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PERFORMANCE ENHANCEMENT OF STRUCTURAL/ACOUSTIC ACTIVE CONTROL SYSTEMS VIA ACOUSTIC ERROR SIGNAL DECOMPOSITION

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1. INTRODUCTION

The active control of free field acoustic radiation is a topic which has received a great deal of academic attention, but (very) few commercial implementations by practitioners. While there are a number of reasons for this, two prominent ones are a lack of suitable electronic control systems, and "physical" acoustics problems which limit the performance of free field active noise control systems. For example, even the earliest studies on the active control of transformer noise pointed to the need for adaptive control systems, and to the (physical acoustics) need for multiple (perhaps dozens of) sound sources and sensors if global sound attenuation is desired. However, large numbers of sources and sensors tend to complicate the design of, and add a great deal of "inertia" to, the adaptive control system. Finding a suitable compromise is a difficult task.

In some instances, where the offending noise source is a vibrating structure, it is possible to attenuate the acoustic field by inputting a controlling vibrational disturbance. This technique has been of some interest to the structural acoustics community, particularly when applied to lightweight structures (such as airplanes and automobiles). Recent work has been directed at optimising the design of embedded transducers, which offer the potential to both improve system performance and reduce the number of sensors required to achieve global sound attenuation. However, there are a number of structures which are simply too bulky for this approach, structures such as large electrical transformers. For these, what is needed is some means of optimising the traditional transducer combination of speakers and microphones so as to achieve global sound attenuation with the minimum number of (adaptive) controller error signals.

This paper considers the problem of optimising the error sensing component of free field active noise control systems, where the sensors are microphones. A methodology will be developed, and the resulting system experimentally trialled on a simple laboratory arrangement.

2. OBJECTIVES

In order to develop an optimal microphone sensing system, it is first necessary to set out two important objectives:

Objective One: Minimise Radiated Acoustic Power

The most common target for an active control system is the minimisation of some global disturbance measure. Included in this category are the measures of structure kinetic energy for vibration control problems, and acoustic potential energy for noise control in an enclosed space. For free field acoustic radiation, the global measure of interest is radiated acoustic power. Obtaining a measure of this quantity for minimisation by the active control system is the first objective of the sensor design.

Objective Two: Use As Few Error Sensor Inputs As Possible

The number of "error sensor" inputs provided to an adaptive control system as feedback on the state of the residual acoustic field has a significant impact upon controller performance (the error sensors inputs are to be minimised by the adaptive control system). If there are a large number of error sensor inputs, the adaptive control system will be (perhaps very) slow to converge to an optimal solution. The hardware requirements, measured in terms of cost and complexity, can also increase markedly. Therefore, a second objective of the sensor system design for practical implementations must be to minimise the number of error sensors inputs required to obtain satisfactory levels of global disturbance attenuation.

3. METHODOLOGICAL BASIS - MODAL FILTERING

While the aim of many active control systems is to provide global disturbance attenuation, it is very difficult to measure global quantities (for minimisation by the control system) in an efficient manner. Traditionally, a large number of point sensors, such as microphones or accelerometers, have been employed in an attempt to infer global trends through bulk data. In active control, this approach is unacceptable, as the use of a large number of sensor signals decreases the performance of the adaptive controller.

An alternative to the above process is to decompose the large number of point sensor signals into a smaller number of higher quality error signals. These "higher quality signals" are orthogonal measures of the global error criterion of interest. For example, if vibration attenuation is the target of the exercise, and so structural kinetic energy is the global error criterion of interest, it can be shown that modal velocities are orthogonal contributors to global error criterion. If, for example, only a few structural modes have resonance frequencies in or near the control system bandwidth, then a large number of point sensor signals can be decomposed into a handful of modal velocity measurements, which in turn can be used as controller error signals. This approach to control system implementation, where a reduction in the amplitude of system modes is explicitly targeted, is referred to as modal control; the decomposition is referred to as modal filtering. (Note that what is meant by decomposition is the weighted addition or subtraction of all sensor signals to produce a single quantity, or single "decomposed" signal.)

While the concepts of modal control and modal filtering are easily conceptualised when applied to structures, they are more difficult to apply to acoustic or structural / acoustic problems. The important point to note is that the error signals which are derived from decomposition of the point sensor measurements must be orthogonal measures of the global error criterion of interest. The ensuing problem is then how to derive these orthogonal quantities, and how to determine the modal filter coefficients which are used in the signal decomposition process, when the sensors are microphones.

