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ACOUSTIC SOURCE BEARING ESTIMATION (ASBE) COMPUTER PROGRAM DEVELOPMENT

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FOREWORD

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SUMMARY

This paper describes the development and usage of an atmospheric Acoustic Source Bearing Estimation (ASBE) program. The basis for this program is a maximum-likelihood method of spectral analysis which was formulated by J. Capon for seismic array processing.

Presented herein is the mathematical development of the Acoustic Source Analysis Technique (ASAT) bearing estimation algorithm that is used in ASBE. Included in this report is the input and output of a test case that was used to validate the algorithm.

SYMBOLS

B	block averaged spectral estimate
B*	complex conjugate of B
E	complex valued column matrix representing steering vectors
E'	conjugate transpose of E
F	block averaged auto and cross spectrum
F*	complex conjugate of F
G	equivalent to EE'
Hz	hertz
K	number of data points in each subblock of data
L	number of data points being analyzed
M	number of sensors
N	number of blocks that L data points are separated into (L=KN)
NSP	number of sensor pairings whose separation of distance is less than or equal to one-half the wavelength of the frequency being analyzed
P	frequency-wavenumber power spectrum estimate
P_w	ASAT power spectrum estimate
R	complex valued square matrix representing the diagonally normalized-block averaged estimate of the cross power spectral densities
R^{-1}	inverse of R
S	complex values square matrix representing the cross power spectral densities obtained from the Fast Fourier Transform
X(i),Y(i)	rectangular coordinates of sensor i
c	speed of sound
f	frequency being analyzed
j	complex variable ($\sqrt{-1}$)
Δt	time step between data points
β	azimuthal look direction

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INTRODUCTION

Acoustic array processing entails the analysis of signals which are carried by propogating atmospheric sound waves. These waves impinge upon an array of sensors from which the received signals are recorded as pressure variations. Through array processing, the pressure values can yield signal characteristics such as power-frequency signature and source bearing.

A major component of signal processing is the estimation of power spectral densities. Techniques which have been devised to estimate spectral values are divided into two major categories - window-based and model-based. The first category, window-based methods, are non-adaptive in the sense that all sets of data are treated alike. The most widely known and used window-based method is the Fast Fourier Transform (FFT). For a single sensor, the FFT will estimate the power-frequency signature of the signal being analyzed. From this signature, estimates can be made for the power spectral density and auto correlation values. When signatures are available from two sensors, estimates can be made for the cross power spectral density and cross correlation values. When a signal signature is required as output from the entire sensor array, the second category of spectral estimation, model-based methods, must be utilized.

Unlike window-based methods, model-based methods of spectral estimation are considered adaptive since they are data dependent. For each set of data being analyzed, the characteristics of the model-based estimator are automatically adjusted. This adaptive capability allows much higher spectral resolution than the FFT technique. Since source bearing is of critical importance in acoustic analysis, the model-based method that was investigated is the maximum likelihood method (MLM). As documented by Capon[1], this Maximum Likelihood Method provides a high-resolution estimate for frequency-wavenumber spectrum. The MLM acts as a filter at a selected frequency which passes that frequency undistorted and rejects all others. For a particular frequency, the MLM spectral estimates are a function of azimuth and are power levels that are received by the sensor array as a whole. Therefore, the azimuth which results in the largest power level is an estimate of source bearing.

Both the MLM and FFT methods of spectral estimation have unique qualities which are beneficial in acoustic array processing. This report will present the derivation and implementation of ASAT, a technique to estimate acoustic source bearing which utilizes the capabilities of each method.

TECHNICAL DISCUSSION

When the FFT is used to estimate each sensors' signal signature the highest frequency in the spectrum is equal to one-half the data sampling rate. This cutoff frequency is referred to as the Nyquist frequency. Additionally, the FFT is symmetric in the sense that there are an equal number of positive and negative frequency components in the spectrum. The frequency resolution of the FFT is therefore determined by this symmetry and the Nyquist frequency. For example, if L datum is sampled at time intervals of Δt and L datum is analyzed by the FFT then:

$$\text{Nyquist frequency} = 1/(2\Delta t) = L/2$$

$$\text{frequency resolution} = (1/2\Delta t)/(L/2) = (L/2)/(L/2) = 1\text{Hz}.$$

In order to increase the frequency resolution, augmenting zeros can be added to the sampled data. Using the example given previously, the Nyquist frequency remains $1/2\Delta t$ and if L zeros are added to the data then:

$$\text{frequency resolution} = (1/2\Delta t)/(2L/2) = (L/2)/L = 1/2\ \text{Hz}.$$

Thus, the frequency resolution has been doubled. An additional benefit of this zero padding is that foldover aliasing has been reduced. This type of aliasing occurs when frequencies higher than the Nyquist frequency occur in the data. These higher frequency components cause power foldover and distort the spectrum at lower frequencies.

