Proceedings of the Institute of Acoustics

A REVIEW OF SOURCE IDENTIFICATION AND QUANTIFICATION TECHNIQUES

F.J. Fahy   Institute of Sound and Vibration Research, University of Southampton, Southampton, SO17 1BJ, UK

1. INTRODUCTION

The objectives of the techniques reviewed here are to locate, identify and evaluate the principal sources of unacceptably high levels of airborne noise in a specific region or at a specific location. The principal purposes of their application are to aid the selection and application of appropriate, cost-effective, noise control measure; to place multiple sources in rank order; to identify responsible agents; to evaluate the effectiveness of applied control measures; and to predict the noise in regions other than the measurement region. The paper does not address the subject of techniques developed for underwater application.

'Source identification' implies a process by which the physical origin of a noise, or a component thereof, is established. There is a degree of ambiguity attached to the term, because radiated sound is usually the final stage in a complex chain of causes and effects. Take the example of the noise radiated by the cover over a gear box in a car. Clearly, radiation of the cover is the immediate source of noise. But this is caused by the vibration of the gearbox casing which is caused by the unsteady forces applied by the various gear shafts to the bearing housings, which in turn are caused by the time variation of inter-tooth forces as they engage and disengage, which in turn (literally), are caused by the torque applied by the cylinder pressures forces, which is caused by combustion, which is caused by mixing air and fuel and burning it. Logically, then, the fuel is the primary cause of the noise and should not be used. This logical *reductio ad absurdum* illustrates one of the major questions which must be addressed by the source seeker: How far back down the chain should one go?

This paper deals principally with techniques designed to identify sources of radiation from the 'outer skin' of a noise generating system. It begins with an attempt to address the fundamental question of how to decompose spatially extended sources into independent components, with special attention paid to the difference between 'coherence' and 'correlation'. It continues with a review of devices and procedures which are available for locating sources of sound radiation under free field conditions, together with comments on their merits and shortcomings. A brief critical account of coherence-based methods of system identification is followed by a review of inverse techniques for allocating source strengths to pre-specified source models. A particular form of inverse method, termed 'Nearfield Acoustical Holography', is then introduced and discussed. Finally, the importance of characterising sources by means of parametric dependence studies is emphasised. It should be understood that the limitations of space preclude the presentation of a comprehensive, in-depth critical analysis of the many techniques and systems available for the task; however, it is hoped that the paper may promote awareness and serve as a jumping off point for those who are fairly new to the challenge of the 'hunting the source'.

2. SOURCE DEFINITION

2.1 A General Criterion for Source Definition

The principal purpose of source identification in an engineering context is to guide the process of specifying and applying noise control measures. Quantification is required for a wider range of reasons including customer information, compliance with regulations, source labelling, and prediction of impact on the environment and on health. For the purpose of identification it makes sense to define an individual noise source in terms of its degree of independence from other simultaneously operating sound generating mechanisms. This criterion arises from the principle of linear superposition. In the case of N time-stationary sources, the mean square pressure generated at any location is given by the sum of N mean square pressures generated by each source *in isolation* plus the sum of all the $N(N-1)$ correlation terms arising from inter-source correlation. If any one source is changed, the effect on the sound pressure level is unpredictable, unless either the inter-source correlation and transfer functions to the observation point are known (extremely unlikely) or the sources are all mutually uncorrelated (statistically independent). On this basis, I define an individual source as 'a highly correlated region of sound generating activity'. In this way, I can be certain that if I reduce its activity by X dB, its contribution to the resulting sound pressure level will fall by X dB. Whether this reduces the overall sound pressure level by a significant amount depends on the relative contributions made by all the other independent sources: this question can only be resolved by appropriate source quantification.

The criteria for resolving the question 'How far back along the chain of noise generating and transmitting processes should one attempt to travel?" are partly technical and overwhelmingly economic. If the ultimate intention is to redesign the whole system, the answer must be " Right back to the fundamental causes of noise generation". If the intention is simply to reduce the noise to an acceptable level by the quickest, simplest and cheapest means possible, it is necessary only to treat the system as a black box of which the strongest outputs should be suppressed.

2.2 Correlation and Coherence

There appears to be considerable confusion in the noise control literature between 'correlation' and 'coherence' ; indeed, some authors treat them as being synonymous, which is not at all the case. The Correlation Function is a real time-domain quantity which is a measure of the 'similarity' of a two time-stationary signals as a function of the time by which one is delayed with respect to the other. If the two signals represent the values of a physical field quantity, such as sound pressure, at different spatial locations, the correlation is called a space-time correlation function. The correlation function may be formed from frequency-band-filtered signals; in which case it is termed a band-limited correlation function.

