOIND ITO
REWIND ITO
GO TO 1000
720 WRITE (ITO) (DATA(I),I=1,NCHAN)
GO TO 700
900 IF (ILOPT.EQ.0) GO TO 670
STOP 'RECHECK SERIALS--ONE IS NOT FOUND'
1000 IF (IFUNC.NE.0) GO TO 1001
GO TO 1000
TO CONTINUE TO STACK SERIALS ON TAPE4
READ INPUT
IF (EOF(I)) 1000,2
900 ENDFILE ITO
STOP 'RECHECK REQUESTED CHANNELS STACKED ON TAPE4'
1001 IF (I) WRITE (6,500)
GO TO 700
REWIND M
READ (M) IG,NCWAN,(NAME(I),I=1,NCHAN),(LIMITS(I),I=1,NCHAN),
IF (EOF(M)) 1000,1060
1050 STOP 'TAPE 4 IS EMPTY'
1060 NCHAN=NCHAN+2
1070 READ (M) (DATA(I),I=1,NCH)
IF (EOF(M)) 2000, 1080
1080 IJKL=IJKL+4
IF (IJKL.EQ.16) CALL FDOCON(DATA,NCH,REF) $12.141
GO TO (1100,1200,1300,1400,1500,1700) IFUNC
STOP 'IFUNC CODE NOT DEFINED PROPERLY'
C C C
C C C
GO TO 1600

DIVIDING CHANNELS

A/B/C/D/...

CONTINUE

GO TO 1600

AVERAGE CHANNELS

(A+B+C+D+...)/(NCH-1)

CONTINUE

1.3, HCHADATA-ADATA/DATA(I)

CONTINUE

GO TO 1600

CHI-SQUARED EQUIVALENCE TEST

IF (IJ.EQ.1) SDATA=0

SDATA=SDATA+ALOG10(ABS(DATA(2)/DATA(3)))(NCH)

GO TO 1070

DATA(2) ADATA IF (IJ.EQ.1) KSN=KSN+1

IF (1.0.EQ.1) CALL TOCONP(AD,DATA(2),REF)

IF (IJ.EQ.1) WRITE (18) KSN,NDAT,(NAMES(I),I=1,NCH),(UNITS(I)),I=1,8)

WRITE (10) (DATA(K),K=1,2)

GO TO 1070

 IF (IFUNC.EQ.6) GO TO 3000

ENDFILE 10

REWIND M

GO TO 1

CONTINUE

COMPUTE CHI-SQUARE

CHISO=1.0*DATA(I)

PRINTS, "CHISO=",CHISO

CALL MDCH(CHISO,DF,IER)

IF (IER.EQ.1) PRINTS, "ERROR IN USING CHI" P=P+1

WRITE (6,5001)

GO TO 5000

CONTINUE

IF (IFUNC.EQ.6) CHI-SQUARED EQUIVALENCE TESTS

WRITE (10) (NAMES(I),I=2,3)

WRITE (10) (NAMES(I),I=2,3)

5000 FORMAT (5X,15X,5X CHI-SQUARED EQUIVALENCE TESTS)

5001 FORMAT (5X,3 THE PROBABILITY THAT CHANNEL 8,18,8 AND CHANNEL 8,
.A1B. / EK 1 ARE STATISTICALLY EQUIVALENT IS & F10.5,2 PERCENT"
END
SUBROUTINE FDBCON(DATA, NCH, REF)
DIMENSION DATA(101)
DO 1 I=2, NCH
DATA(I)=(10.*10.(DATA(I)/10.))*REF
1 CONTINUE
RETURN
END
SUBROUTINE TDBCON(DAT, REF)
IF (DAT) 2,1
2 CONTINUE
RETURN
END
APPENDIX D. - Program for Correcting Sound Pressure Levels

PROGRAM FLYOVER INPUT, OUTPUT, TAPE1INPUT, TAPE6OUTPUT, TAPE1TAPE2TAPE10)

PROGRAM FLYOVER IS TO CORRECT OUR FLYOVER SOUND PRESSURE LEVEL DATA GENERATED BY THE ACOUSTICS ANALYSIS PROGRAM (APP). 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

FLYOVER 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
Ex: 2: 101000 CORRECT FOR INSTRUMENTATION AND INV. SO.

