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ADAPTIVE ARRAY PROCESSING FOR SOURCE LOCATION IN FAST REACTORS

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1. INTRODUCTION

Acoustic surveillance techniques are being developed for the detection of extremely unlikely local faults in liquid metal fast breeder reactors. A typical reactor of this type, Fig 1, will have a core consisting of over 500 replaceable units called sub-assemblies. Each sub-assembly contains several hundred power generating fuel pins.

It is planned to monitor the sub-assemblies for incipient local boiling of the coolant using a two-dimensional array of microphones.

An approximate half scale underwater model of the Prototype Fast Reactor (PFR) Core is being used to provide experimental data. Preliminary results, [1], using the delay and sum beamformer and frequency domain beamformer, have shown that acoustic sources may be located to within one sub-assembly.

More sophisticated "adaptive" array processing strategies offer the promise of increased sensitivity of the instrumentation to a fault, increased signal to noise ratio, SNR, in the array output and higher resolution of individual sources.

In this paper the delay and sum beamformer and the frequency domain beamformer are compared with a more sophisticated adaptive technique, which is first described. The experiment and data analysis are outlined, and results using the three techniques presented.

2. THEORY

For the purpose of this paper the pressure sampled by the transducer is divided into two components, namely "noise" and "signal". Define the "signal" to be the acoustic signal produced by the fault condition (which may or may not be present). Define the "noise" to be everything else, that is flow noise, transducer self-noise, pump noise etc. The purpose of the signal processing is first of all to flag when a "signal" is present and subsequently to characterise the sources. For example the number and positions of the sources and the individual signal spectra are required.

To obtain the most benefit from the signal processing strategy, we must make as much use as possible of any <u>a priori</u> information that may be available. For example, we know that the signal and noise components have different frequency content, pump noise for example being dominant at lower frequencies. Frequency discrimination, either by pre-filtering or by using a frequency domain beamformer, is therefore an important component of the signal processing. Prior to some incipient fault, there is a relatively long period during which the statistics of the noise, in particular the auto- and cross-spectral densities at the sensors, can be measured. Given these spatial and temporal characteristics, we require the signal processing algorithm to discriminate against that noise optimally.

Adaptive array processing can significantly suppress the contribution from noise with high spatial correlation. Unpredictable physical features such as entrainments of gas or vapour within the reactor core may cause high signal attenuation at some sensors. We require the additional capability to suppress contributions from sensors with low SNR.

An adaptive array processing strategy which satisfies these requirements is the orthogonal beamforming or eigenvalue decomposition technique (see for example Owsley [2], Bienvenu and Kopp [3]). This technique is summarised below.

Define the following quantities:

a) Estimated cross spectral density matrix $\underline{R}$. In practice there are a finite number of matrices $\underline{R}$ at discrete frequency values f. Each matrix has dimensions n x n (where n is the number of sensors) and is positive definite. The matrix is subject to statistical variability due to the finite length of sample used to estimate it.

b) Estimated cross-spectral density matrix $\underline{N}$ for the noise (again positive definite).

c) Steering vector $\underline{p}$. This vector contains the phase lag information corresponding to the current array focal point. It is a function of frequency.

The array output power at frequency f for a non-adaptive beamformer may be expressed as

$$P(f,\underline{p}) = \underline{p}^H \underline{R}\, \underline{p} \qquad (1)$$

where the superscript H denotes the conjugate transposed.

Generally adaptive methods replace expression (1) by

$$P(f) = \underline{w}^H \underline{R}\, \underline{w} \qquad (2)$$

where $\underline{w}$ is a "weight vector". The relative amplitudes and phases of $\underline{w}$ are now chosen to optimise some measure of the array performance subject to some constraint on $\underline{w}$.

In the orthogonal beamforming approach we require to maximise the total array output power subject to the constraint that the array response to the "noise" is unity, i.e.

$$\underline{w}^H \underline{N}\, \underline{w} = 1 \qquad (3)$$

The maximum value of P(f) may now be shown to be given by

$$P_{max}(f) = \lambda_1 \quad (4)$$

where λ_1 is the maximum eigenvalue of the equation

$$\underline{R}\,\underline{w} = \lambda\,\underline{N}\,\underline{w} \quad (5)$$

The optimum weight vector is the eigenvector $\underline{w}_1$ corresponding to λ_1.