The Fourier transform of the correlation function is the complex 'Cross-Spectral Density'. This expresses the relative phase between the related components of two signals as a function of frequency. Coherence is a frequency domain function which reveals the degree of statistical relationship between the signals.

The band-limited *zero-time-delay*-correlation function is given by the integral over the frequency band of the cross-spectral density, as illustrated by Figure 1. The most important conclusion is that two signals, or processes, can be *fully coherent* at *each* frequency, but if the phase of the cross spectrum varies widely and irregularly over the frequency band considered, the *correlation can be close to zero*, irrespective of time delay. For example, the coherence between of accelerations measured at two points on a structure driven only by a *single* broad band, or multi-frequency, vibrational source, will be unity; but the band-limited correlation will decrease with both separation distance and bandwidth, provided that no single vibration mode dominates the response, because each mode contributes more or less independently to the phase of the cross spectrum. The spatial extent of a correlated region may be expressed in terms of a correlation scale.

These observations suggest that it is unwise to apply unnecessarily fine spectral resolution in the application of identification and quantification techniques to broad band noise sources.

3. SOURCE LOCATION

The term 'source location' implies the application of some procedure designed to identify the position(s) of the source(s) of an observed effect. Source location is implemented by making observations of the sound field generated by the system under investigation. In cases of sources located in effectively *free field* conditions, observations may be made in both *far field* and *near field* regions (see Section 3.1 below). Where sources are located within enclosed, or partially enclosed, volumes of fluid, far field techniques are inapplicable.

It would appear, at first sight, that the effectiveness of source location would increase as the field observation positions approach the physical radiator. However, this is not necessarily the case because sound fields generated by spatially extended source regions become increasingly complex as the source region is approached, rendering their interpretation increasingly problematic. The form of the region of a sound field in which measurements are made for the purposes of source location and evaluation strongly influences the effort required to implement the technique and the nature of the information generated. More detailed information becomes available as the source is approached, but the increased effort is not always justified.

3.1 Field Regions

The sound field generated by a source system operating in *Free Field* conditions may be divided into three regions - Far Field, Geometric Near Field and Hydrodynamic Near Field. The Far Field is characterised by the following features. The distance from 'source' is much greater than the maximum linear dimension of whole active source region. [In principle, this should relate to dimension of largest correlated source region; but it is not practicable.] The sound intensity is radially directed. The frequency-dependent intensity directivity is governed by phase differences between elemental coherent regions of the source distribution and the path length differences to the field point. The wave field over the aperture of a far field detector which subtends a small angle at the source is effectively uni-directional and plane (i.e. unique source direction).

The Geometric Near Field is the region in which the 'source' subtends a large angle at the observer position so that the range of path lengths from the source surface to the field point is of the same order as the average distance; the contributions of elemental source regions are range dependent. The spatial distribution of the sound field is usually extremely complex. No simple relation exists between intensity and pressure.

There is no general geometric criterion which defines the extent of a Hydrodynamic Near Field: it depends upon the source characteristics. It decays rapidly with distance from the source surface and generally extends only a small distance in terms of a wavelength. The field is dominated by non-propagating, evanescent, components which radiate no sound energy to the far field. The intensity is strongly reactive and the active intensity often exhibits circulation of energy between sources and sinks. The magnitude of the particle velocities greatly exceed $p/\rho c$ and p and u nearly in quadrature. The local sound pressure is controlled predominantly by local source activity (e.g. panel acceleration). Extrapolation of near field intensity into the far field is invalid.

Within volumes of fluid which are either fully or partially enclosed by strongly reflecting surfaces, the far field concept is inapplicable, but, in large volumes, the geometric near field still dominates the direct field and the hydrodynamic near field remains largely intact because it is dominated by non-propagating components which do not produce reflections from any bounding surfaces. Techniques for locating sources under reflective field conditions must be capable of distinguishing 'real' sources from their 'images' in reflecting boundaries.

3.2 Principal Source Location Techniques and Systems

3.2.1 Human Auditory System

The human hearing system is directional in terms of perceived loudness of steady sound above about 300 Hz. However, the auditory signal processing system also uses temporal signal structure to locate sound sources (differential time of arrival and detection of first arrival). The effectiveness of location is strongly influenced by the number and spatial distribution of significant uncorrelated noise sources; the relative strengths of direct and reflected sound; the directional distribution of arriving wavefronts and the autocorrelation function of the signal; all of which influence interaural cross-correlation [1]. In broad terms, the human localisation capacity increases with increase of autocorrelation decay rate (better with isolated (>80 ms) transient or broadband continuous noise in non-reverberant surroundings), although the capacity to localise a single broad band noise source in very reverberant surroundings is remarkable.