A NAMELIST SINPUT5 IS EXPECTED NEXT.

NAMELIST SINPUT5

VARIABLE TYPE AND DESCRIPTION DEFAULT
ICODE INTEGER FLAG FOR WHICH THETAS ON TAPE1 TO CONSIDER
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 OF THTA

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

STRCR REAL ARRAY 5001-5999. 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
HGT    REAL    DISTANCE FROM GROUND TO MICROPHONE (FEET)

REF    REAL    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

IVAR   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 5000.0 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 FLAG FOR PRINTING WHEN IVAR=1

0: PRINT DELTA SPL VALUES
1: PRINT CORRESPONDING SPL VALUES

FOX A. MUELLER  D. RIDDLEY

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(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)`

`DATA (2), HAMPS(2), IUNITS(R), IHDR(8)`

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

`DATA CODE, THTA, N, STRCR/0, 10X0.0, 0, 8008-999./, HGT/30./,`

`P, T, ALT, RH/100*-999./, IVAR, F1, F2/0, 0., 50000e/,`

`NBLKS, NMICS, IPRINT/15.0, 512/`

`DATA CORR, DTSPL, D2SPL, D3SPL, D4SPL, DSSPL/I, 5*0.0/, REF/15.0/`

`READ ARRAY IFNC FROM INPUT`

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

`5111 FORMAT (6I11) IF (EOF(6)) 1000, 2`

`INITIALIZE VARIABLES`

`PI=.14159265`

`PO.1013.25`

`T0=283.15`

`WRITE NAMELIST $INPUT 5C`

`3 READ (5, INPUT)`

`IF (EOF(5)) 1000, 4`

`CONTINUE`

`WRITE INPUT INFORMATION TO OUTPUT FILE`

`WRITE (6,5222)`

`5222 FORMAT (IH11, //, SX, *IFNC - CODE FOR CORRECTIONS (0 MEANS NOT APPLIED: XT - 1 MEANS APPLIED)X, SX, *IN THIS ORDERX, //, 1X,`

`2 *INSTRUMENTATION CORRECTIONSX, //, 1X,`

`3 IDEUTIVE AMPLIFICATIONSX, //, 10X,`

`4 REVERSE SQUARE LAWX, //, 1X, 10X, ATHERMIC ABSORPTIONX, //, 10X,`

`5 GROUND IMPEDANCEX, //, 1X, 10X, DOPPLER FREQUENCY SHIFTX, //)`

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

`WRITE (6, INPUT)`

`READ HEADER RECORD, AND TR, RE, THETA, X, Y, AND Z (THE FIRST 7 DATA RECORDS TO DETERMINE IF THE FILE IS ONE REQUESTED BY CODE AND THTA)`

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

`1 (IHBR(I), I=1,8)`

`IF (EOF(1)) 900, 20`

`20 IF=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 IF=1+1`

`IF (IF .EQ. 1) TR=DATA(2)`

`IF (IF .EQ. 2) RE=DATA(2)`

`IF (IF .EQ. 3) THETA=DATA(2)`

`IF (IF .EQ. 4) X=DATA(2)`

56
IF (IJ .EQ. 5) UEL = DATA(I2)
IF (IJ .EQ. 6) ZE = DATA(I2)
IF (IJ .NE. 7) GO TO 25
100 IF (ICODE .NE. 3) GO TO 120
DO 110 I = 1,10
IF (THTA .GE. (THTA(I)+0.1) .AND. THTA .LE. (THTA(I)+0.11))
1 GO TO 120
110 CONTINUE
OTHERWISE CALL SKIPFF
CALL SKIPFF(SLTAPE1, 1)
GO TO 10
120 KCNT = KCNT + 1
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 = 992.
WRITE (10) DUM, TR
DUM = 992.
WRITE (10) DUM, RE
DUM = 992.
WRITE (10) DUM, THETA
DUM = 992.
WRITE (10) DUM, XRAD
DUM = 992.
WRITE (10) DUM, UEL
DUM = 994.
WRITE (10) DUM, ZE
DUM = 996.
WRITE (10) DUM, YRAD
IF (KCNT .GT. 1) GO TO 180
ADD 273.15 TO TEMP ARRAY
DO 135 I = 1, 25
135 IF (T(I) .NE. -999.) T(I) = T(I) + 273.15
COMPUTE AVERAGE TEMPERATURE FOR AVERAGE C FOR DF, AND CA
AT = 0.
ICNT = 0
140 I = I + 1
150 IF (T(I) .EQ. -999.) OR. I .GT. 25) GO TO 150
IF ((ALT(I) .LT. (HGTS * 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 * 343.23548 / T(I)
CAVG = (343.23548 * T(I)) / AT
COMPUTE DELTA HEIGHT IN ALTITUDE
DHGT = ALT(I) - ALT(2)
COMPUTE C CLOSEST TO GROUND FOR GROUND IMPEDANCE

