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NUMERICAL ANALYSIS OF SOUND-PROOFING OF A TELEPHONE BOOTH BY MEANS OF ACTIVE AND PASSIVE METHODS

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Abstract

The paper deals with the development of a hybrid noise control system, aimed at improving the acoustical comfort of open telephone booths used in noisy streets. A shell-formed, open telephone booth was selected to be the subject of the feasibility study. High frequency noise is designed to reduce by placing sound absorbing lining on the inner surface of the shell. For the control of low frequencies an active system is to be developed.

The paper reports on the numerical acoustic analysis and optimisation of the hybrid control system. Traffic noise sources are modelled by a set of point sources or by a line source, and the sound field inside the booth is calculated by means of an indirect boundary element approach. The optimisation is aimed at selecting the appropriate number and placement of the secondary sources and prediction of the performance of the designed active noise control system. In order to have an efficient optimisation tool, a hybrid method consisting of numerical calculations and analytical simulation has been developed.

It was shown that the assumed ANC system is expected to be effective. Nevertheless, a few stumbling blocks related to the planned system are also pinpointed.

1. INTRODUCTION

The paper demonstrates the application of a hybrid simulation method, aimed at predicting the performance of an ANC system to be used in an open public telephone booth. The booth, or rather just a shell is assumed to be alongside a public road and is highly disturbed by traffic noise, therefore it is to be silenced by combination of both active and passive means. The current work is focussed on the feasibility of an effective ANC system and on the prediction of its expected performance.

In order to solve the problem effectively, a special hybrid method has been developed (Figure 1): the operation of the acoustic system is simulated by means of boundary element (BE) calculation with and without the added multiple-channel active noise control (ANC) system, and the operation of that ANC system is realised by a numerical simulation once again. In particular, the numerical acoustics software package SYSNOISE was used to perform the BE calculations and MATLAB for the ANC simulations.

This is a practical method to investigate a model of an acoustic noise problem, with and without its possible ANC solution, as well as to calculate the performance before it is realised. A special feature of the hybrid technique is that the calculations are carried out in the frequency domain by using the multiple-channel FXLMS algorithm, since this is the common and straightforward procedure in numerical acoustics.

Boundary Element Method (Sysnoise)	**ANC simulation** (Matlab)	**Boundary Element Method** (Sysnoise)
Calculate: - Pressure field in the acoustic system - Necessary frequency transfer functions	Calculate: - The optimal strengths of secondary sources	Recalculate the pressure field in the acoustic system

Figure 1 The special hybrid method

2. FREQUENCY-DOMAIN FXLMS ALGORITHM

The frequency-domain ANC, using the frequency-domain FXLMS algorithm [Kuo and Morgan], transforms the signals into frequency domain by means of the fast Fourier transformation (FFT). The updating procedure is based on the complex LMS algorithm: it is able to adapt both the real and imaginary part of the filter coefficient $w(n)$, while minimising the error signal.

The updating procedure can be found in many textbooks [Widrow]:

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu\mathbf{x}^*(n)e(n) \tag{1}$$

The frequency-domain multiple-channel FXLMS algorithm is depicted in Figure 2 [Hayking]. Let us consider that the reference signal, $x(n)$, is stored in an L-point data buffer:

$$\mathbf{x}(n) \equiv [x(n) \quad x(n-1) \quad \ldots \quad x(n-L+1)]^T, \tag{2}$$

and then transformed to the frequency-domain signal vector $\mathbf{X}(n)$:

$$\mathbf{X}(n) \equiv [X_0(n) \quad X_1(n) \quad \ldots \quad X_{L-1}(n)]^T = \mathrm{FFT}[\mathbf{x}(n)] \tag{3}$$

Signal $\mathbf{X}(n)$ is filtered (multiplied) by the corresponding adaptive or non-adaptive weights.