The system becomes confused in the geometric near field (many arrival directions), in small reflective environments such as passenger compartments of cars, in large reverberant enclosures containing a tonal or narrow band source or many strong broadband sources. Cupping the ears increases directional sensitivity and excludes lateral and rear arrivals. But, the system does not general provide quantitative information, unless through laborious comparison techniques.
This system should always be used to make an initial survey: don't forget to take it with you!

3.2.2 Temporary screening/covering

In this is a rather obvious technique the whole system under investigation is wrapped in a layer of sound absorbent material under an impermeable covering sheet. Selected areas are sequentially uncovered and the radiated sound levels measured. Covers are increasingly ineffective as frequencies decrease below about 500 Hz Covering parts of correlated source regions can lead to serious error. The technique is extremely time consuming and labour intensive.

3.2.3 Directional Microphones

Individual directional microphones are generally used in the far field under free field conditions where reflections will not confuse the observations. Hence, any source, however complex, appears as a single source of spherical waves.

Combination microphones incorporate a small number of individual sensors in a compact unit of which the outputs are combined to produce enhanced directional sensitivity (e.g. hyper-cardioid) [2,3,4]. They are generally poor for source location applications. However, it should be noted that the intensity *null* of a p-p intensity probe gives a very sensitive indication of the position of an isolated source in free field. It is also surprisingly effective in reverberant environments because the intensity of a 'diffuse' reverberant field is, on average, zero.

End-fire microphones, also known as 'tube', 'line' and 'wave-interference' microphones', comprise one or more tubes leading to a pressure microphone. The single tube system consists of a tube with a microphone located at one end. Sound is conducted to the microphone *via* a series of individual holes arranged along the length of the tube, or a continuous slit along a generator of the tube. The acoustic resistance of the holes/slit is varied along length to produce less variation of directivity with frequency. The multi-tube ('rifle' or 'shot gun' mic.) consists of a bundle of parallel tubes of different length, all leading into a common microphone chamber. As frequency increases, the main directivity lobe becomes narrower and the number of side lobes increases. The frequency response of conventional systems varies rapidly and irregularly with small changes of angle of incidence. This severely compromises its use for quantitative work. However, a fairly recently developed form [24] has a considerably more regular frequency response.

The practical advantages of end-fire microphones are as follows: they are cheaper and simpler than combination mics; they are quite useful for survey work around outdoor installations; they exhibit good rejection of sound from rear; they may be used effectively in arrays (see 'SYNTACAN' in the following section). The disadvantages are as follows: they require a complicated calibration procedure; they are too long and cumbersome for effective performance below 1000 Hz; they are very sensitive to wind disturbance, although windscreens can help considerably; they are not well suited to quantitative evaluation of source strength except under free field, far field conditions.

Acoustic mirrors convert omnidirectional pressure microphones into focused sensors. Parabolic mirrors are appropriate for far field work; elliptical mirrors are used for near field focusing. Since the aperture must be greater than a wavelength to be effective they are generally heavy and cumbersome. The gain is proportional to 1/f. They must be constructed very precisely. The placing of an elliptical mirror is extremely critical because it has little depth of focus. Mirrors produce multiple reflections if in the proximity of large radiating surfaces.

3.2.4 Microphone Arrays

An array comprises of number of microphones arranged in a precise geometric pattern. The outputs are processed so as to extract wavefront information. Very complex and sophisticated antennae and associated signal processing procedures have been developed for underwater use.

The conventional beam-former (additive antenna) consists of an array of nominally identical microphones arranged at uniform intervals along a straight line. The output consists of the summed microphone signals. It is assumed that the source propagates a coherent wavefront to the sensors and that the wavefront shape known as a function of source position. The upper frequency limit to avoid directional ambiguity is given by $f = c/d$, where c is the speed sound and d is the microphone separation. The half-angle ϑ_C of the main directivity lobe is given $\sin \vartheta_C = \lambda/L$, where L is the array length. The spatial resolution at a given frequency can only be increased by increasing the size of the aperture (i.e. increasing L). Increasing the number of microphones within a given antenna length suppresses the side lobes and increases the aliasing frequency.

Additive antennae can be steered by electronically delaying the signal from each microphone by a time proportional to its distance from the centre Pressure microphones may be replaced by particle velocity sensors (mic. pairs) to provide a multiplicative $\cos \vartheta$ factor on directivity or directional combination mics to reduce sensitivity to sound arriving from the rear.

A conventional beamformer may be focused for use in the geometric near field of a large source by applying suitable time delays and sensitivity weighting to each microphone signal [15, 20]. However, spatial discrimination (resolution) not usually sufficient except at rather high frequencies. Also, the selectivity is vulnerable to strong sources local to outer microphones. They are not useful in reflective environments for obvious reasons.

