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NOISE SOURCE IDENTIFICATION AND MULTIPLE SOURCE IMAGING TECHNIQUES

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INTRODUCTION

A number of different techniques may be used to identify noise sources in industrial situations, but in many cases the simpler methods such as spectral identification or the measurement of the sound power of each source are insufficient or impractical. In recent times correlation and coherence techniques have been available but the results have not always been convincing due to processing limitations and difficulties in interpreting the data. With the increased power of modern micro-computers and data acquisition systems many of the problems that were encountered with these techniques may be overcome and they may be developed to give greater precision. At the University of Compiègne partial coherence and selective intensity techniques have been applied and developed for industrial noise source identification problems since the late seventies and more recent work has concentrated on different types of estimators which may be applied to improve the accuracy of the identification in specific circumstances. This paper describes the results of some simple experiments that were carried out to show the effects of using different types of estimator to calculate the transfer functions and the contribution of each source to the sound pressure and the sound intensity at a particular point when several sources are present. In each case these methods require a reference signal representing each source which is generally obtained using a microphone in the nearfield of the source, or an electrical signal from the generator when loudspeakers are used . The problem is then reduced to one of attributing the measured input signals to each source when their contributions are not independent which is generally the case when microphone reference signals are used,and then correctly calculating the transfer functions between each source reference point and the output point. In the situations where the output of interest is the sound pressure a single output. microphone is sufficient,but if the sound intensity is required two, four, or even six output measurement points may be required to analyse the intensity in one, two or three dimensions. These methods have also been developed further at Compiègne to produce selective acoustic arrays for multiple source identification. The problems involved in these techniques are generally more difficult when the sources are in a reverberant environment so the experiments were carried out in relatively reverberant conditions.

TRANSFER FUNCTION ESTIMATORS

When a single-input single-output system is used to model a noise transmission problem the transfer function is usually calculated using the H_1 estimator which eliminates the effects of extraneous noise at the output using the cross and auto spectral densities of the input and output signals.

$$H_1 = \frac{G_{xy}}{G_{xx}} \qquad (1)$$

G_{xy} is the cross-spectral density and G_{xx} the auto-spectral density of the input When the input signal is contaminated by extraneous noise uncorrelated with the output signal the resulting increase in the input auto-spectral density gives an underestimate for the transfer function using this etimator. The coherent output power (COP) which measures the contribution of the source at the output will also be underestimated in the same proportions because of this error.

$$COP = |H_1|^2 . G_{xx} \qquad (2)$$

The coherence function is given by the division of the COP by the auto-spectral density of the output and gives a measure of the accuracy of the identified source model for the system. The value of the coherence function is unity in situations where a single uncontaminated source contributes to give a single uncontaminated output. If an external source which is uncorrelated with the input also affects the output point the coherence will be reduced in proportion to the contribution of this second source,but the COP will remain accurate as an estimator for the contribution of the first source. In real measurements there is always a degree of contamination of the source reference signals which leads to errors in the calculated source contributions.
One way of overcoming the effects of noise at the input is to inverse the form of calculating the transfer function working from the output to the input. This gives the H_2 estimator,but although it works well if there is only one source and extraneous noise at the input only,it is impossible to apply in multiple source situations.
A method which takes account of the effects of noise at the input and the output is the H_v estimator which assumes an equal level of extraneous noise at the input and the output points.

$$H_v = \frac{G_{yy} - a}{G_{yx}} = \frac{G_{xy}}{G_{xx} - a} \qquad (3)$$

The output spectral density G_{yy} is reduced by the supposed contribution of the extraneous noise a and the input spectral density by the same amount leaving an expression which may be solved for a and H_v. The value for H_v is obtained from the lowest solution for a which corresponds to the lowest eigenvalue of the specral matrix $|G_{xxy}|$ which in a single input case is given by $\begin{bmatrix} G_{xx} & G_{xy} \\ G_{yx} & G_{yy} \end{bmatrix}$