ALTM=99999
DO 170 I=1,Nn,66
  IF (ALTII .EQ. -999.) GO TO 170
  IF (ALTII .GE. ALTM) GO TO 160
  ALTM=ALTII
  ID=I
160 CONTINUE
170 C&I=(343.234SORT(ID)/100)/0.3048

COMPUTE VARIABLES FOR CORRECTIONS

CORR 15 FOR THE DOPPLER FREQUENCY SHIFT
ANCH IS THE AVERAGE MACH NUMBER
DTSPL IS FOR THE INVERSE SQUARE LAW
D2SPL IS FOR CONVECTIVE AMPLIFICATION
DK IS A NUMER BASED ON MACH NUMBER
DR IS THE PATH DIFFERENCE BETWEEN REFLECTED AND
DIRECT PATH
AKDR IS THE PRODUCT OF DK AND DR

180 AMCH=VEL/CAVG
CHAMP=(1.-AMCH*COS(THETA])/180.
DTSPL=20.4ALOG10(AFREF/REF)
D2SPL=20.3*24+2*ALOG10(CORR)

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

DR=SORT(RE*4.4E+8HGT)/RE

WRITE TABLE HEADINGS ON OUTPUT IF IVAR=1

IF (IVAR NE. 1) GO TO 133
WRITE (6,5444) (IHDR(L),L-1,8)
5444 FORMAT (1H1,///,8010)
WRITE (6,5444) TR,RE,THETA,XRAD,VEL,ZF_,YRAD
5444 FORMAT (//,SX,T*,E12.5,4X,STHETA-E,E12.5,4X,THETA-E,E12.5,4X,
1X*E12.5,4X,THETA-E,E12.5,4X,1X*E12.5,4X,THETA-E,E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,1X*E12.5,4X,
ORIGINAL PAGE IS OF POOR QUALITY

2 (CAI$1,12X,*I$,12X,$(AA)12X,12X,$(GI)*,7X,*$,$,*$)
3 (12X,$FINAL$,$/$)
130 CONTINUE

C READ AND APPLY REQUESTED CORRECTIONS
C
C 200 READ (1) F,SPL
IF (EOF(1)) GO TO 210
210 IF (ITAPE .NE. 0) GO TO 200
SPL1=SPL1+SPL2+SPL4+SPL5+SPL3
IF (ITAPE .NE. 1) GO TO 456
215 IF (IFNC(1) .EQ. 0) GO TO 300
C
C 208 READ (1) F,SPL
IF (F.GE. F1 .AND. F .LE. F2) GO TO 215
IF (ITAPE .NE. 0) GO TO 200
FFF-F
SPL1=SPL1+SPL2+SPL4+SPL5+SPL

C 215 IF (IFNC(1) .EQ. 0) GO TO 300
C
C 210 IF (ITAPE .NE. 0) GO TO 200
C
C INSTRUMENTATION CORRECTIONS
C
FF+DB1=DB2+DB3=0.0
IF (F.GT. STRCR(1,1)) GO TO 220
IND=0
DO 260 I=1,200
IF (STRCR(I,1) .LT. -999.) GO TO 280
IF (F.LT. STRCR(I,1)) GO TO 270
IND=I
FF=STRCR(I,1)
DB1=STRCR(I,3)
DB2=STRCR(I,4)
270 CONTINUE
260 DO 260 I=1,200
IF (STRCR(I,1) .LT. -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)
270 CONTINUE
IND=IND+1
RATIO=(F-FF)/(STRCR(IND,1)-FF)
DB1=DB1+RATIO*(STRCR(IND,2)-DB1)
DB2=DB2+RATIO*(STRCR(IND,3)-DB2)
DB3=DB3+RATIO*(STRCR(IND,4)-DB3)
SPL1=SPL1+DB1 +DB2 +DB3
SPL2=SPL2+SPL1+SPL3
SPL3=SPL3+SPL2+SPL4+SPL5+SPL6
SPL4=SPL4+SPL3+SPL5+SPL6
SPL5=SPL5+SPL4+SPL6
SPL6=SPL6+SPL5+SPL6