A development of the conventional beamformer, the High Resolution Synthetic Acoustic Antenna (SYNTACAN), has been commercially available for some time [16, 17]. Its original purpose was to provide a means of locating individual uncorrelated sound sources at long distances e.g. noise of industrial plant affecting local community at distances of between hundreds of metres and 1 km. More recently it has been used to locate sources on wind turbines. The sensors are end-fire microphones and the total number of sensors is minimised by arrangement in a sparse array. An array length of 70 m produces angular resolution of a few degrees down to 100 Hz. The results are impressive, as illustrated by Figure 2. The system is complex and expensive, and time consuming to install. However, the expense and effort should be compared with the costs to the client (and, maybe, the consultant) of an incorrect source diagnosis.

3.2.5 Sound Intensity Measurement

In the technique of Reverse Vector Projection, the intensity vector distribution is determined in the geometric near field of a source system and the vectors are reverse projected towards the source surface. This procedure is fairly reliable in cases where one source region is dominant, or where or two *uncorrelated* sources are well separated in space, because the net intensity at any point is the vector sum of the intensities generated by the individual sources. However, it does not effectively separate two closely spaced, uncorrelated sources of similar strength unless the intensity measurements are made rather close to the source surface, where the presence of the hydrodynamic near field may confuse the issue.

Intensity distributions in reflective enclosures can be very complex because of interference effects and acoustic modal resonances, and are, in general, not capable of interpretation in terms of source location [6]. The combination of reverberant behaviour and pure tone, or narrow band, excitation renders the technique impotent. This is a major problem in vehicle compartments below about 300 Hz because of lack of sound absorption. At higher frequencies, the problem is less severe, and intensity measurement can detect air leaks through door seals, pedal holes, etc. - but so can the human ear, especially if cupped by the hands! [Warning! It can be very misleading to project intensity components *normal* to a sound intensity measurement surface back to a parallel or conformal source surface. The normal component may even be the smallest of the three orthogonal components.]

The other principal method of applying sound intensity measurement to source investigation consists in measuring sound power radiated by selected segments of the surface of a source system. It is more a quantification technique than a location technique. It is generally successful for efficient broadband radiators (e.g. I.C. engines) radiating into free field. It is unreliable for inefficient radiators, especially thin vibrating panels like windows because the local power does not necessarily propagate into the far field: it may well return to the source structure in a neighbouring region.

Both techniques can be applied in a time-windowed (or 'gated') mode which is linked to the operating cycle of a source system. This can be most helpful in separating individual sources which are not separable on the evidence of long-time averages. Transient sound intensity analysis is also available for application to sources which operate over brief periods of time [6]. However, it requires the recording of many events to be effective

In cases of sources operating on the boundaries of enclosures, there is good reason not to rely upon normal intensity scans over the boundaries to indicate troublesome sources. The target variable for noise control is sound pressure, not sound intensity, and there is no simple relation between boundary intensity and sound pressure at an interior point: the pressure is determined by interference between sound waves arriving from all sources and their reflected fields. It is not unknown for the suppression of an apparently strong boundary source to *raise* the sound pressure level at a critical receiver point.

Near field intensity scans may be used to estimate the strengths of (assumed) uncorrelated monopole source distributions which represent vibrating surfaces, and which are subsequently used, in combination with reciprocally measured transfer functions, to estimate individual source contributions to sound pressure at some point(s) of concern (e.g. operator's head position, pass-by monitor) [30, 31]. Some inverse source identification techniques have used intensity as the field variable.

4. SOURCE IDENTIFICATION TECHNIQUES

4.1 System Identification via Coherence Analysis

The objective of coherence analysis as applied to acoustic/vibrational systems is to attribute linear system output quantities (sound) to input quantities (vibration, fluid motion, combustion, etc) on the basis of some minimum model of a system [7-14]. Thus the located 'sources' are bound to specific outputs at specific target points of interest, rather than simply being generators of noise in some extensive region of concern. This subject is too specialised to be covered in any detail herein. However, some general remarks are in order.

A system is instrumented with a set of transducers which are considered, on the basis of a physical understanding of the mechanics of the dynamic behaviour, to provide 'indicator' signals which are representative of the actions of the principal independent sources and, it is hoped, rather insensitive to the actions of all the other sources. Since wave-bearing systems distribute the effects of local disturbances throughout the whole system, it is clearly impossible to avoid 'contamination' of any one indicator signal by many others. They are thus more- or-less correlated. Attribution of a sound signal to any one source via an indicator signal is consequently problematic. Coherence techniques aim to resolve this difficulty, but a principle weakness is that the theory is based upon an assumption that signals are white and Gaussian, an idealised model which real acoustic and vibration signals rarely approach.