The multi-channel processing technique developed by Capon also requires spectral estimates for each sensor. Unlike the

FFT, the MLM utilizes a block averaging method to estimate a signal's signature. In this method a sample of a sensor's recorded signal is subdivided into blocks which contain an equal number of datum. The number of blocks must be greater than or equal to the number of sensors. For an array containing M sensors, each having a signal length of L data points which is subdivided into N blocks of K points such that $N \geq M$ and $L = KN$, each sensor's block averaged spectral estimate in the nth block at a preselected frequency f is:

$$B_{in} = (K)^{-1/2} \sum_{m=1}^{K} K_{i,m+(n-1)K} e^{jm2\Pi f\Delta t} \qquad \begin{array}{l} i=1,\ldots,M \\ n=1,\ldots,N. \end{array} \qquad (1)$$

The frequency resolution of the block averaging method is lower than that of the FFT. Whereas the FFT could have a resolution of 1 Hz or less, this method would result in a resolution of only $1/(K\Delta t)$. For example, if $L=1/\Delta t = 2000$, $M=9$ and $N=10$, then $K=200$ and the block averaged frequency resolution would be $2000/200 = 10Hz$.

Once spectral values have been estimated for each block in each sensor, the MLM requires that auto and cross spectrum be calculated by:

$$F_{il} = \frac{1}{N} \sum_{n=1}^{N} B_{in} B_{ln}^{*} \qquad i,l=1,\ldots M \qquad (2)$$

where B* indicates complex conjugate. A diagonal normalization is then performed by dividing F_{i1} by $[F_{ii}F_{11}]^{1/2}$. These spectrum values are then arranged in a square matrix as:

$$\begin{matrix} F_{11} & F_{12} & F_{13} & \cdot & F_{1M} \\ F_{21} & F_{22} & F_{23} & \cdot & \cdot \\ F_{31} & F_{32} & F_{33} & \cdot & \cdot \\ F_{41} & \cdot & \cdot & \cdot & \cdot \\ \cdot & \cdot & \cdot & \cdot & \cdot \\ F_{M1} & \cdot & \cdot & \cdot & F_{MM} \end{matrix}$$

where the li indexed values below the diagonal are the complex conjugates of the il indexed values above the diagonal. Once this complex valued matrix has been arranged, its inverse must be calculated. Situations can exist where the matrix is singular and thus has no inverse. If there is not enough reasonable data in each block or if the signal is transient in nature, then the matrix will be singular. When these conditions arise, the frequency under study and possibly the entire database will not be able to be analyzed using the MLM technique.

If the inverse of the spectrum matrix exists, then the high-resolution estimate for the frequency-wavenumber array spectrum is:

$$P = [E'R^{-1}E]^{-1} \qquad (3)$$

where P is the real, positive valued power estimation of the spectrum; E is a column matrix whose complex elements represent steering vectors for each sensor; E' is the conjugate transpose of E; and R^{-1} is the inverse of the complex valued square matrix representing the diagonally normalized block averaged estimation of the cross power spectral densities at frequency f. Each element of the matrix E is of the form:

$$E(i) = \exp[-j2\pi f(X(i)\mathrm{COS}\beta + Y(i)\mathrm{SIN}\beta)/c] \qquad (4)$$

where f is the frequency, X(i) and Y(i) are the rectangular coordinates of sensor i, β is the azimuthal look direction, and c is the speed of sound for the local atmospheric conditions.

Implementation of equation 3 is governed by the following assumptions:

1. analysis is done on a frequency by frequency basis
2. the output from a sensor is a wide-sense stationary discrete-time parameter random process with zero mean
3. the output from the sensor array comprises a homogeneous random field

Under these assumptions, equation 3 can be used to estimate the bearing of the source by determining which azimuth maximizes the power P at a selected frequency f.

The preceeding discussion has presented a frequency-power spectral estimation technique, the FFT, and a multi-channel processing technique, the MLM. Each technique has qualities which are beneficial in acoustic array processing. Namely, the

FFT has high frequency resolution and anti-aliasing whereas the MLM will estimate source bearing. Spectral estimates obtained via the MLM were shown to be of low frequency resolution and could also give rise to a singular matrix. Therefore, it would be advantageous to incorporate the positive qualities of each technique into a unified model. The remainder of this paper will present the mathematical development of the ASAT algorithm and the implementation of this hybrid model of bearing estimation.

