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THE SYNTHESIS OF A DIFFUSE SOUND FIELD WITH A NEAR-FIELD ARRAY OF LOUDSPEAKERS

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Abstract

This paper shows practical results on the experimental synthesis of an acoustic diffuse field over a test partition separating a source room from a receiving room. The objective is to validate a novel approach in order to reduce the variability of sound transmission measurements in the low-frequency domain. The approach is based on the generation of an acoustic diffuse field with a near-field array of sixteen loudspeakers suitably driven and located in the source room side of a transmission suite. The sources drive signals are optimised to reconstruct the same spatial correlation structures than an ideal diffuse field over a sufficiently dense grid of microphones distributed a short distance away from the test panel surface. The technique enables to compensate for the modal behaviour of the source room which is a major limitation when sound transmission measurements have to be performed using a small sized reverberation chamber. Experimental results show the efficiency of the approach to provide a measure of the sound reduction index that only depends on the properties of the panel itself over a broad frequency range, especially below the source room Schroeder frequency. Empirical criteria are proposed on the number of partially correlated sources required to achieve an accurate reconstruction of the assumed diffuse field statistics over the test panel. This is analysed in relation with the influence of the degree of diffusivity upon the sound reduction index.

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INTRODUCTION

Several experimental studies in room acoustics have shown evidence of the variability of low frequency sound transmission measurements [1, 2]. The insulating properties of a test partition are measured in a sound transmission suite made of two reverberation rooms coupled *via* the partition. The variability of sound insulation measurements has been observed at low frequency through inter-laboratory comparisons. The lack of reproducibility of sound insulation measurements is mostly due to the non-diffuseness of the sound fields which are dominated by a few normal modes in this frequency range. Improvements have been proposed to the existing standard ISO 140-3 such as increasing the number of loudspeakers positions, positioning microphones close to the source room side of the test panel or using an absorbing back-wall in the receiving room [3, 4]. Although the dispersion associated with the reproducibility was reduced, the problem remains of the lack of diffusivity of the source room sound field typically below the room's Schroeder frequency and close to the test panel.

A new methodology has been recently proposed as a potential solution to make the validity of the results independent of the source room specific parameters. It is based on the laboratory simulation of an acoustic diffuse field with an array of loudspeakers located in the source room close to the test partition and driven by suitable time-domain signals [5, 6]. The theoretical feasibility of generating spatially correlated random pressure fields has also been investigated for the laboratory reproduction over an aircraft panel of wall-pressure fluctuations associated to a turbulent boundary layer [7, 8].

The paper attempts to show the practical feasibility of synthesizing an acoustic diffuse field over the surface of a test panel in a series of loudspeakers array simulation experiments performed either in an anechoic environment or in a reverberant one. We focus on the physical limitations of the synthesis in relation to the number of sources required for an accurate simulation as well as the ability of the controller to equalize for the source room resonances at low frequencies.

THEORETICAL BACKGROUND

The methodology is considered of designing least-squares control filters which would drive an array of acoustic sources in order to generate a random pressure field with specified statistical properties at the outputs of a set of microphones close to the test panel. The statistical properties of the desired pressure field, namely an acoustic diffuse field, are characterised by a cross-spectral density matrix, $\mathbf{S}_{dd}$, between the pressures $\mathbf{d}$ at the microphones,

$$\mathbf{S}_{dd} = \mathrm{E}\left[\mathbf{d}\mathbf{d}^{\mathrm{H}}\right]. \tag{1}$$

The cross-spectral density between the pressures at two microphones located a distance r apart when subject to an acoustic diffuse field is given by

$$S_{dd}(r;\omega) = S_0(\omega)\frac{\sin kr}{kr} \tag{2}$$

in which $S_0(\omega)$ is the point-power spectral density at any single microphone and k is the acoustic wavenumber [9]. With reference to the block diagram shown in Figure 1, the desired microphone signals are assumed to be generated by passing a number of uncorrelated white noise reference signals $\mathbf{x}$ through a filter matrix $\mathbf{D}$ that can be deduced from an eigen-factorization of $\mathbf{S}_{dd}$ [6, 7].

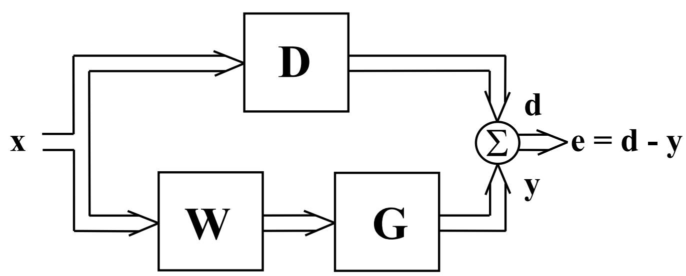


Figure 1 – Block diagram for the calculation of the least-squares matrix of optimal filters.