Ordinary Coherence analysis is based upon the assumption of a known set of uncorrelated inputs. The Ordinary Coherence Function between each input and each output indicates the relative contribution of that input to the output. For a single input-output system it provides an indication of the possible presence of noise and/or non-linear behaviour.

The Multiple Coherence Function provides a check that all significant inputs have accounted for and as an indicator of the presence of noise in the signals and/or non-linear behaviour of the system.

In vibroacoustic systems, structural vibration at any point is usually caused by a number of independent mechanical/fluid dynamic source mechanisms (e.g. wind turbulence/tyre vibration/engine vibration in a car). Signals from accelerometers on a vibrating structure, which may be considered to represent the source of radiated sound, will be partially correlated. The Partial Coherence Function is used to transform a number of partially correlated inputs into a set of equivalent uncorrelated inputs. One 'input' signal selected, and the components of the other input spectra which are coherent with it are removed from their corresponding autospectra. The process is then applied successively to all input signals to produce conditioned spectra. The technique is effective in practice only if some rank ordering of the inter-source signal 'contamination' is employed on the basis of *a priori* knowledge of the system. It produces ordinary coherence functions between conditioned inputs and outputs. It is often difficult to understand the resulting global correlation pattern in physical terms and it is impossible to determine the absolute contribution of each physical source. One can only attribute conditioned outputs to conditioned inputs. The PCF also suffers from numerical problems in cases where selected inputs are coherent.

Principal Component Analysis reduces a set of measured inputs to a set of incoherent 'principal' inputs by weighting the actual inputs in such a way as to minimise the weighted signal correlations. It is based upon Singular Value Decomposition of a set of cross-spectra of inputs and outputs. The corresponding 'source' distributions are called 'virtual sources': it is usually difficult to relate virtual source distributions to the physical source mechanisms. Ordinary coherence functions can be formed between virtual inputs and outputs, and the multiple coherence function indicates the presence of omitted inputs, noise and non-linearity.

System identification analysis based upon coherence and principal component analyses requires the measurement of a number of input indicator signals which is at least as large as the number of independent sources, and which are effective indicators of the actions of these sources. This is generally more practicable where structure-borne sound is transmitted from the primary excitation region to the receiving space via a small number of discrete (individual) connections (e.g. suspension units of a vehicle; engine mounts).

The results of these forms of analysis are very sensitive to the choice of inputs: inappropriate selection will lead to misleading/erroneous conclusions. Coherence techniques have not proven to be particularly useful in identifying sources of airborne sound radiation at the solid-air interface, particularly because structure-borne and airborne waves are readily transmitted between sites of indicator signal transducers, and individual resonant structural modes are essentially spatially coherent over the whole modal system, however excited.

4.2 Inverse Techniques

Inverse Techniques fall into two broad categories. In the Assumed Source Model, measured sound field data are interpreted in terms of an *assumed source model* on the basis of a set of discrete theoretical or empirical transfer relationships between source elements and field quantities (usually by matrix inversion). In the Field Projection Technique, measured field data are mapped back to the assumed source region by means of field transformations which represent solutions to the acoustic wave equation. This category includes Nearfield Acoustical Holography (NAH)

4.2.1 Assumed Source Model

The principle of the technique is as follows:

(i) A model of the source distribution is assumed (e.g. type, number (*N*), location) of which the individual 'strengths' are unknowns to be sought.

(ii) Analytical expressions, numerical models (BEM/FEM), or experimental transfer functions, which relate the acoustic field generated to the assumed sources are obtained: this is the transfer model. Experimental transfer functions are usually determined reciprocally [34].

(iii) Sound field data (e.g. pressure auto- and cross spectra) are measured at an array of points which *must equal, or usually exceed* , *N* in number.

(iv) The transfer matrix is inverted to yield the source strengths. This process is illustrated by Figure 3 for single frequency sources. Ref. [26] provides an example of the technique. This procedure is usually carried out in the frequency domain, but it can also be carried out in the time domain, in which case impulse response functions replace transfer functions [33].

Practical problems include the following: measured data are subject to noise contamination which can introduce large errors into the deduced source strengths; the transfer matrix may be ill-conditioned, in particular because of the presence of very small magnitudes at certain locations and frequencies. Conditioning is also dependent upon the location of selected observation points. Therefore it is necessary to measure sound field data at more than N points. This generates an overdetermined problem, which reduces the influence of uncorrelated noise and provides a consistency check. The problem is solved by minimising the difference between the measured and predicted field quantities. Various mathematical procedures are applied to improve the conditioning of the matrix. These are beyond the scope of this paper, but are discussed in Ref. [33]. The success of this technique is critically dependent upon an appropriate choice of source model for which general selection criteria are not well established.