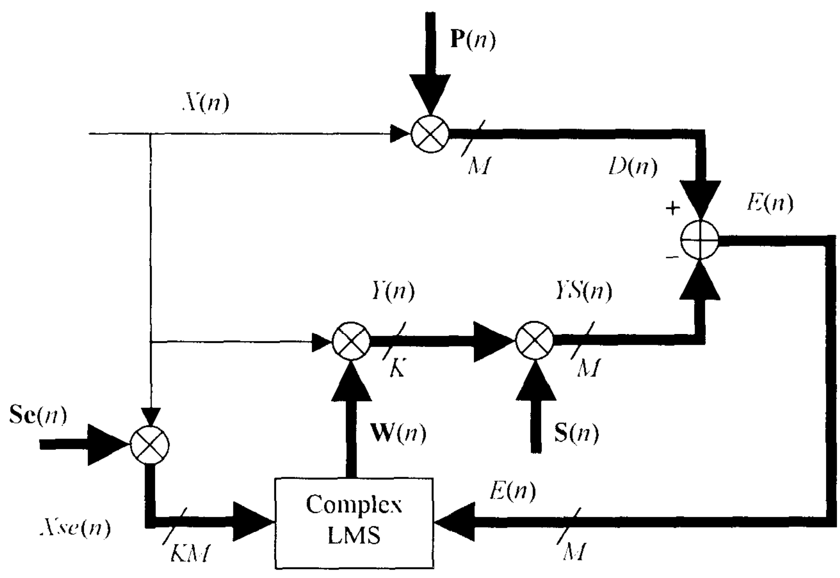

Figure 2 Multiple-channel frequency-domain FXLMS algorithm

The lth primary noise signal can be computed as:

$$\mathbf{D}_l(n) = \mathbf{X}(n)\mathbf{P}_l(n), \qquad l = 0,1,\ldots,M-1, \tag{4}$$

and the signal of the lth secondary source is also expressed as:

$$\mathbf{Y}_l(n) = \mathbf{X}(n)\mathbf{W}_l(n), \qquad l = 0,1,\ldots,K-1. \tag{5}$$

The error signals measured by M error microphone can be expressed as:

$$\mathbf{E}(n) = \mathbf{D}(n) - \mathbf{S}(n)\mathbf{Y}(n) \tag{6}$$

where the primary signal vector is defined as:

$$\mathbf{D}(n) \equiv [\mathbf{D}_1(n) \quad \mathbf{D}_2(n) \quad \ldots \quad \mathbf{D}_M(n)], \tag{7}$$

and the signal vector of the secondary sources is:

$$\mathbf{Y}(n) \equiv [\mathbf{Y}_1(n) \quad \mathbf{Y}_2(n) \quad \ldots \quad \mathbf{Y}_K(n)], \tag{8}$$

and the $M \times K$ block matrix between the secondary sources and the error microphones is the following:

$$\mathbf{S}(n) \equiv \begin{bmatrix} \mathbf{S}_{11}(n) & \mathbf{S}_{12}(n) & \ldots & \mathbf{S}_{1K}(n) \\ \mathbf{S}_{21}(n) & \mathbf{S}_{22}(n) & \ldots & \mathbf{S}_{2K}(n) \\ \vdots & \vdots & \ddots & \vdots \\ \mathbf{S}_{M1}(n) & \mathbf{S}_{M2}(n) & \ldots & \mathbf{S}_{MK}(n) \end{bmatrix} \tag{9}$$

By using the complex LMS algorithm, we can obtain the updating procedure of the frequency-domain multiple-channel FXLMS algorithm:

$$\mathbf{W}(n+1) = \mathbf{W}(n) + \boldsymbol{\mu}\mathbf{E}(n)\mathbf{Se}^H(n)\mathbf{X}(n) \tag{10}$$

where:

- $\mathbf{Se}(n)$ is the estimated $\mathbf{S}(n)$, (11)
- $\boldsymbol{\mu}(n) \equiv [\boldsymbol{\mu}_1(n) \quad \boldsymbol{\mu}_2(n) \quad \ldots \quad \boldsymbol{\mu}_M(n)]$, and $\boldsymbol{\mu}_l(n) = \dfrac{\mu}{\hat{P}_l(n)}$, $\qquad l = 0,1,\ldots L-1$, (12)
- $\hat{\mathbf{P}}_l(n) = (1-\alpha)\hat{\mathbf{P}}_l(n-1) + \alpha|\mathbf{Xse}_l(n)|^2$ (13)