4. MODAL FILTERS FOR ACOUSTIC RADIATION

To derive a modal filtering-like decomposition technique for acoustic radiation problems, consider a specific example: control of acoustic radiation from a vibrating structure. Extensions to the methodology for other acoustic radiation problems are possible. Limiting consideration to a finite number n structural modes, the acoustic power output W of a vibrating structure can be written in matrix form as

$$W = \mathbf{v}^{H}\mathbf{A}(\omega)\mathbf{v}, \tag{1}$$

where $\mathbf{v}$ is a (n x 1) vector of modal velocities, and $\mathbf{A}$ is an (n x n) frequency dependent weighting matrix, the terms of which can be derived by either near field or far field integration [1]. The weighting matrix $\mathbf{A}$ is real and symmetric, and can diagonalised via an orthonormal transformation:

$$W = \mathbf{v}^{H}\mathbf{A}\mathbf{v} = \mathbf{v}^{H}\mathbf{Q}\Lambda(\omega)\mathbf{Q}^{-1}\mathbf{v} = \mathbf{u}^{H}\Lambda(\omega)\mathbf{u}, \tag{2}$$

where $\mathbf{Q}$ is the orthonormal transformation matrix, the columns of which are the eigenvectors of the weighting matrix $\mathbf{A}$, Λ is a diagonal matrix of eigenvalues of $\mathbf{A}$, and $\mathbf{u} = \mathbf{Q}^{-1}\mathbf{v}$.

There are two important points to note concerning the above result. The first is that as the matrix Λ is diagonal, the expression for radiated power in equation (2) is decoupled. This means that the quantities in the transformed vector $\boldsymbol{u}$ are independent contributors to the global error criterion. These independent quantities are made up of combinations of structural modes, as defined by the eigenvectors of the weighting matrix $\boldsymbol{A}$ (for a specific example, see [2]). The second important point to note is that the frequency dependence of the power equation (1) rests entirely with the eigenvalues (this is an approximation [2]). This means that if the structural modal amplitudes are measured and combined in a static (non-frequency dependent) prescribed way (prescribed by the "modal filter" coefficients, which are contained in the eigenvectors of $\boldsymbol{A}$), and the resulting signals weighted (by frequency dependent eigenvalues) and minimised, then the global error criterion will be explicitly minimised [1,2].

This idea is acceptable when the structural modal amplitudes can be minimised, and when the control source is a vibration transducer (which will alter the structural modal amplitudes). However, the above approach is not readily applicable when the sensors are microphones and the sources are speakers.

In order to implement the above strategy when acoustic sources and sensors are being used, note that the acoustic pressure at a number of points m in space due to acoustic radiation from a number n of structural modes can be expressed in matrix form as

$$\boldsymbol{p} = \boldsymbol{Z}(\omega)\boldsymbol{v}, \tag{3}$$

where $\boldsymbol{p}$ is the (m x 1) vector of complex pressures at the points of interest, and $\boldsymbol{Z}$ is the (m x n) matrix of (frequency dependent) radiation transfer functions from the modes to the points. If this relationship is substituted into equation (2), the modal filtering exercise described therein then becomes a function of pressures measured at points in space, rather than structural modes. As the problem is now formulated in the acoustic domain, it is possible to use microphones to measure the signals. It is also possible to implement the control system using speakers.

There is one remaining theoretical problem to overcome with the above approach. That is, if the relationship in equation (3) is simply substituted into equation (2) to derive modal filter coefficients based upon pressure measurements, the resulting coefficients will be frequency dependent (as the relationship between the structural mode amplitudes, upon which the methodology is based, and the acoustic pressure at the measurement points will be frequency dependent). This is an unwanted complication, as the modal filtering-type procedure which is desired is based upon the idea that a set of point measurements can be weighted and added or subtracted to yield error signals which are orthogonal measures of the global error criterion; frequency weighting implies that some form of pre-filtering is required. In order to overcome this problem, it is possible to break the radiation transfer matrix $\boldsymbol{Z}$ into a "base" frequency independent part, and a frequency dependent "correction factor". The former is combined with the eigenvectors to derive the frequency independent modal filter coefficients, which the latter is lumped with the frequency dependent eigenvalues.