If there is no "signal" present and furthermore $\underline{R}$ and $\underline{N}$ are estimated exactly then $\underline{R} = \underline{N}$ and all of the eigenvalues of equation (5) are equal to 1. This does not happen in practice because of sampling errors and in this case the eigenvalues are clustered randomly in the neighbourhood of 1. If a single source "switches on", one eigenvalue emerges from this cluster and assumes some larger value. Thus λ_1 may be used in the "detection" role and some detection threshold set. To "locate" a single source the correlation

$$\eta(\underline{r}_F) = |\underline{p}^H\underline{w}_1|^2 / \left\{(\underline{p}^H\underline{N}^{-1}\underline{p})(\underline{w}_1^H\underline{N}\underline{w}_1)\right\} \quad (6)$$

may be tabulated or plotted over all candidate source positions $\underline{r}_F$ (N.B. $\underline{p}$ depends on $\underline{r}_F$). The maximum is located in order to give an estimated source position.

3. HALF SCALE MODEL OF THE CORE

The model consists of 61 brass tubes with hexagonal sections 70 mm across the flats and 1.67 m long. These simulate the sub-assembly wrapper tubes and are assembled on a rig support stool in a 10,000 gallon tank filled with water. Several layers of air filled 'bubble' plastic material are wrapped around the model to simulate in a qualitative way the acoustic effect of the thermal insulation in PFR. Fig 2 shows a photograph of the completed assembly, with the acoustic array positioned above.

The acoustic array consists of 18 hydrophones mounted in 3 metre long 8 mm diameter stainless steel tubes. These are positioned in a two-dimensional array 85 mm above the model in the position illustrated in Fig 3. A hydrophone was used as a sound source. This is denoted by S in Fig 3. Only twelve of the eighteen hydrophones may be used at any one time due to the limited number of analogue to digital recorders available.

4. DATA ACQUISITION SYSTEM

The data acquisition system digitised and stored data from twelve channels simultaneously. Total record lengths of approximately 1s were stored for each channel at a sample internal of 40μs.

The acoustic source was driven from a signal generator through a power amplifier. The resulting signals from the hydrophones pass through low noise preamplifiers to main amplifiers. After amplification the signals were digitised by analogue to digital converters controlled by the PDP 11-04 computer. The digital records were then transferred to the PDP 11-44 computer for analysis.

5. EXPERIMENTAL PROCEDURE AND RESULTS

The array was commissioned as described in [1]. Only ten of the twelve channels of data were used in the analysis due to equipment failure. In this experiment the ability of the techniques to locate a continuous broadband noise source in the presence of background noise was investigated. The hydrophone shown in Fig 3, position S , was driven with continuous broadband noise to simulate the advanced stages of boiling where pulse rates are high. The driving signal was low pass filtered at 8 kHz to reduce spatial aliasing. A second broadband noise source was placed in the tank outside the model sub-assemblies.

Data from the source signal and source signal plus background noise were stored at a sample interval of 40μs giving a total record length of 983 ms for each channel. The data have been analysed by scanning over a one metre square area in the plane of the source. The resulting matrix of correlation values obtained from each of the techniques are written to files for subsequent display and plotting.

For relatively high signal to noise ratios both the time domain and the frequency domain beamformers have been shown [1] to give clear detection and accurate location of the source.

The comparisons presented here are for low signal to noise ratios at each sensor (~ -10dB) which is close to a realistic limit of the capability of the time domain beamformer. The results given are based on 400 data values corresponding to a record length of 16 msec. This small number of data points was chosen to facilitate a comparison between the different techniques, a limit imposed essentially by the computing requirements of the time domain beamformer.

Figures 4 to 6 are contour and isometric plots from the results obtained from each of the three techniques. The contour plots Figures 4a, 5a and 6a show only the very top of the source distribution for clarity. The contours are plotted at 0.98, 0.96 and 0.94 etc of the maximum correlation value.