3. SIMULATION RESULTS

The investigated acoustic systems, i.e. the telephone booth, is modelled by a geometrical mesh for the BE calculation, as it is depicted in Figure 3. This highly simplified mesh consists of several small plane surfaces. To simulate any kind of sound source we have to define the velocity of a part of the model surface for the investigated frequencies. These vibrating surfaces represent the noise sources and the secondary loudspeakers driven by the ANC system. In order to get information (pressure, particle velocity, etc.) for any other spatial points (different from the nodes of the surface mesh) the additional points have to be defined. These additional points are referred to as field points, or field point mesh.

Let us define the frequency-domain transfer functions between the vibrating surface and the pressure at the desired field point, using unity velocities for all investigated frequencies. The task of the active noise control system is to approach these transfer functions, in order to generate the necessary „anti" pressure with the least possible error.

The general form of these frequency transfer function is define as:

$$p_{(x,y,z)}(\omega) = F(\omega)v(\omega) \tag{14}$$

where the particle velocity excitation is unity.

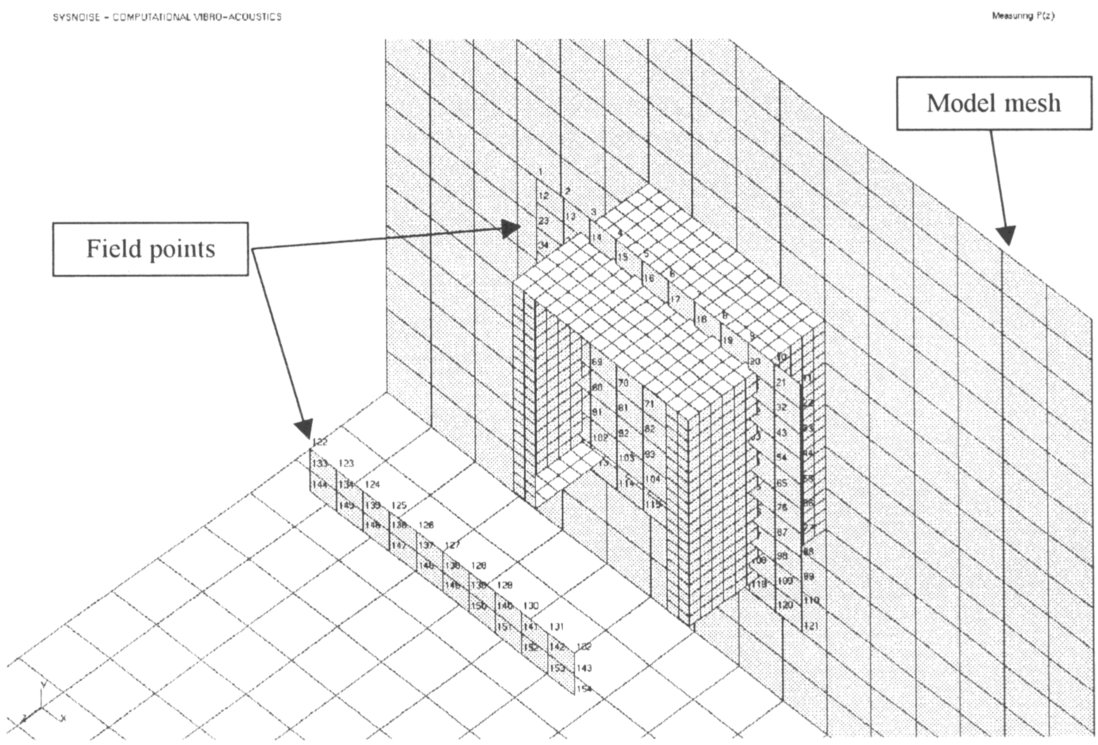

Figure 3 The phone booth

The simulation consists of three steps:

- Calculate spatial pressure values and the necessary transfer functions by means of the boundary element method, or in another words, calculate the pressure values at the field points.
- Make a model of a multiple-channel frequency-domain ANC system simulation by means of a signal processing simulation, using MATLAB.
- Use the calculated optimal parameters of the ANC, calculate the pressure field again.