Multiple input versions of the H_1 estimator and H_v estimator exist which enable the effects of correlated inputs to be taken into account providing that source reference signals can be obtained for each source that is correlated with others to construct a correct model of the interactions between each of the inputs to the system model.The partial coherence techniques which are derived from the multiple input version of the H_1 type of estimation enable an analysis to be made of the effects of different possible types of interaction that could take place between the inputs and aid in the interpretation of the results.
Another type of estimator that can be used in multiple input system analysis is the virtual coherence or principal component type of analysis where the mutually uncorrelated components in the input signals are separated out by using the

cross spectral matrix of the input signals and solving the eigenvalues and the associated eigenvectors which may then be used to diagonalise the matrix for each frequency. The diagonalised matrix may then be used directly in association with the eigenvector matrix to calculate virtual transfer functions and virtual output contributions for these virtual sources. The expressions relating the cross spectra between the inputs 1 and 2 for a two input one output system, the cross spectra between the inputs and the output and the two transfer functions H_{1y} and H_{2y} are given by;

$$\begin{matrix} G_{1y} \\ G_{2y} \end{matrix} = \begin{bmatrix} G_{11} & G_{12} \\ G_{21} & G_{22} \end{bmatrix} \begin{matrix} H_{1y} \\ H_{2y} \end{matrix} \qquad (4)$$

The eigenvalues of the input matrix yield an eigenvector matrix which ,when postmultiplied by the transfer function vector of the virtual system is equal to the true transfer function vector of the system under examination. This technique has supposedly certain advantages in defining the number of mutually independent inputs to the system but the physical interpretation of the virtual system obtained is difficult.|1| It may however,prove possible to develop this technique to give greater accuracy than partial coherence methods in situations where the inputs are contaminated by external noise and particularly when the correlation between the inputs is high.

APPLICATIONS TO SOUND INTENSITY

The techniques that have been discussed previously may be applied directly to source identification problems or to calculate the transfer functions required in multiple input modal analysis. An alternative approach is to use such methods to improve other experimental techniques such as sound intensity measurements or acoustic arrays. Such methods have been developed and applied to noise transmission and radiation measurements at Compiègne over the last five years and have proved their value in situations where the existing methods fail.
The sound intensity may be calculated directly from the cross spectral density of two microphone signals using the method introduced by Fahy.|2|

$$I = -\frac{\mathrm{Im}(G_{12})}{\rho w d} \qquad (5)$$

where $\mathrm{Im}(G_{12})$ is the imaginary part of the cross spectral density and ρ the densityof air. The angular frequency is denoted by w and the line joining the acoustic centres of the microphones defines the direction of the acoustic intensity vector. The microphone spacing d is chosen to be inferior to a sixth of the wavelenth to minimise the errors of approximation of the pressure gradient by finite difference which are implied in the formula.
In a multiple source situation the preceeding expression is not capable of discriminating between the intensity components produced by each source which can lead to difficulties in interpretation when the measurements are analysed. The contribution of each source may be analysed directly by noting that the cross spectral density due to a source s may be expressed in terms of the transfer functions between the source and the two microphones. The expression is given here in its simplest form which assumes that the inputs to the system are

uncorrelated enabling a single input single output model to be applied for each microphone of the intensity probe.

$$(G_{12})_s = H^*_{s1}.H_{s2}.G_{ss} = H^*_{s1}.G_{s2} \qquad (6)$$

where * denotes the complex conjugate and the transfer functions between the source and the two microphones are given by H_{s1} and H_{s2} respectively. The initial developements of this method assumed that the transfer function was calculated using the H_1 method but the first form of equation (6) may be developed in terms of other types of transfer function estimators and extended to the multiple input case for correlated inputs. In comparison with the H_1 type of estimate the other techniques should give more accurate results in certain situations.