All three techniques produce source locations within a sub-assembly diameter of the true source position. The time domain beamformer contour plot Fig 4a shows two main peaks, one in the sub-assembly containing the source, the other in the adjacent sub-assembly. The position of the maximum correlation value lies in the sub-assembly containing the source. The frequency domain beamformer contour plot Fig 5a shows a clear indication of the source location. The orthogonal beamformer plot Fig 6a shows a peak spread over both the source sub-assembly and adjacent sub-assembly. The associated isometric plots Figures 4b, 5b and 6b provide a useful comparison between the results. The time domain beamformer and frequency domain beamformer have low maximum correlation values. This is because the experimental data have an average signal to background noise level approaching the limit detectable by these two techniques. This is illustrated in Figures 4b and 5b. In Fig 4b, the correlation peak is lost in the background, and in Fig 5b the peak is only just discernable. A comparison of the source signal to background noise is given in Table 1, the source level averaged over all transducers is approximately 9dB below the background noise and about 7dB lower than the background on the best single detector (channel 2). We would expect enhancement in signal to background noise level to be given approximately by the square root of the number of transducers, which in this case corresponds to 10dB.

The orthogonal beamformer is much more successful in separating the signal from the background noise. This is illustrated in Fig 6b, where the correlation peak can be easily distinguished above the background.

6. CONCLUSIONS

1. The techniques investigated are all capable of locating the acoustic sources in the presence of background noise.

2. Successful location was achieved using only 400 data points corresponding to a record length of 16 msec. The frequency domain techniques are capable of processing much larger quantities of data points, yielding increased sensitivity and improved resolution.

3. The more sophisticated orthogonal beamformer offers the advantage of clearer discrimination against background noise.

References

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[2] N L Owsley: 'Adaptive Data Orthogonalisation'. Proc IEEE ICASSP, Tulsa, Okla., April 1978, pp109-112.

[3] L Kopp, G Bienvenu and M Aiach: 'New approach to source detection in passive listening.' Proc IEEE ICASSP, 1982 (Paris), pp779-792.

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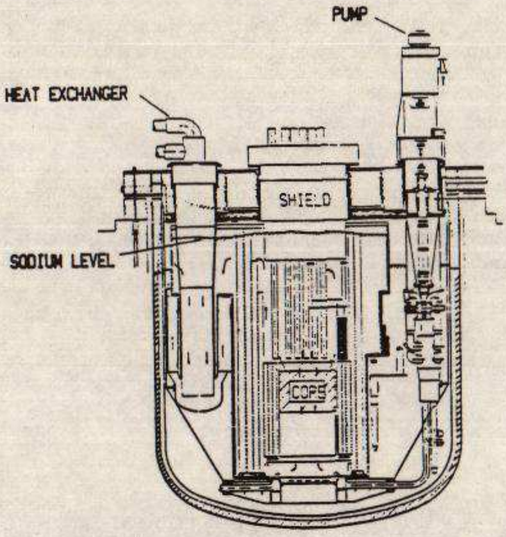


FIGURE 1 : LIQUID METAL COOLED FAST BREEDER REACTOR

FIGURE 2 : PHOTOGRAPH OF MODEL OF CORE

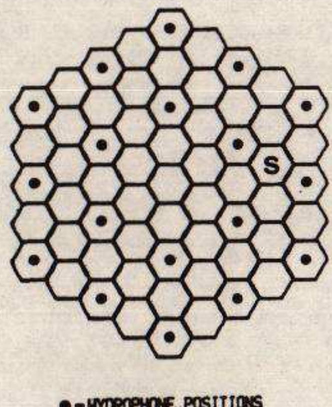
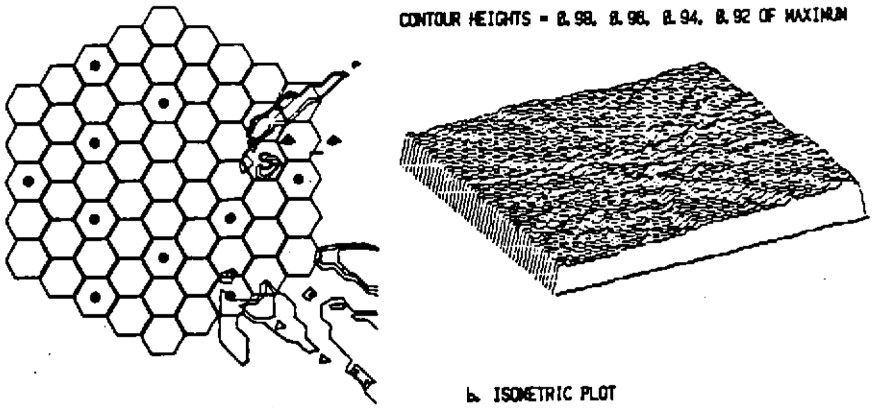