The reference signals $\mathbf{x}$ are assumed to drive the matrix of control filters $\mathbf{W}$ which generates the input signals to an array of loudspeakers. $\mathbf{G}$ is the matrix of acoustic responses between the near-field loudspeakers and the microphones. The problem is how to best design the matrix of control filters $\mathbf{W}$ such that the microphones output $\mathbf{y}$ is as close as possible to those due to the desired pressure field $\mathbf{d}$. The optimal least-squares matrix of filters is then given by

$$\mathbf{W}_{\mathrm{opt}} = \left[\mathbf{G}^{\mathrm{H}}\mathbf{G}\right]^{-1}\mathbf{G}^{\mathrm{H}}\mathbf{D} = \mathbf{G}^{\dagger}\mathbf{D}, \tag{3}$$

where $\mathbf{G}^{\dagger}$ is the pseudo-inverse of $\mathbf{G}$. $\mathbf{W}_{\mathrm{opt}}$ has been obtained to minimise the sum of the mean square error signals $\mathrm{Tr}\left[\mathbf{e}\mathbf{e}^{\mathrm{H}}\right]$ [7]. When normalised by the sum of the mean square microphone signals due to the desired pressure fields $\mathrm{Tr}\left[\mathbf{d}\mathbf{d}^{\mathrm{H}}\right]$, one obtains the normalised minimum mean square error,

$$J_e = \frac{\mathrm{Tr}\left[\left(\mathbf{I} - \mathbf{G}\mathbf{G}^{\dagger}\right)\mathbf{S}_{dd}\right]}{\mathrm{Tr}\left[\mathbf{S}_{dd}\right]}, \tag{4}$$

which quantifies the degree to which the incident power of the target pressure field has been reproduced by the array of loudspeakers. To fully characterize the accuracy of the simulation, one may introduce a spatial error criterion,

$$\varepsilon_{dy} = \frac{\left\| \widetilde{\mathbf{S}}_{dd} - \widetilde{\mathbf{S}}_{yy} \right\|}{\left\| \widetilde{\mathbf{S}}_{dd} \right\|}, \tag{5}$$

where $\widetilde{\mathbf{S}}_{dd}$ and $\widetilde{\mathbf{S}}_{yy}$ are respectively the spatial cross-correlation functions of the desired and simulated pressure fields with respect to a reference microphone (chosen as the center microphone). This criterion quantifies the deviation of the simulated correlation structures from the one due to an ideal diffuse field as given by Eq. (2).

THE LOUDSPEAKER ARRAY DRIVING EXPERIMENT

In order to assess the practical feasibility of the synthesis technique, an experimental facility has been designed with an array of 4×4 woofer loudspeakers of 210 mm diameter which are driven by a set of optimal signals in order to reconstruct an acoustic diffuse field over a sufficiently dense grid of miniature electret microphones uniformly distributed over the surface of a baffled test panel (see Figure 2).

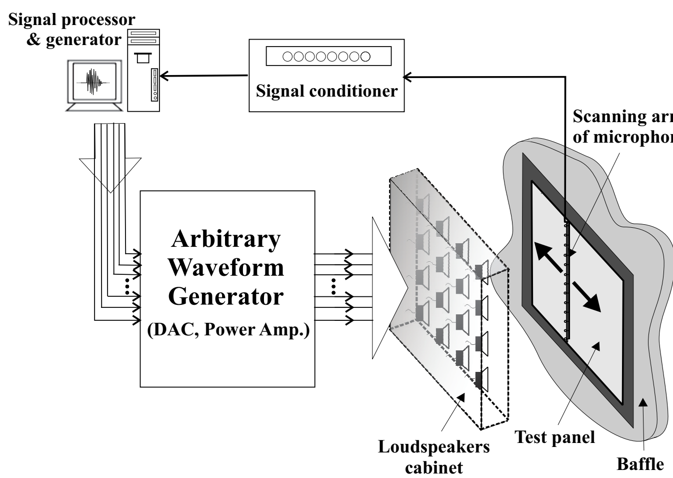


Figure 2 – Experimental set-up for the laboratory synthesis of a diffuse pressure field

The test panel is clamped along its edges and has dimensions 920×740 mm. Calculations performed both in a free-field and in a reverberant environment have shown that a suitable separation distance between the array of acoustic sources and the panel should be about the distance between two adjacent loudspeakers, i.e. about 220 mm, in order to lower the condition number of the plant response matrix $\mathbf{G}$.