DERIVATION OF THE ASAT LOCATION METHOD

The high resolution power estimate equation is:

$$P = [E'R^{-1}E]^{-1} \tag{5}$$

since P is a scalar, the inverse of this equation is:

$$1/P = E'R^{-1}E \tag{6}$$

multiply both sides by E'R:

$$(E'R)/P = E'R^{-1}E(E'R) \tag{7}$$

multiply both sides by E:

$$(EE'R)/P = EE'R^{-1}(EE'R) \tag{8}$$

since P is a scalar:

$$(EE'R) = PEE'R^{-1}(EE'R) \tag{9}$$

therefore:

$$PEE'R^{-1} = \text{identity matrix } I \tag{10}$$

multiply both sides of equation 10 by R:

$$PEE'R^{-1}R = IR \tag{11}$$

therefore:

$$P(EE') = R. \tag{12}$$

Equation 12 eliminates the need of computing the inverse of matrix R. Since the inverse is no longer required, matrix R need not be computed from the block averaged-diagonally normalized method. Instead, the matrix can be obtained from the Fast Fourier Transform values. This then allows the use of the higher frequency resolution and anti-aliasing features of the FFT.

APPLICATION OF THE ASAT LOCATION METHOD

Let the matrix G represent (EE'), matrix S represent the cross power spectral densities obtained from the Fast Fourier Transform, and P_w represent the power estimate. Equation 12 can be rewritten as:

$$P_w G = S. \tag{13}$$

Since P_w is a scalar value, every element of matrix S must equal every element of matrix G multiplied by P_w. Therefore, P_w must equal every element of S divided by each corresponding element of G:

$$P_w G(i,j) = S(i,j) \Rightarrow P_w = S(i,j)/G(i,j). \tag{14}$$

In practice, the value of P_w will vary since the S values are estimates obtained from a small segment of signal time history. To account for this variance of P_w, an ensemble averaged value will be used:

$$P_w = \frac{\sum_{i=1}^{M} \sum_{j=i}^{M} (S(i,j)/G(i,j))}{NSP} \tag{15}$$

The double summation over M sensors in equation 15 is constrained to i-j sensor combinations whose separation distance is less than or equal to one-half the wavelength of the frequency being analyzed. By applying this constraint, spatial aliasing is avoided. NSP is the number of sensor pairings which satisfy this distance constraint.

Bearing estimation can be accomplished by using equation 15, the ASAT algorithm.

ASBE PROGRAM FLOW ANALYSIS

The program which was developed to locate acoustic sources (Acoustic Source Bearing Estimation - ASBE) can be analyzed in six parts.

These operational components are:

1) read test site parameters from file MIKE
2) read sensor output data from file TIFT
3) convert sensor output data from dynes/sq-cm to Pascals and apply Hamming window
4) compute cross power spectral densities by Fast Fourier Transform and compute ensemble averaged values for each frequency
5) locate N peak ensemble averaged values where N is the degrees of freedom (number of sensors less 1)
6) for each peak ensemble averaged value:
    a) compute power values at regular intervals of azimuth
    b) locate the peak values in the azimuthal power array, these values give the azimuths from which the source signals are assumed to radiate.

ASBE PROGRAM PARAMETER STATEMENTS

Certain arrays in the ASBE program are dimensioned by parameter statements. The user set parameter declarations are:

Variable	Description
MAXNS	integer number of sensor output values to read
NMIKES	integer number of sensors
SPS	real value for the sampling rate of the data (units: samples per second)
DELAZ	real value defining the resolution to be used in the azimuth power calculation (units: degrees)

ASBE PROGRAM INPUT/OUTPUT

Input to the ASBE program consists of the free formatted file MIKE and the unformatted file TIFT.

File MIKE describes test conditions and contains:

Record	Variables	Description
1	LABEL	test name, maximum of 80 characters
2	NMK,AIRT	integer value defining number of sensors used, real value defining test site air temperature in degrees Farenheit
3	PEAKI,PEAKF	real values defining the minimum and maximum values of the frequency range (hertz) to use in the peak search
4	X,Y	coordinates for sensor #1 in feet
5	X,Y	sensor #2 coordinates
	. .	
	. .	
	. .	
NMK+3	X,Y	sensor #NMK coordinates

Note that the X,Y values defining each sensor position are in reference to a sensor which was selected as the origin and has a X-Y location of (0.,0.).

File TIFT contains the recorded pressure data and its structure is:

Record	Variables	Description
1	ISN,NC,(NAM(I),I=1,NC),(IU(I),I=1,NC),(IHD(J),J=1,8)	integer test serial number integer number of channels recorded, 10-character name of each channel, 10-character units name for each channel, 80-character header name for the test
2	(CH(I),I=1,NC)	CH(1): real-time value, CH(2) thru CH(NC): real pressure values for NMK sensors at time CH(1)
	. . .	
MAXNS+1	(CH(I),I=1,NC)	pressure values at ending time value

The formatted output file from ASBE is named BEARING. It contains a list of program parameter values, an echo of input file MIKE, the calculated peak frequencies and associated power values, and the peak azimuth/power values for each peak frequency. An example of files MIKE and BEARING appear in figures 2, 3a, and 3b.