To summarise, the end result of this process is the derivation of a set of "modal filter coefficients". These coefficients are used to convert a set of point acoustic pressure measurements into orthogonal measures of acoustic power. This orthogonal measures are actually the radiation patterns of orthogonal groupings of structural modes referred to as "radiation modes" or "transformed modes" [1]. Using this process, a large number of point measurements can be decomposed into a small number of high quality measurements, to be minimised by an active control system.

5. IMPLEMENTATION

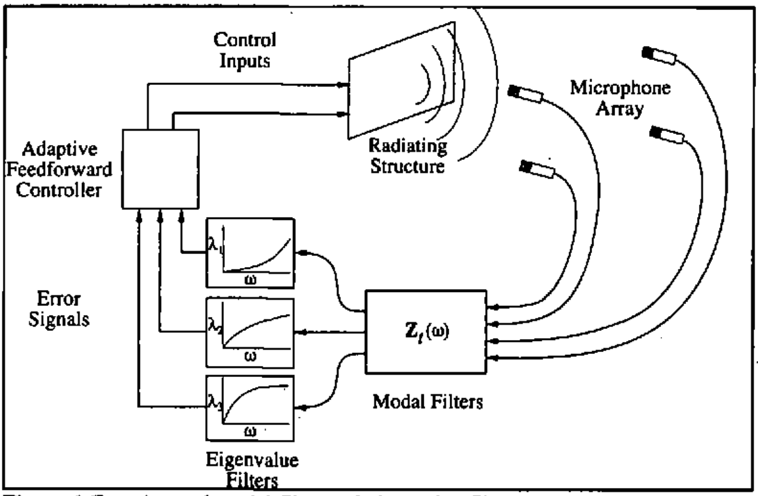

Figure 1 Experimental modal filter and eigenvalue filter arrangement.

The above procedure was experimentally implemented for the problem of controlling acoustic radiation from a small vibrating panel (the case considered in reference [2] for vibration error measurements). A sketch of the experimental arrangement is shown in figure 1. To implement the previously derived error decomposition techniques in a practical system, it was necessary to find a way to overcome the problem of using complex numbers in the analysis, and real numbers in the practical system. To overcome this problem, the phase difference between sensors was "normalised" by inserting sensor dependent time delays into the system.

Illustrated in figure 2 is a comparison between minimising the

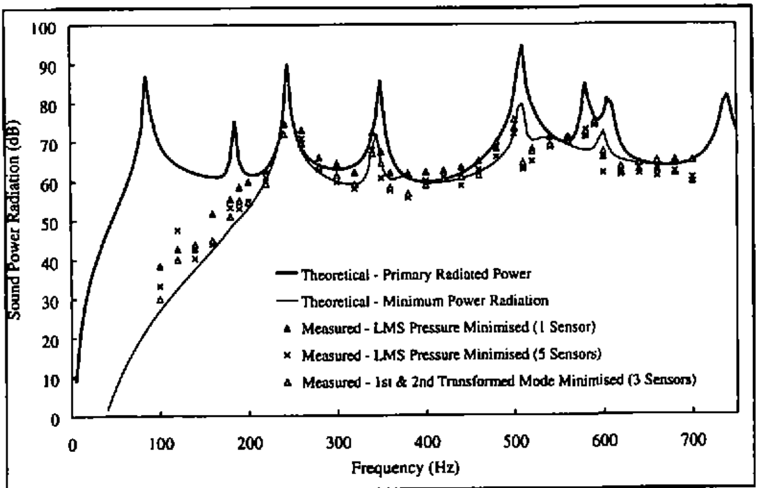

Figure 2 Experimental controlled radiated acoustic power, minimising acoustic pressure signals and decomposed signals.

acoustic pressure at 5 locations in space, and minimising the two orthogonal measures (referred to as "transformed modes" in the figure) of acoustic power using three sensors. The results are very similar. However, the pressure experiments require 5 error signal inputs to the control system, while the modal filtering approach requires only two.

6. CONCLUSIONS

It is possible to decompose a large number of point acoustic pressure measurements into a small number of "high quality" active control system error signals and obtain similar or improved results. The decomposed signals should be orthogonal measures of the global error criterion, which in the case considered here is acoustic power. These measures can be the radiation patterns of orthogonal combinations of structural modes, or could perhaps be simple monopole / dipole / quadrapole patterns.

References

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