FIGURE 3 : HYDROPHONE ARRAY AND SOURCE POSITION

MAXIMUM CORRELATION = 0.147

CONTOUR HEIGHTS = 0.98, 0.96, 0.94, 0.92 OF MAXIMUM



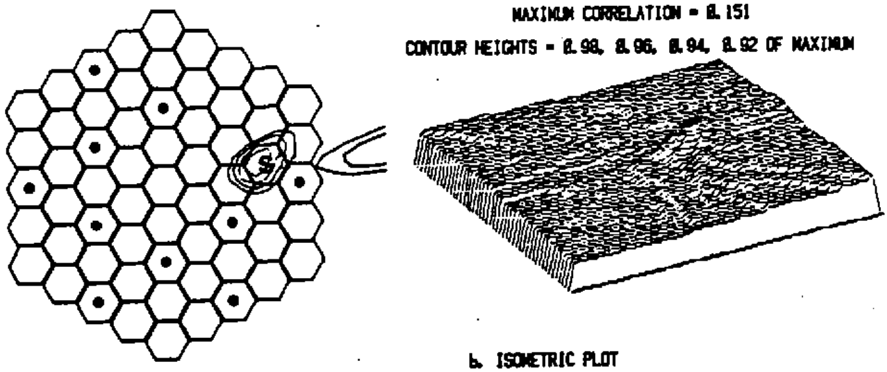
b. ISOMETRIC PLOT

a. CONTOUR PLOT

FIGURE 4 : PLOTS SHOWING THE OUTPUT FROM THE TIME DOMAIN BEAMFORMER FOR A BROADBAND SIGNAL IN THE PRESENCE OF BROADBAND BACKGROUND NOISE

MAXIMUM CORRELATION = 0.151

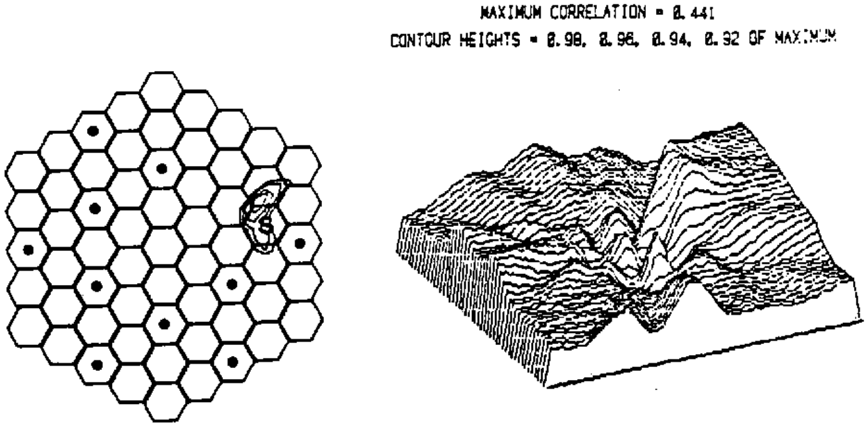
CONTOUR HEIGHTS = 0.98, 0.96, 0.94, 0.92 OF MAXIMUM



b. ISOMETRIC PLOT

a. CONTOUR PLOT

FIGURE 5 : PLOTS SHOWING THE OUTPUT FROM THE FREQUENCY DOMAIN BEAMFORMER FOR A BROADBAND SIGNAL IN THE PRESENCE OF BROADBAND BACKGROUND NOISE



a. CONTOUR PLOT

b. ISOMETRIC PLOT

FIGURE 6 : PLOTS SHOWING THE OUTPUT FROM THE ORTHOGONAL BEAMFORMER FOR A BROADBAND SIGNAL IN THE PRESENCE OF BROADBAND BACKGROUND NOISE

CHANNEL NUMBER	AVERAGE SIGNAL TO BACKGROUND NOISE (dB)
1	-9.7
2	-6.8
3	-8.5
4	-8.7
5	-9.2
6	-11.2
7	-9.5
8	-8.3
9	-12.3
10	-8.75
AVERAGE	-9.1

TABLE 1 : AVERAGE SIGNAL TO BACKGROUND NOISE AT EACH HYDROPHONE FOR A BROADBAND NOISE SOURCE WITH BROADBAND BACKGROUND NOISE