SAMPLE TEST CASE

The acoustic source which was used to benchmark the ASBE program consisted of a stationary, gasoline fueled, auxiliary power unit (APU) equipped with a muffler. With respect to the sensor array origin (sensor #5), the APU was located at 120 degrees azimuth and at a distance of 50 feet. The sensor array was comprised of nine B&K half-inch sensors arranged in three triangles with three sensors in each triangle. Figure 1 shows the test site geometry and sensor array setup.

Sensor output data for this test was digitized at 2000 samples per second for 30 seconds. For the benchmark analysis, this database was separated into 60 TIFT files. Each file contained 1000 samples covering a one-half second time block. Parameter statements setup for these 60 runs were:

Parameter	Value
SPS	2000.0
MAXNS	1000
NMIKES	9
DELAZ	1.0

Each run was executed on a Control Data Corporation CYBER 860 computer and required approximately 2.75 seconds of CPU time for each frequency analyzed. The input file MIKE used in these runs is shown in figure 2.

As a representative example of program execution, figures 3a and 3b show output file BEARING for the run covering the first one-half second of data. A histogram showing the typical variance of P_w (obtained from equation 15) is presented in figure 4. The power values used in this plot were calculated using a frequency of 140 hertz and a bearing of 121 degrees. Approximately 46 percent of the sensor pairings had power values that were within ± 0.6 percent of the mean. Figure 5 is a plot of the ensemble averaged spectrum magnitude. It is from this data that the peak frequencies used in this analysis are selected. The eight peaks selected for this run are highlighted by an asterisk. Polar power-bearing plots for each of the eight selected peak frequencies appear in figure 6. At each frequency the nominal direction for the maximum power is 120 degrees. It can be seen that sidelobes become more apparent at higher frequencies. This effect is caused by the sensor locations used in the test. As the frequency increases, more sensor pairings are excluded in the analysis since they have separation distances which are greater than one-half the wavelength of the frequency being analyzed. Thus, there is less data to use in the bearing estimation and this results in less discrimination of the sidelobes. In order for this effect to be minimized, the highest frequency would have to be known prior to the test so that the sensor array could be sized accordingly.