EXPERIMENTAL PROCEDURE

A series of simple experiments was devised to test the effects of modifications to the methods of analysis and the effects of changes in the signal processing parameters. Although these methods have been developed and applied for a number of years by the authors to industrial problems the significance of any improvements obtained by modification of the estimators is easier to analyse in the closer controlled conditions of a laboratory experiment. The tests were carried out using one or two sources (loudspeakers) in one of the rooms of our timber frame acoustic transmission measurement facility. Direct acquisition of the signals was possible using a microcomputer based signal processing system which performed the calculations necessary for the different types of estimator using programmes developed by our research team. Reference signals were obtained from the generator signals feeding the loudspeakers or by using microphones close to the loudspeaker to represent each source. Measurements representing the output of the radiation and transmission problem were made either inside the source room or on the other side of the outer wall using an acoustic intensity probe. Comparisons of sound pressure contributions or sound intensity contributions of each source at each output point could be made and compared with the results obtained when one of the sources was switched off.
The room used for these experiments was relatively small but was designed to be comparable with a typical timber frame home.The reverberation time of this room as tested was superior to 1s. and the timber frame walls were lightly damped adding to the difficulties in modelling the transfer of energy by the acoustic and vibrational transmission paths. In order to optimise the results in such conditions the Fourier transforms used to calculate the cross spectra used in the model should be as long as possible to include as many of the reflections as possible in the calculated transfer functions. For this reason most of the data obtained in this experiment was processed on blocks of 8192 points.

EXPERIMENTAL RESULTS

The analysis of the effects of transform length on the results of the calculated output power may be initiated by comparing the results obtained at the same sampling frequency but in one case with 2048 points and in the other case with 8192 points. A single source was used in the room and the output point was a microphone on the other side of the outer wall. The generator signal was used as a reference in this test,the results of which are illustrated by the coherence functions shown in figure 1.

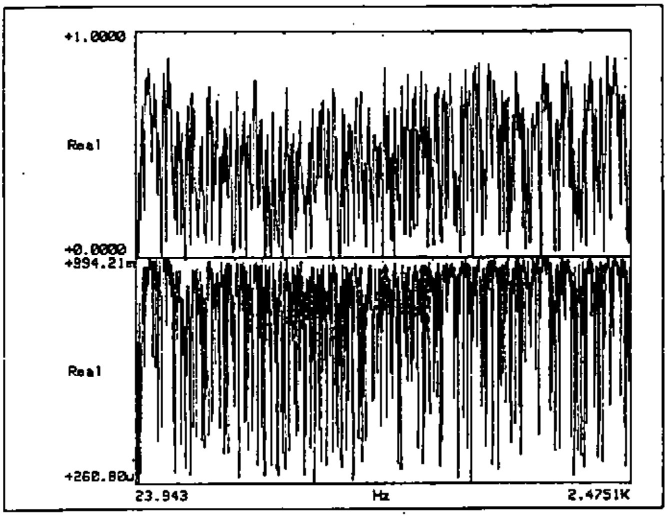


Figure 1. Coherence obtained using a) 2048 & b) 8192 points

The increase in the coherence brings it up to the limiting value of unity as the accuracy of the model is increased.The coherence was further improved by using a microphone in the reverberant field thus shortening the path length,but the number of points could not be increased any further.
Figures 2 and 3 illustrate the same effects on the sound intensity which is measured using the selective technique and compared with the direct measurement.

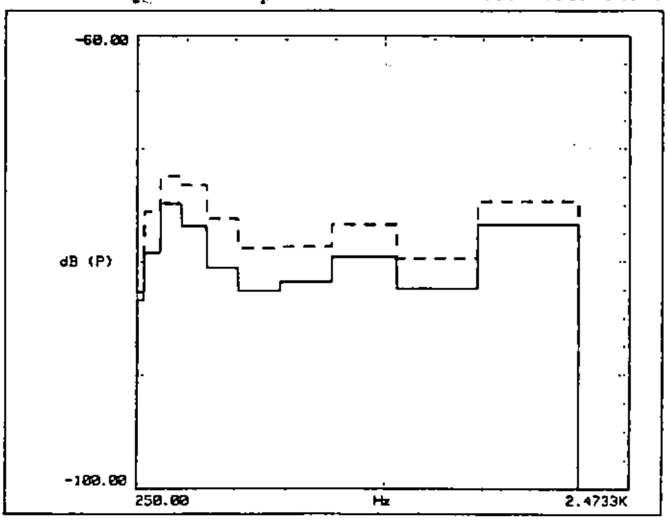


Figure 2
Selective and direct sound intensity on outer wall using 2048 points.
1/3 octave plot
ref. dB 1 W/m^2