4.2.2 Field Projection Techniques

The most widely used field projection technique is Nearfield Acoustical Holography (NAH) [25, 27, 28]. Field projection is based upon the principle that the whole of a sound field is completely determined by the field conditions on its bounding surfaces (i.e. 3-D output from 2-D input) [18, 19]. For bounding surfaces which take simple geometric forms (e.g. planar, cylindrical, spherical) the relation between boundary conditions and field quantities take analytically explicit forms which satisfy the wave equation with periodic sources. Measurements are made of the spatial distribution of amplitude and phase of a field quantity (normally sound pressure) on a hologram surface which conforms with the source surface, and the whole sound field, including that in the proximity of the sources generating the field, is reconstructed, with the aim of identifying the principal sources of noise (Figure 4). The spatial distributions of all sound field quantities, such as pressure, particle velocity and intensity, over whole of the radiated field can be derived.

If the measurements are not made sufficiently close to the source surface, they do not receive contributions from nearfield components of the sound field. In this case, the process of reconstruction of the source does not capture features having a spatial scale less than half a wavelength. NAH overcomes this limitation. However, because near field components decay exponentially with distance from the source surface, the inverse process of reconstruction involves exponential amplification of those components. Consequently, the effect of any errors in the sampling of the sound field caused by inadequate dynamic range of the measuring system, or extraneous noise, can produce large reconstruction errors in the small scale structure of the source and its near field. The 'damage' can be limited by suppressing the troublesome high wavenumber components. Another practical problem is that the measurement array is limited in extent which requires the application processing procedures to eliminate false images of the source. Both filtering processes inevitably 'damage' the image and must be applied with great care.

With aperiodic time-stationary sources of sound, the relative amplitudes and phases of the sound field as sensed by the measurement array microphones are determined from cross-spectra. Many noise sources are broadband in frequency and also comprise a number of independent (uncorrelated) source components. In order to increase the efficiency of the procedure the cross-spectra must be conditioned by reference to a set of indicator signals measured close to the source which are considered to be representative of the actions of the independent sources. Only as many 'sources' as indicator signals are identified. In order to implement this procedure in a logistically and computationally practicable way, Brüel & Kjær have developed a special form of holographic measurement system termed Spatial Transformation of Sound Fields (STSF) [21-23]. This has been developed to the point where it is now possible to visualise the evolution in time of non-stationary sources of noise such as vehicles during pass-by tests.

Field projection techniques using mono-layer measurement arrays suffer from the problem that sound arriving at the measurement surface from the opposite side to the source being imaged cannot be discriminated from sound radiated directly by the source. Consequently, it cannot be applied in reflective environments or where other coherent sources are operating.

4.3 Source Characterisation

All the techniques described above relate to the 'outer skin' of a source system. They do not reveal the mechanism which drive this 'skin'. Therefore they should be used in conjunction with parametric investigations of the character of the driving system. These are based upon studies of the following features of a source: variation of sound power with speed and load (where possible); spectral character of the noise and its dependence upon operating speed and load (where possible); (ii) temporal characteristics of the noise in relation to the dynamic and kinematic features of the operating cycle [35]. Listening to slowed down recordings of source noise can be most helpful in relating acoustic events to mechanical events.