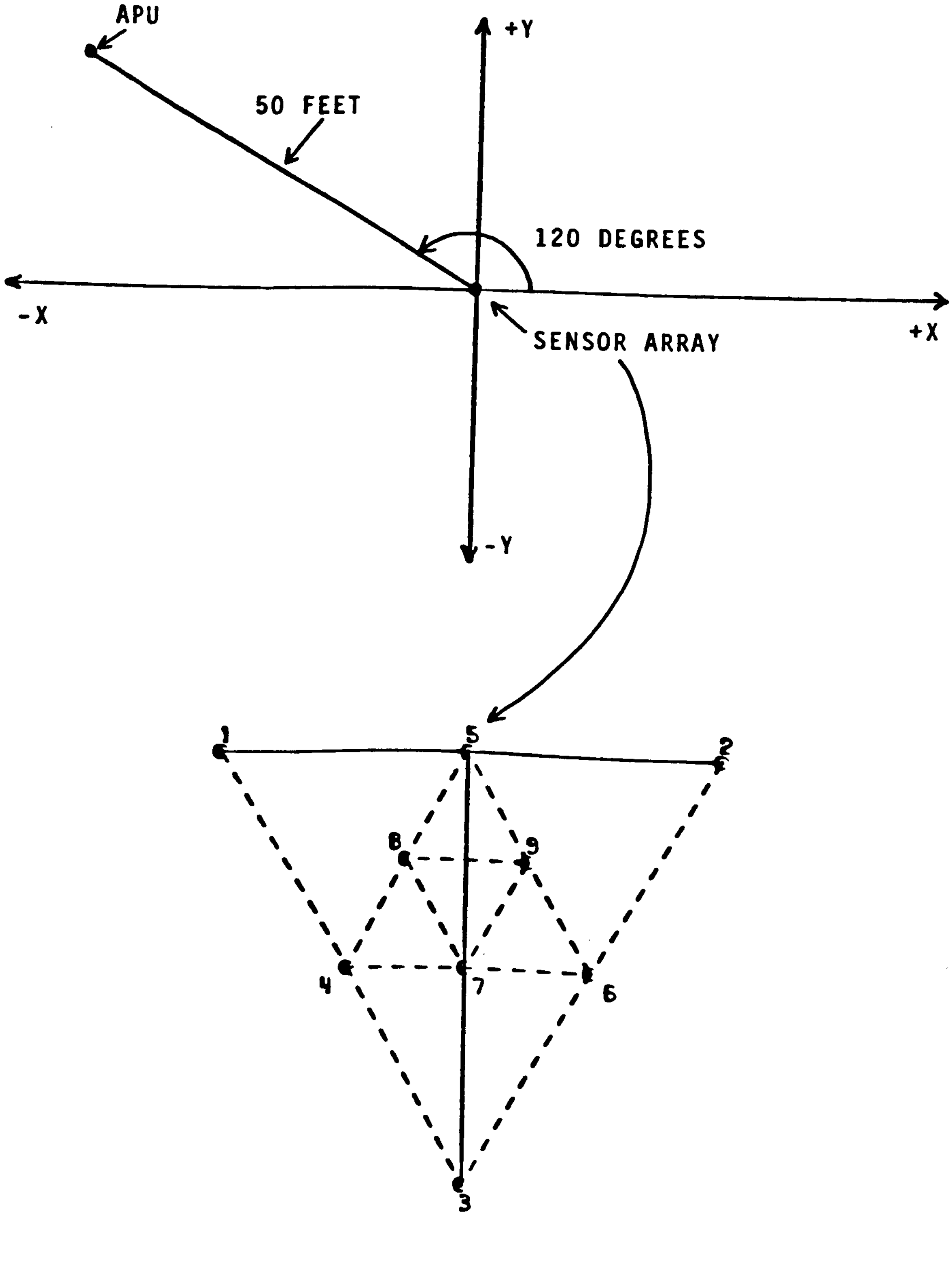

Figure 1: Test Site Setup

RECORD CONTENTS		DESCRIPTION
APU WITH MUFFLER		label
9	56.0	number of sensors, air temperature (degree Farenheit)
50.0	250.0	frequency range (hertz)
-4.0	0.0	X-Y coordinates for sensor 1 (feet)
4.0	0.0	X-Y coordinates for sensor 2 (feet)
0.0	-6.9	X-Y coordinates for sensor 3 (feet)
-2.0	-3.4	X-Y coordinates for sensor 4 (feet)
0.0	0.0	X-Y coordinates for sensor 5 (feet)
2.0	-3.4	X-Y coordinates for sensor 6 (feet)
0.0	-3.4	X-Y coordinates for sensor 7 (feet)
-1.0	-1.7	X-Y coordinates for sensor 8 (feet)
1.0	-1.7	X-Y coordinates for sensor 9 (feet)

Figure 2: Input File MIKE

***** RUN DIRECTIVES *****

PROGRAM PARAMETER STATEMENT SETS THE SAMPLES PER SECOND = 2000.0
PROGRAM PARAMETER STATEMENT SETS THE AZIMUTH RESOLUTION (DEG) = 1.0
PROGRAM PARAMETER STATEMENT SETS THE NUMBER OF DATA POINTS = 1000
PROGRAM PARAMETER STATEMENT SETS THE NUMBER OF MICROPHONES = 9
CASE LABEL : APU WITH MUFFLER
NUMBER OF MICROPHONES : 9
AIR TEMPERATURE (DEG F) : 56.0
SEARCH FOR FREQUENCY PEAKS WITHIN RANGE OF 50.0 TO 250.0 HERTZ
MICROPHONE COORDINATES :

MIC #	1	X : -4.0	Y :	0.0
MIC #	2	X : 4.0	Y :	0.0
MIC #	3	X : 0.0	Y :	-6.9
MIC #	4	X : -2.0	Y :	-3.4
MIC #	5	X : 0.0	Y :	0.0
MIC #	6	X : 2.0	Y :	-3.4
MIC #	7	X : 0.0	Y :	-3.4
MIC #	8	X : -1.0	Y :	-1.7
MIC #	9	X : 1.0	Y :	-1.7

***** TIFT DATA PROCESSING *****

SERIAL NUMBER : 15
NUMBER OF CHANNELS : 10
HEADER : A/C TRACKING/LANGLEY TEST NO. AC RUN NO. 5

CHANNEL	NAME	UNITS
1	TIME	SECONDS
2	MIC1	DYNES/CM2
3	MIC2	DYNES/CM2
4	MIC3	DYNES/CM2
5	MIC4	DYNES/CM2
6	MIC5	DYNES/CM2
7	MIC6	DYNES/CM2
8	MIC7	DYNES/CM2
9	MIC8	DYNES/CM2
10	MIC9	DYNES/CM2

START TIME = 20999.6812 END TIME = 21000.1807

Figure 3a: Output File BEARING

***** PEAK SEARCH RESULTS *****

PEAK #	FREQUENCY (HZ)	SOUND PRESSURE LEVEL (DB)
1	140.0	56.5
2	116.0	56.1
3	93.0	54.1
4	163.0	51.9
5	209.0	51.3
6	186.0	50.7
7	233.0	49.7
8	70.0	47.5