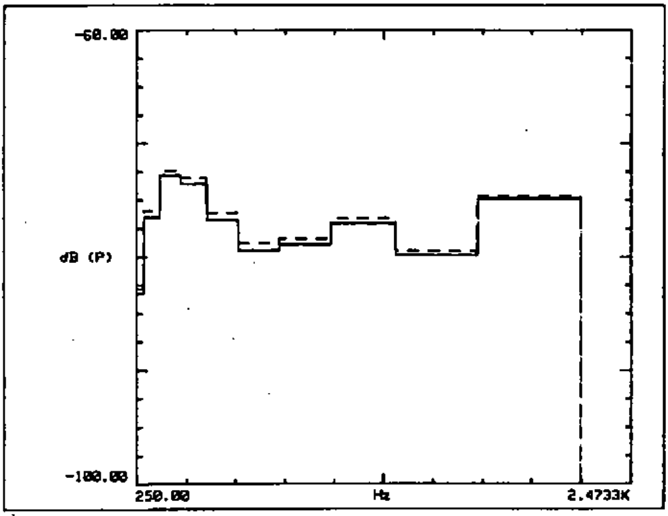


Figure 3
Selective and direct sound intensity on outer wall using 8192 points
1/3 octave plot
ref. dB 1 W/m^2

The selective technique (full line) underestimates the true result given by the direct method (broken line) by 2-3 dB in the first case and .2-.6 dB in the second. When the second source is switched on the direct intensity increases by 12-15 dB but the selective intensity remains unaffected illustrating the noise rejection capacity of this technique.(Figure 4) The technique can therefore be used in transmission measurements in noisy conditions but industrial source identification problems require the use of microphone or accelerometer reference signals.

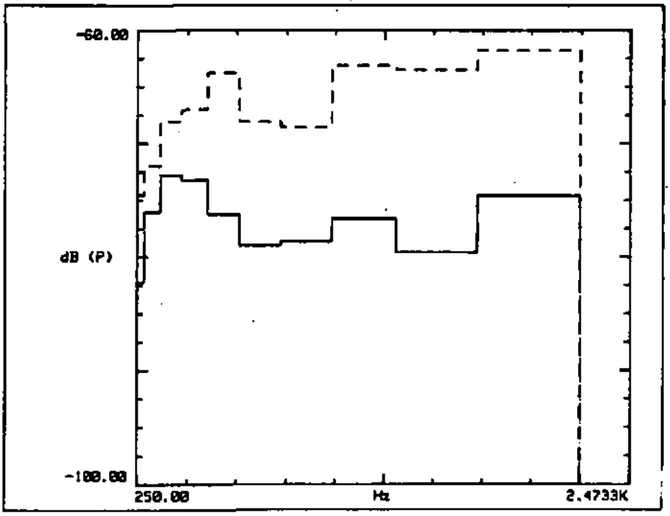


Figure 4
Selective and direct sound intensity on outer wall using 8192 points
1/3 octave plot
ref. dB 1 W/m^2
2nd source active

In order to illustrate the problems of contamination using microphones as source references experiments were carried out inside the room in the reverberant field

The calculated contribution of the first source at the output point in the room is compared in Figure 5 with the true contribution obtained when the second source is switched off.(lower graph) The transfer function is correctly calculated by partial coherence techniques but the inter source contamination results in an over estimate of the source contribution . This multiple input COP is a good estimate of the maximum possible contribution of the source. In order to obtain a better estimate the origins of the captured source reference signals must be identified to discover which part is due to contamination from the other source.

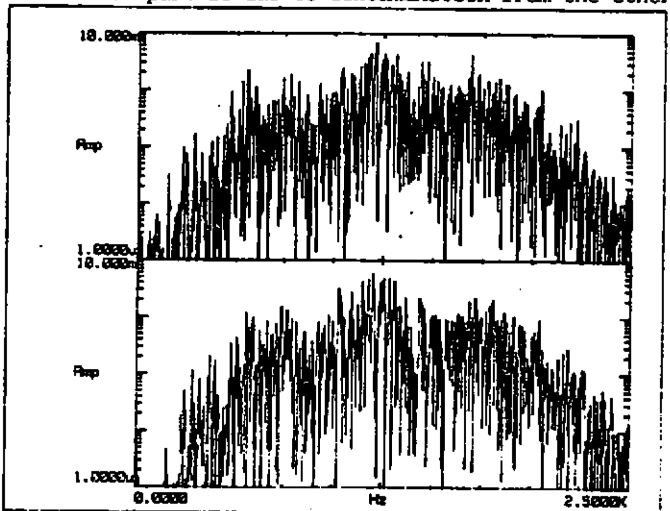


Figure 5
Source contribution both in action (upper)
Source contribution alone (lower)
arbitrary log scale