During the simulation process, the loudspeakers are driven by a number of partially correlated optimal signals synthesized by a 16-channel arbitrary waveform generator programmed by a PC. The drive signals to the actuators are given by $\mathbf{u}_{\mathrm{opt}} = \mathbf{W}_{\mathrm{opt}}\mathbf{x} = \mathbf{G}^{\dagger}\mathbf{D}\mathbf{x}$. They are generated off-line and then played out during the synthesis process, which does not constrain the control filter to be causal.

The generation of the optimal signals requires the measurements of a number of plant responses between the loudspeakers and the microphones. The design of a cost efficient synthesis experiment would require to keep this number as low as possible. As it will be shown later, choice of the numbers of acoustic sources is directly related to the minimum correlation area of the acoustic diffuse field we aim to reproduce. Figure 3 shows the influence of an increasing density of microphones on the accuracy with which a diffuse pressure field is reproduced. A general trend is that the accuracy degrades above 700 Hz, independently of the number of sensors that has been chosen. As expected, the best possible reduction in the mean-square error (4) is achieved when the number of microphones tends towards the number of loudspeakers, but at the expense of the spatial resolution of the reconstructed correlation structure. Choice has been made of a grid of 7×8 microphones which sets a resolution of 925 mm and an upper limit of 1.8 kHz for the synthesis to be carried out without spatial aliasing.

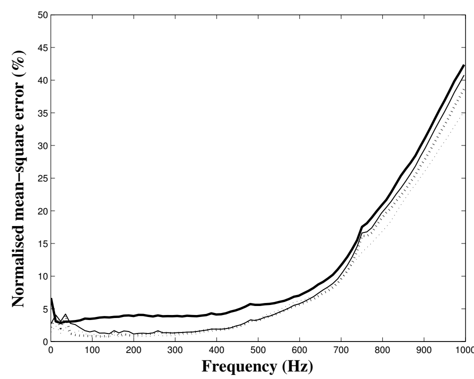
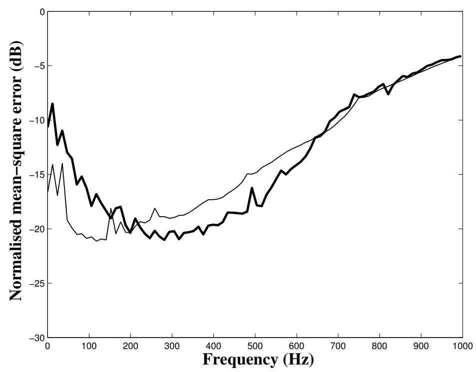


Figure 3 – The normalised mean-square error for simulations in which 4×4 *loudspeakers are used in a semi-anechoic room to reproduce a diffuse pressure field over an increasing number of microphones (*5×6*, thin dotted;* 7×8*, thick dotted;* 13×16*, thin solid and* 15×18*, thick solid).*

RESULTS AND DISCUSSION

Two series of experiments have been performed for the simulation of an incident acoustic diffuse field in various acoustic environments: first, over a baffled clamped panel located in a semi-anechoic chamber of 125 m^3 and second over the source room side of a test panel separating a reverberant room of 43 m^3 from a quiet receiving room of 120 m^3. The performances in either environment are compared in Figures 4, 5 and 6 with respect to the mean-square error, the spatial error and the shapes of the assumed – reconstructed correlation structures.



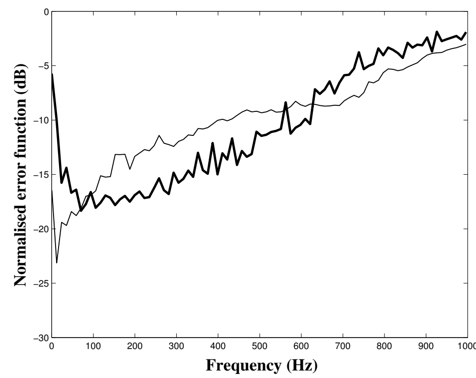


Figure 4

Figure 5

Figure 4 – The normalised mean-square error for simulations in which 4×4 *loudspeakers are used to reproduce a diffuse pressure field over a grid of* 7×8 *microphones in a semi-anechoic room (thin line) and in a reverberant room (thick line).*

Figure 5 – The normalised spatial error for simulations in which 4×4 *loudspeakers are used to reproduce a diffuse pressure field over a grid of* 7×8 *microphones in a semi-anechoic room (thin line) and in a reverberant room (thick line).*