***** SOURCE LOCATION RESULTS *****

FREQUENCY (HZ)	# PEAKS	AZIMUTH (DEG)	POWER (DB)
140.0	2	121.0	56.4
		300.0	52.0
116.0	3	122.0	56.0
		341.0	48.9
		244.0	48.9
93.0	1	121.0	54.1
163.0	1	120.0	51.9
209.0	3	121.0	51.2
		347.0	47.5
		254.0	47.3
186.0	2	118.0	50.9
		251.0	45.7
233.0	3	120.0	49.5
		254.0	46.6
		348.0	46.6
70.0	1	119.0	47.5

Figure 3b: Output File BEARING (cont.)

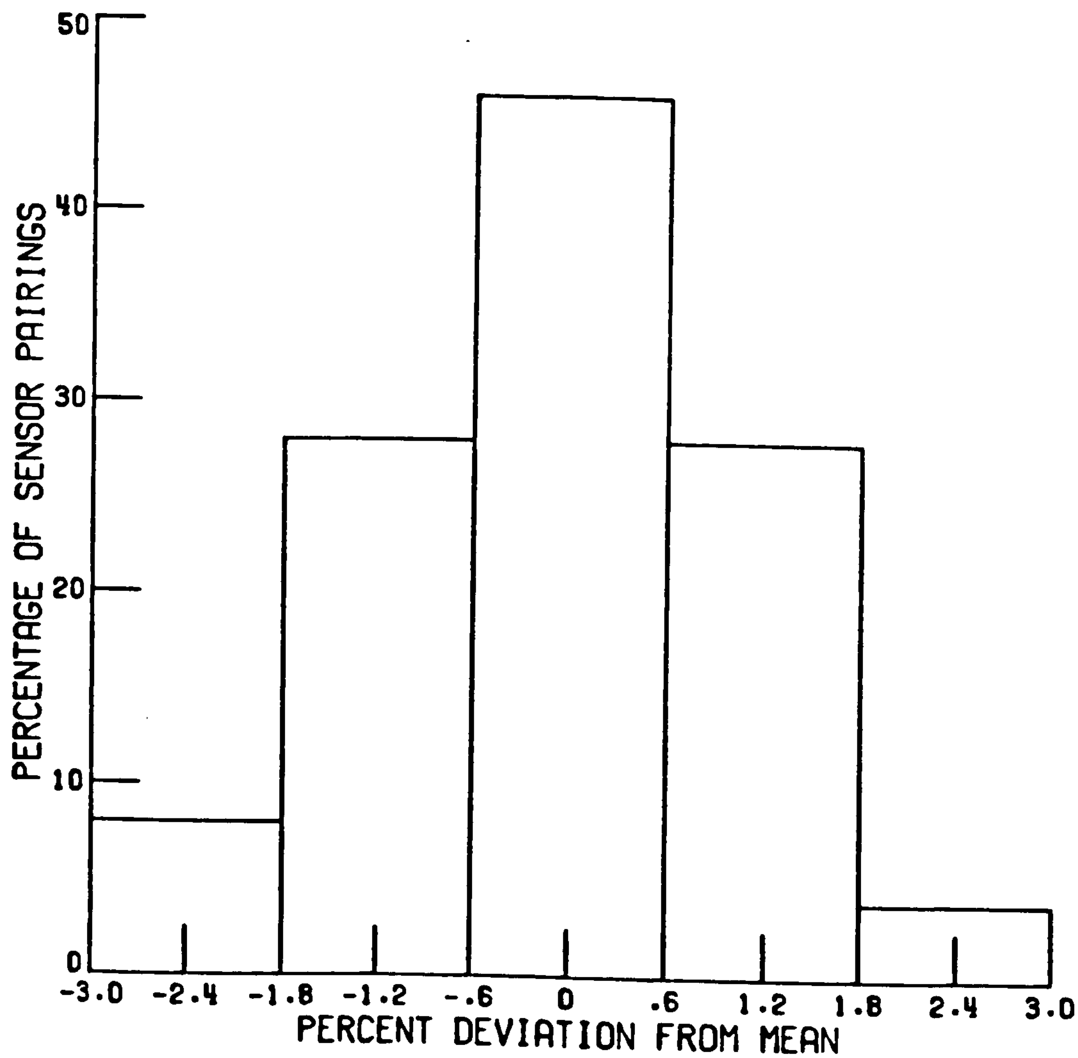

Figure 4: P_w Variance

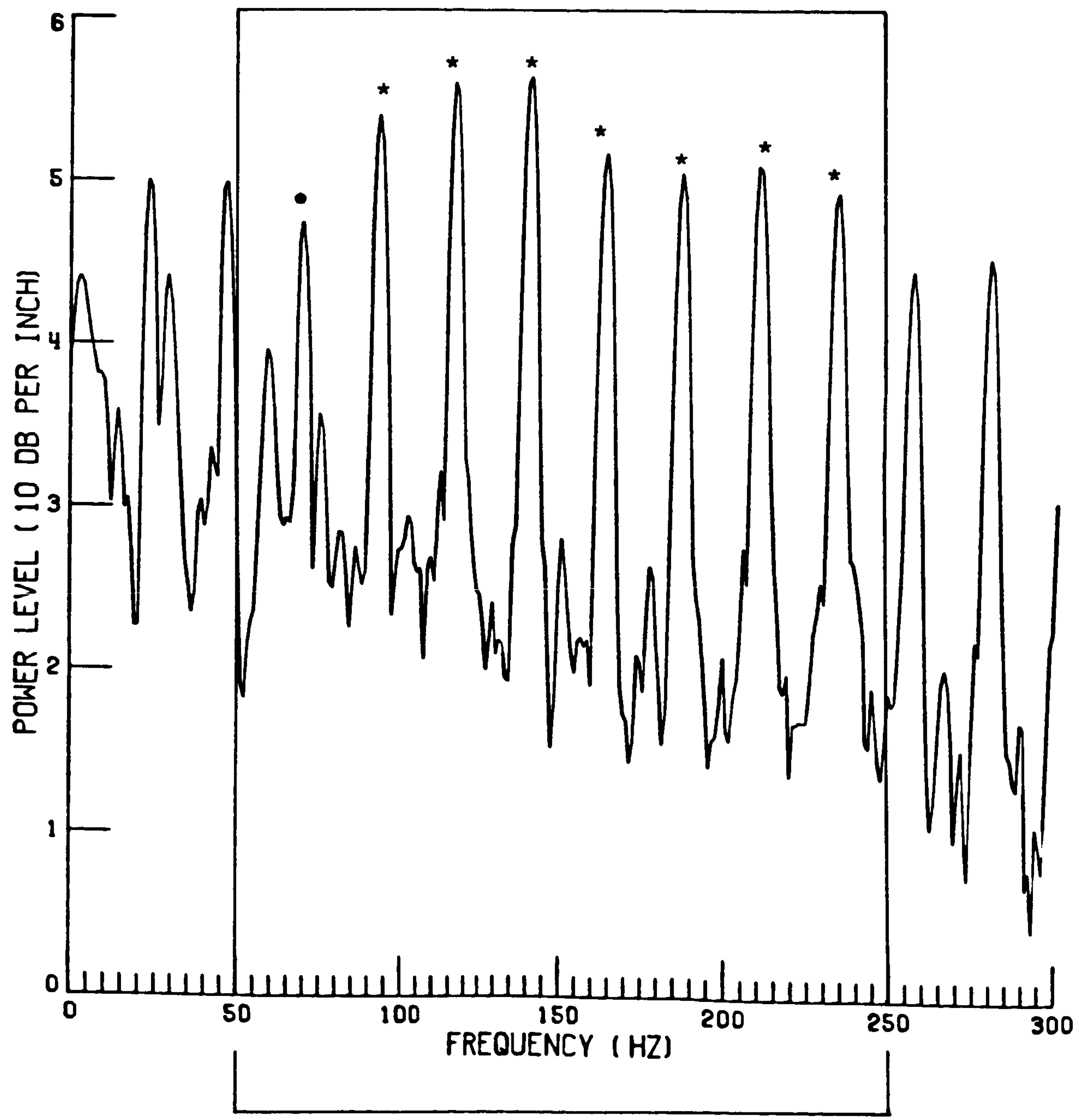

Figure 5: Spectrum Magnitude
(*: Peak Power)