The virtual input technique was tested to try to separate the input contributions (Figure 6)

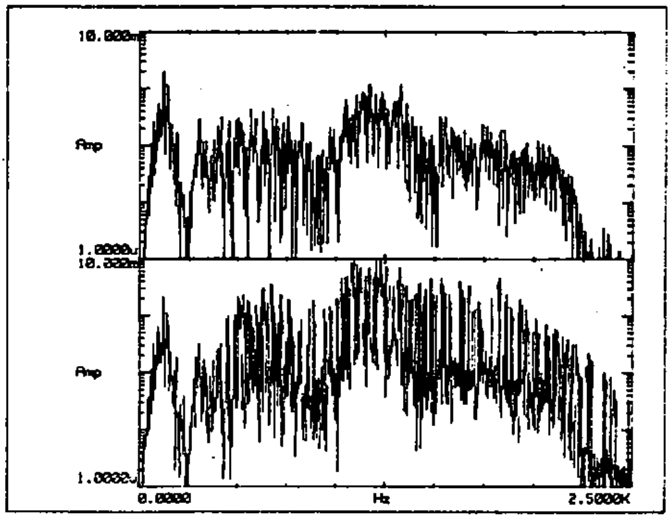
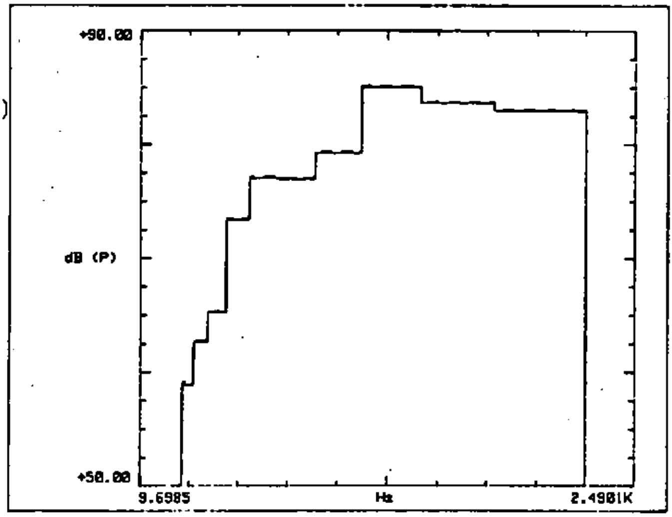


Figure 6
Input signal source 2 alone(upper)
Virtual input 2 both sources active(lower)

It can be seen in Figure 6 that the input spectrum is recognisable in its form but that breakthrough to a higher level has occured due to the stronger source contaminating the input signal.The two sources were in fact independent but the sorting of the mutually uncorrelated inputs is arbitrary at each freqency.
A last result of interest is that obtained using the H_v estimator for sound intensity.It was hoped that the effects of too short a transform length could be corrected by using this technique. The result shown in Figure 7 shows a small improvement but not as much as expected. This is partially due to the fact that the errors due to coherence losses are smaller inside the room than outside because of the shorter path lengths involved.

Figure 7
Sound intensity using H_v estimator (broken line) 2048 points comparison with H_1 estimator (full line)



The results need to be compared with those obtained in more difficult conditions to see if the improvements expected can be obtained.

CONCLUSIONS

The estimators that have been discussed would seem to have some interesting properties but require further developements to obtain significant improvements over the existing techniques for source identification and selective intensity measurements. Most measurement situations however are not as difficult as those experienced here in terms of source reference contamination.

REFERENCES

|1| PRICE, S., BERNHARD, R., Virtual coherence: a digital signal processing technique for incoherent source identification. Proc. 4th Int. Modal Analysis Conf. pp.1256-1262 1986

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