A Review of Source Identification - F.J. Fahy, University of Southampton

REFERENCES

1. Blauert, J. *Spatial hearing: the psychophysics of human sound location.* MIT press, Cambridge, MA, USA (1983).
2. Olson, Harry, F. *Acoustical Engineering.* Professional Audio Journals, Inc., Philadelphia, USA (1991) (Previously published by Van Nostrand Co. Inc., New York, 1957).
3. Beranek, L.L. *Acoustical Measurements* (second ed.). American Institute of Physics, New York (1988).
4. *Microphones: an anthology.* Audio Engineering Society, Inc. New York (1979).
5. Lesueur, C. (ed). *Rayonnement acoustique des structures.* Editions Eyrolles, Paris (1988).
6. Fahy, F.J. *Sound Intensity* (second ed.). E& FN Spon, London (1995)
7. Jenkins, G.M. and Watts, D.G. *Spectral Analysis and its Applications.* Holden-Day, San Francisco (1968).
8. Otte, D. Doctoral Thesis, Katholieke Universiteit Leuven, Leuven, Belgium.
9. Sas, P. Introduction and overview on noise source location techniques. In '*Proceedings of the Second International Seminar on Noise Source Identification and Numerical Methods in Acoustics*, eds. P. Sas and K. Fyfe, Katholieke Universiteit Leuven, Leuven, Belgium (1987), 1-17.
10. Bendat, J. and Piersol, A.G. *Engineering Applications of Correlation and Spectral Analysis.* Wiley-Interscience, John Wiley & Sons, USA (1980).
11. Desanghere, G. and Snoeys, R. Noise source identification using frequency response functions. (See Ref. 9 for source), 12-14.
12. Otte, D., Sas. P., Snoeys, R., Van de Ponseele, P., and Leuridan, J. Use of principal components analysis and virtual coherences for dominant noise source identification. (See Ref. 9 for source), G1-5.
13. Wagstaff, P., Bouizem, B., Bohineust, X., and Henrio, J. C. Source identification using partial coherence, virtual coherence and principal inputs. In *Proceedings of Inter-Noise 88*, Avignon, France, 183-186.
14. Filippi, P.J.T., Habault, D., and Piraux, J. Noise source modelling and identification procedures. *J. Sound. Vib.*, 124(2), (1988) 285-296.
15. Elias, G. and Malarmey, C. Utilisation d'antennes focalisées pour la localistions des sources acoustiques. In *Proceedings of the 11th International Congress on Acoustics*, Paris (1983), 163-166.
16. Boone, M.M. and Berkhout, A.J. Theory and applications of a high resolution synthetic acoustic antenna for industrial noise measurements. *Noise Control Engineering Journal* 23 (1984) 60-68.
17. Boone, M.M. Design and development of an acoustic antenna system for industrial noise source identification. In '*Underwater Acoustic Data Processing*' ed. Y.T. Chan; NATO ASI Series E: Applied Sciences, Vol. 161, Kluwer Academic Publishers, Netherlands (1989).
18. Weinreich, G. and Arnold, E.B. Acoustic holography in spherical co-ordinates. In *Proceedings of Inter-Noise 86*, Cambridge, MA, 1223-1228.
19. von Hoffmann, B. Personal computer implementation of nearfield acoustic holography. Report E 89-05, Department of Applied Acoustics, Chalmers University of Technology, Gøteborg, Sweden (1989).
20. Guilhot, J.P., Bescombes, M., and Legros, C. Localization and identification of noise sources. In *Proceedings of Inter-Noise 88,* Avignon, France , 171-174.
21. Hald, J. STSF- a unique technique for scan-based Nearfield Acoustic Holography without restrictions on coherence. In Brüel & Kjær Technical Review, No. 1 (1989).
22. Ginn, K.B. and Hald, J. STSF - Practical instrumentation and applications. In Brüel & Kjær Technical Review, No. 2 (1989).
23 Hald, J. Spatial Transformation of Sound Fields (STSF) Techniques in the Automotive Industry. In Brüel & Kjaer Technical Review, No. 1 (1995).
24. Shulman, Y. Reducing off-axis comb-filter effects in highly directional microphones. *J. Audio. Eng. Soc.* **35**(6), (1987) 463-469.
25. Williams, E. G. Introduction and survey of recent developments in nearfield acoustical holography. In *Proceedings of Inter-Noise 88*, Avignon, France (1988) 203-208.

26. Kim, G.-T. and Lee, B.-H. 3-D sound source reconstruction and field representation using the Helmholtz integral equation. *J. Sound. Vib.* **136**(2) (1990) 245-261.
27. Wetta, P., Beguet, B., Parent de Curzon, E. Experimental Analysis of wheel/rail interaction by nearfield acoustical imaging. *J. Sound. Vib.* **120**(2) (1988) 255-265.
28. Maynard, J.D., Williams, E.G. and Lee. Y. Nearfield acoustic holography: I Theory of generalized holography and the development of NAH. *J. Acoust. Soc.* Amer. **78**(4) (1985) 1395-1413.
29. Fisher, M.J. Engine noise source breakdown: theory, simulation and results. In *Proceedings of the AIAA 7th Aeroacoustics Conference*, Paper AIAA-81-2040, Palo Alto, CA (1981).
30. Verheij, J. Reciprocity method for the quantification of airborne sound transfer from machinery. In ed. M.J. Crocker, Auburn University, Alabama, USA (1992) 591-598.
31. Zheng, J., Fahy, F.J. Anderton, D. Application of a vibro-acoustic reciprocity technique to the prediction of sound radiated by a motored I.C. engine. *s* **42**(1994) 333 -346.
32. Adin Mann III, and Pascal, J. Locating noise sources on an industrial air compressor using broadband acoustical holography from intensity measurements (BAHIM). *Noise Control Eng. Jour.* **39**(1992) 3-12.
33. Nelson, P.A. Some inverse problems in acoustics. *Proceedings of 6th International Congress on Sound and Vibration, Copenhagen* (1999).
34. Fahy, F.J. The vibroacoustic principle and its applications. *Acustica* **81**, (1995) 544-558 .
35. Fahy, F.J. and Walker, J.G.(eds)*n*. E&FN Spon, London (1998).