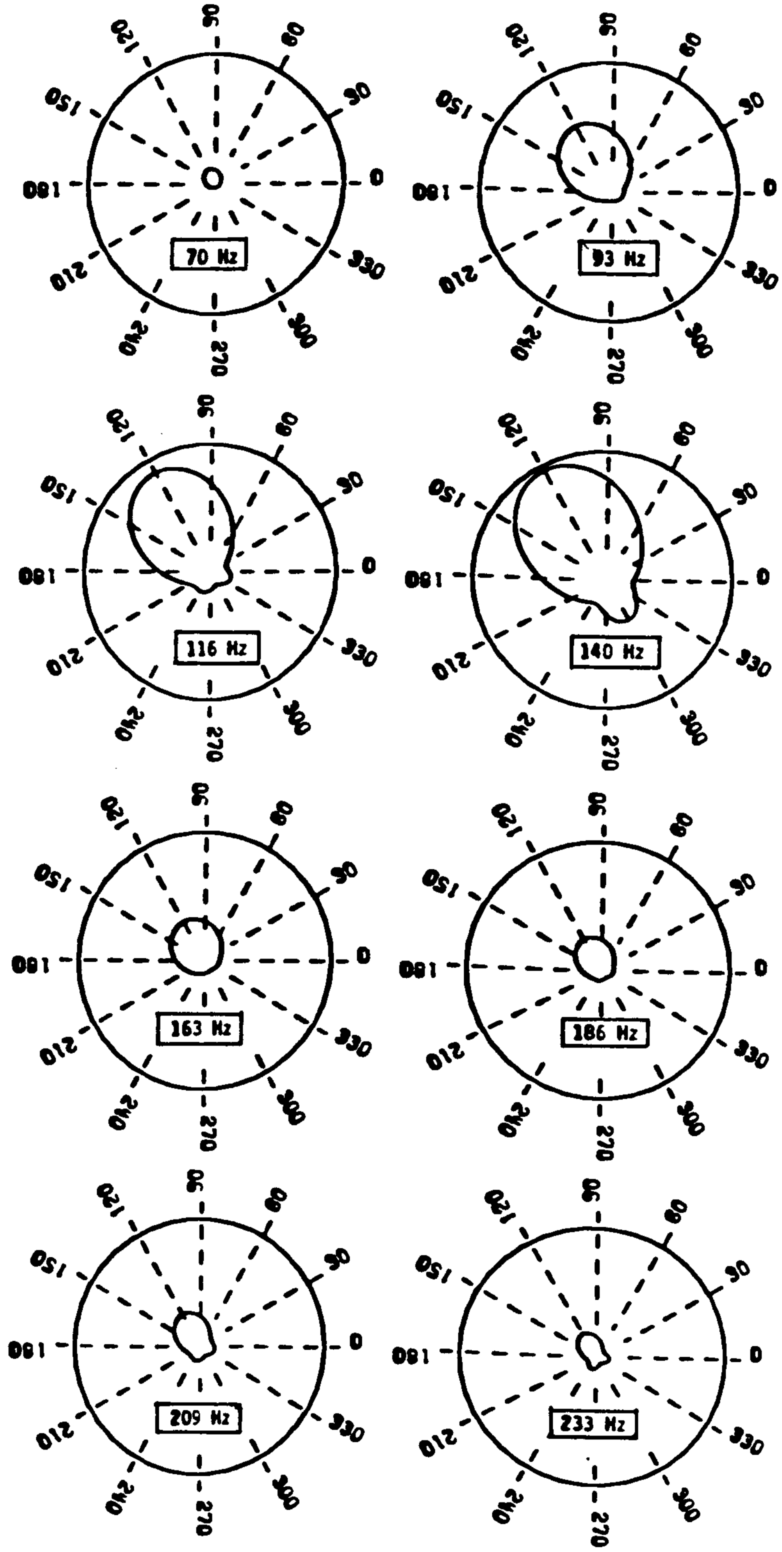

Figure 6: Source Bearing

CONCLUDING REMARKS

This paper has presented the mathematical derivation of a new acoustic source bearing estimation method, the ASAT algorithm, and described the acoustic array processing computer program, ASBE, which was developed to use it. By incorporating the high frequency resolution of the Fast Fourier Transform (FFT) and the bearing estimation capability of Capon's Maximum Likelihood Method (MLM), ASAT will estimate the bearing of an acoustic source while minimizing the effects of foldover aliasing and spatial aliasing. Unlike the MLM, ASAT does not require the inverse of the sensor array's cross power spectral density matrix. This feature allows the analysis of data which may yield a singular matrix.

Using the ASBE program, a benchmark test case consisting of 60 runs was analyzed and the results show a high accuracy in locating the sound source. Polar bearing plots from the first run show how sidelobes became more apparent when a subset of the sensor data was used in the analysis. Since the highest frequency to be analyzed is usually not known, this sidelobe effect is an acceptable alternative to deleting those frequencies which create spatial aliasing.

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1. Capon, J.: "High-Resolution Frequency-Wavenumber Spectrum Analysis", Proc. IEEE, vol. 57, pp. 1408-1418, August 1969.

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15. Supplementary Notes

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16. Abstract

This paper documents a new bearing estimation algorithm (Acoustic Source Analysis Technique - ASAT) and an acoustic analysis computer program (ACOUSTIC SOURCE BEARING ESTIMATION-ASBE) which were developed by Computer Sciences Corporation for NASA Langley Research Center. The ASBE program is used by the Acoustics Division/Applied Acoustics Branch and the Instrument Research Division/Electro-Mechanical Instrumentation Branch to analyze acoustic data and estimate the azimuths from which the source signals radiated.

Included in this document are the input and output from a benchmark test case.

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