C 300 IF (IFNC(2) .EQ. 0) GO TO 310
C
C CONVECTIVE AMPLIFICATION
C
C 310 IF (IFNC(3) .EQ. 0) GO TO 320
C
C INVERSE SQUARE LAW
C
C 320 IF (IFNC(4) .EQ. 0) GO TO 400
C
C ATMOSPHERIC ABSORPTION
C
DSPL=0.0
DO 350 I=1,25
IF (ALT(I) .GT. 999. ) GO TO 350
IF ((ALT(I) .LT. (HGT*.3648)) .OR. (ALT(I) .GT. (ZE*.3648)))
1 GO TO 350
C=3.43335427*(T(I)/273.16)**1.348765*(T(I)/273.16)**-0.509888*ALG(T(I)/273.16)
1 T=0.047480*.00818*(1-10.28/(2.5688*(T(I)/273.16)-1.))
350 CONTINUE

59
GROUNDED IMPEDANCE

DUM = -(AS2.0*PI*FRS/FSDR/CGI)*XAS2.0)
IF (DUM.GT.-675.84) 401 C0NTINUE
D3SPL = 10.0*LOG10 (2. + 2.8*2.0*COS(2.0*PI*FRS/FSDR/CGI)*W)
SPL = SPL + D3SPL

DOPPLER FREQUENCY SHIFT

F = FSCORRW
D3SPL = D3SPL
SPL1 = SPL + D3SPL + DTSPL - D2SPL - D2SPL
SPL2 = SPL + D3SPL + DTSPL - D2SPL - D2SPL
SPL3 = SPL + D3SPL + DTSPL - D2SPL - D2SPL
SPL4 = SPL + D3SPL + DTSPL - D2SPL - D2SPL
SPL5 = SPL + D3SPL + DTSPL - D2SPL - D2SPL

SET UP AND PRINT VALUES IF IVAR = 1

IF (IFNC(6).EQ. 0) GO TO 430
C0NTINUE
SPL = SPL1 + SPL2 + SPL3 + SPL4 + SPL5 + SPL
IF (IPRINT.NE. 0) WRITE (6,5555) FFF, SPL1, SPL2, SPL3, SPL4, SPL5, SPL
5555 FORMAT (2X, F10.4, 8(4X, F10.4))
WRITE (10) F, SPL
GO TO 200
ENDFILE 10
GO TO 10
END
END OF PROGRAM
APPENDIX E. - Program for Acoustic Directivities

PROGRAM SPLHTC (INPUT, OUTPUT, TAPE1, INPUT, TAPE2, OUTPUT, TAPE3, TAPE4, TAPE5, TAPE6, TAPE7, TAPE8, 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 OUPAP3, OUPAP4, AND OUPAP5 APPLIED.

FOR OUID FORWARD SPEED EFFECTS
(A. MUELLER)

D. GRIDLEY (1981):

SPLHTC MAINTAINED BY SYSTEM DEVELOPMENT CORPORATION

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

II. NAMELIST #NAME:
A. FREQ IS THE DESIRED FREQUENCY AT WHICH TO GENERATE THE RADIATION PATTERN.
B. ERR IS THE BANDWIDTH WITHIN WHICH TO FIND SPL VALUE (RANGE OF FREQ-ERR TO FREQ+ERR IS USED).
C. ISSUE IS THE FLAG FOR SUBTRACTING BACKGROUND (0 = DO NOT SUBTRACT, 1 = SUBTRACT)
D. ISUMTPV IS THE FLAG FOR COMPUTATIONS (0 = HIGHEST VALUE IN RANGE, 1 = SUM OF TWO HIGHEST SPL VALUES IN THE RANGE).