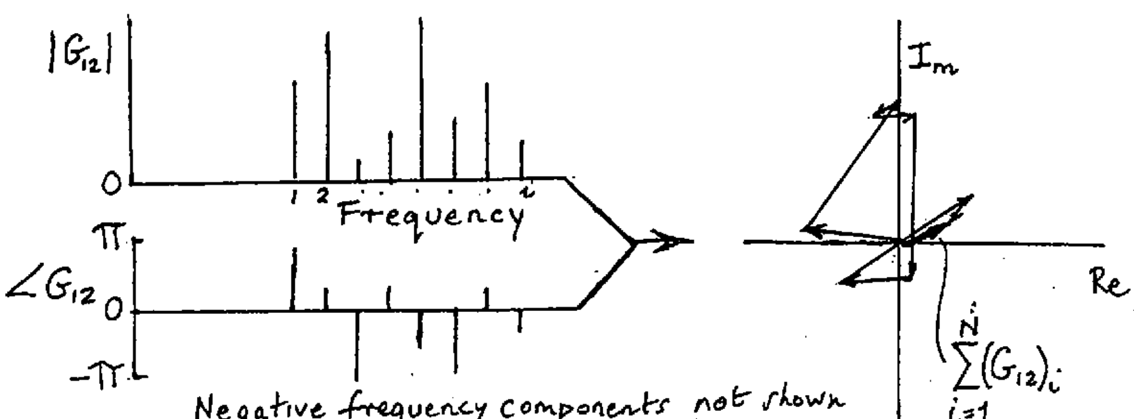

Figure 1 Zero-time-delay cross correlation function is the frequency integral of the cross-spectral density.

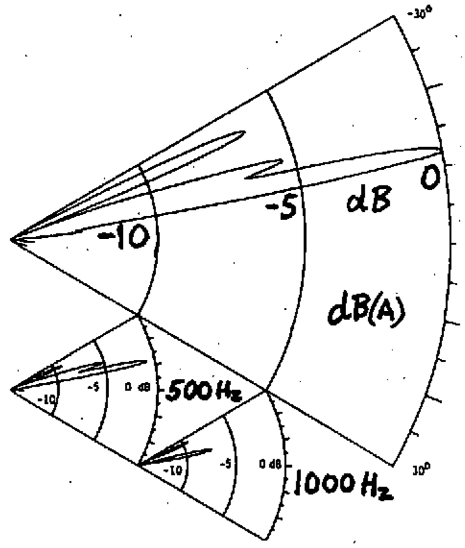

Figure 2 Polar diagram of the noise sources of the wind turbine, showing three distinct noise sources: the nacelle (–12°) the left rotor root (–16°) and the left rotor tip (–24.5°).

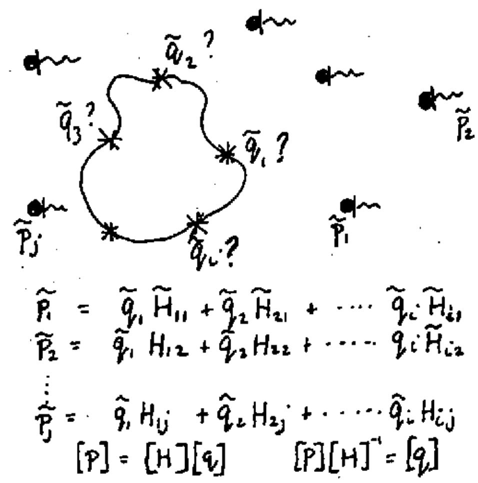

Figure 3 Principle of inverse method of source identification

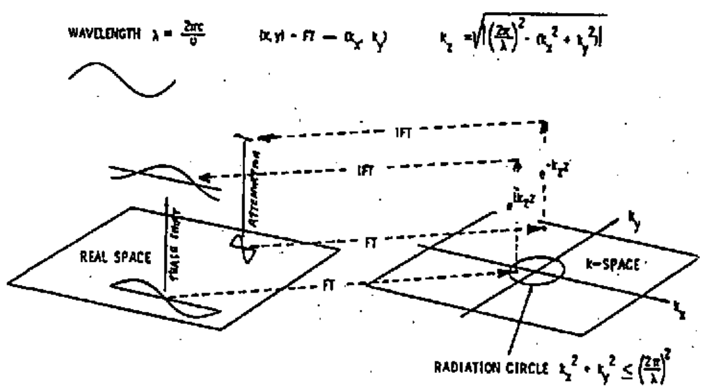

Figure 4 Transformations employed in NAH (after Ref. 28).