An accurate reproduction of the assumed pressure field can be obtained when the mean-square error reduction and the spatial error fall below 10 dB and 5 dB, respectively. This is achieved up to about 680 Hz in either the semi-anechoic or the reverberant room. Also, we note from Figures 4 and 5 that synthesis of the spatial correlation structure of a diffuse pressure field can be achieved over a broader frequency range in the semi-anechoic room (up to 850 Hz) with respect to the reverberant room (up to 680 Hz). Clearly, it is the number of sources per unit correlation length of the simulated pressure field that mainly determines the limitation performances of the synthesis. In theory, two independent components per unit acoustic wavelength are required in order to accurately reproduce a diffuse field. Assuming 16 sources, this criterion predicts an upper frequency limit of 825 Hz above which the simulation is not accurate. This is confirmed by the experiment. These results show the ability of the control filters to equalize the source room resonances over a broad frequency range as well as the feasibility of synthesizing an ideal acoustic diffuse field over the test panel surface below the room Schroeder frequency, i.e. below about 450 Hz [10].

Therefore, the methodology enables to limit deficiencies in diffusivity over a test panel at low frequencies in a reverberant room. A relevant point is now to assess the influence of diffusivity on the sound reduction index of the test partition, for the new approach to show improvements over the classical method which made use of a set of far-field uncorrelated loudspeakers in the source room. A theoretical study has already shown that the near-field array clearly eliminates the modal influence of the source room on the sound reduction index, especially when the source room has large dimensions and presents high modal overlap at low frequencies [5, 6].

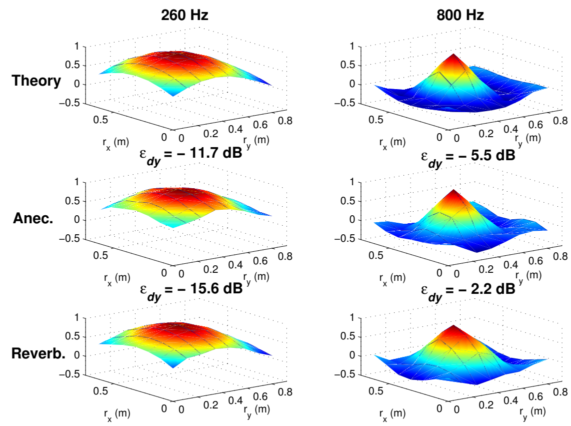


Figure 6 – Spatial correlation structures when perfect reproduction of an acoustic diffuse field at 7×8 *microphones is assumed (top), that achieved using* 4×4 *optimally driven loudspeakers either in a semi-anechoic environment (middle) or in a reverberant environment (bottom).*

Another limiting factor is the sensitivity of the panel sound reduction index to non-ideal diffuse sound fields or semi-diffuse sound fields, as those observed in practice over the source room side of the test panel, typically below the room Schroeder frequency. Below coincidence, both resonant and non-resonant modes contribute to the sound transmission mechanisms through the panel [11]. With large thin plates, predominant non-resonant transmission occur below coincidence and the resonant sound transmission component can then be neglected [12]. In these situations where non-resonant modes govern the sound transmission mechanism, diffusivity is an important factor since the panel response exhibits a strong sensitivity to inaccuracies in the excitations, in particular to deviations from ideal diffusivity. On the other hand, diffusivity is *a priori* not critical for predominant resonant transmission. The use of a near-field array of loudspeakers for the reproduction of an acoustic diffuse field at low frequency should therefore improve the measurement of the sound reduction index of large thin partitions located in a sound transmission suite.

CONCLUSIONS

The practical feasibility has been assessed of generating a random pressure field with spatial statistical properties similar to a diffuse acoustic sound field using an array of loudspeakers driven by optimal time-domain signals. Synthesis experiments have been performed with a near-field array of 16 uniformly distributed loudspeakers located above a grid of microphones covering the panel test side. It has been found that the use of an array of 4×4 acoustic sources enables accurate reproduction of an acoustic diffuse field up to about 680 Hz either in a semi-anechoic room or a small reverberant room, thus confirming the upper bound criterion requiring at least 2 acoustic sources per unit acoustic wavelength.

Moreover, the synthesis of a diffuse sound field on the source room side of a test partition appears to be feasible below the room Schroeder frequency, thus allowing the use of reverberant rooms with a reduced size for sound transmission tests. This methodology should also present advantages to reduce the variability of sound transmission measurements for large thin test partitions whose response is dominated by non-resonant modes.

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