DIMENSION NAMES (2), JUNITS (2), JNAMES (2), DATA (2), NAMES (2), JUNITS (2),
1.
THETAB (50), DATA (100), NAME (2)
2.
NAMELIST /NAME/FREQ, ERR, ISSUE, ISUMTPV
DATA ERR/200, FREQ/2000, ISSUE/0, ISUMTPV/0,
1.
NAMES(1) THETA, 10H SPL
2.
JUNITS(1) DEGREES, 10H DB
3.
READ (5, NAME)
4.
IF (EOF(1)) 1000, 5
5.
WRITE (6, NAME)
6.
READ (7, NAME)
7.
IF (EOF(1)) 1000, 5
8.
WRITE (6, NAME)
9.
READ (1) ISN, NCHAN, (NAMES(1), 1=1, NCHAN), (JUNITS(1), 1=1, NCHAN),
10.
(JUNITS(1), 1=1, NCHAN), (NAMES(1), 1=1, NCHAN), (JUNITS(1), 1=1, NCHAN),
11.
J=0
12.
DBMAX=-99999.
13.
IF (X.EQ. 1) WRITE (8) ISN, NCHAN, (NAMES(1), 1=1, NCHAN),
14.
(JUNITS(1), 1=1, NCHAN), (JUNITS(1), 1=1, NCHAN), (JUNITS(1), 1=1, NCHAN),
15.
J=0
16.
READ (1) DATA (1), 1=1, NCHAN
17.
IF (EOF(1)) 35, 40
18.
STOP 'FREQUENCY REQUESTED IS TOO LARGE'
19.
J=J+1
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  
DBMAX = DATA(2)  
GO TO 52  
52 DBMAX = DATA(2)  
GO TO 30  
55 IF (DATA(2) .LT. DBMAX) DBMAX = DATA(2)  
GO TO 30  
60 IF (ISUMTYP .EQ. 0) GO TO 35  
DBMAX = 10.**((DBMAX/20,1)*0.0002)  
DBMAX = 10.**((DBMAX/20,1)*0.0002)  
DBMAX = 10.**LOG10((DBMAX**DBMAX**DBMAX**DBMAX*10.0002))

WRITE (9) THETA, DBMAX
READ (1)
IF (EOF(1)) 10.70  
70 CALL SKIP(96TAPE1,1)  
GO TO 10  
500 ENDFILE 9  
REUNID 1  
IF (IPSUB .EQ. 0) GO TO 1  
ENDFILE 9  
REUNID 2  
PLACE BKG INTO ARRAY THETAB AND DBB

510 READ (2) ISN  
IF (EOF(2)) 520,530  
STOP 'NO RADAR FILE INFO.'  
530 IJ = 0  
540 IJ = IJ+1  
READ (2) THETAB(IJ), DBB(IJ)  
IF (EOF(2)) 550,540

550 IKT = IJ+1  
READ SPL VS THETA AND SUBTRACT BKG--ONLY THOSE THETAS WITH A CORRESPONDING THETAB WILL BE WRITTEN TO TAPE10.

600 READ (9) ISN, NCHAN, (HAMES(I),I=1,NCHAN), (IUNIT(I),I=1,NCHAN),  
(1HDR(I),I=1,NCHAN)  
IF (EOF(9)) 1000,610  
610 WRITE (10) ISN, NCHAN, (HAMES(I),I=1,NCHAN), (IUNIT(I),I=1,NCHAN),  
(1HDR(I),I=1,NCHAN)  
620 READ (9) THETA, DB  
IF (EOF(9)) 750,630

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
IF((DB-DBBL) .GT. 3.) GO TO 660
WRITE (6,6600) ISH, THETA
6600 FORMAT(/,3 FOR SERIAL #,16,#5/H IS C OR = 3 FOR ANGLE #,F10.4)
IF ((DB-DBBL) .LE. 0.) GO TO 670
GO TO 665
665 IF ((DB-DBBL) .GE. 10.) GO TO 670
670 DB+DB+10. LOG (DBL-1.10.*((DBL-DB)) 10.)
WRITE (10) THETA, DB
ENDFILE 10
ENDFILE 10
GO TO 100
100 GO TO 1
1000 STOP "END OF PROGRAM"
END
REFERENCES