First, the position of the reference microphone and the error microphones has to be arranged. One reference and four error microphone points were defined, indicated by point number 61, 63, 72, 74, 138 (Figure 3). Then four secondary sources were chosen next to phone (these are the parts of the model surface), two on its left and two on its right, and the traffic noise was generated by a line source placed 3 meters far, parallel to the phone booth.

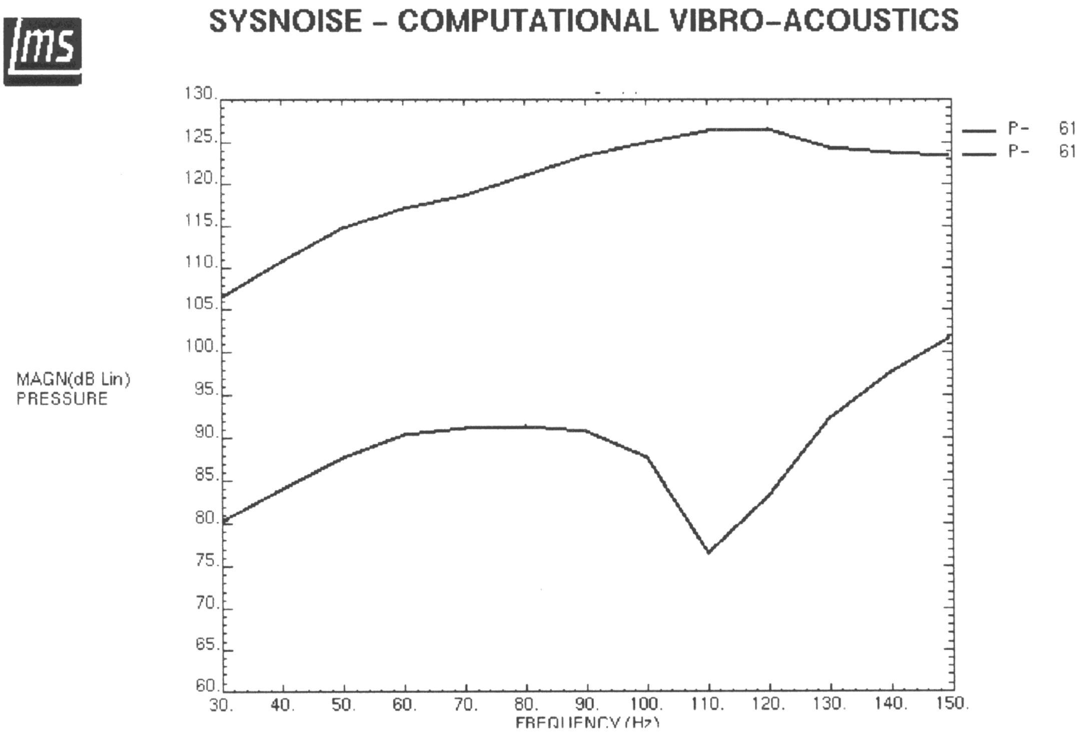

Figure 4 Pressure at point of the first error microphone (field point 61), with ANC (below), and without (above)

Eventually, the three steps of the simulation as discussed above are being executed.

The pressure value at the point of the first error microphone is shown in Figure 4. The pressure distribution in the whole plane of error microphones at 100 Hz is shown in Figure 5.

4. CONCLUSION

As one can see, this hybrid method seems to be a viable procedure to investigate an unknown acoustical system including ANC. In case of phone booth the ANC system had about 20 dB attenuation at all error points. This result is relatively high and virtually does not depend on the location: similar results can be obtained inside the phone booth along in the whole plane of error microphones.

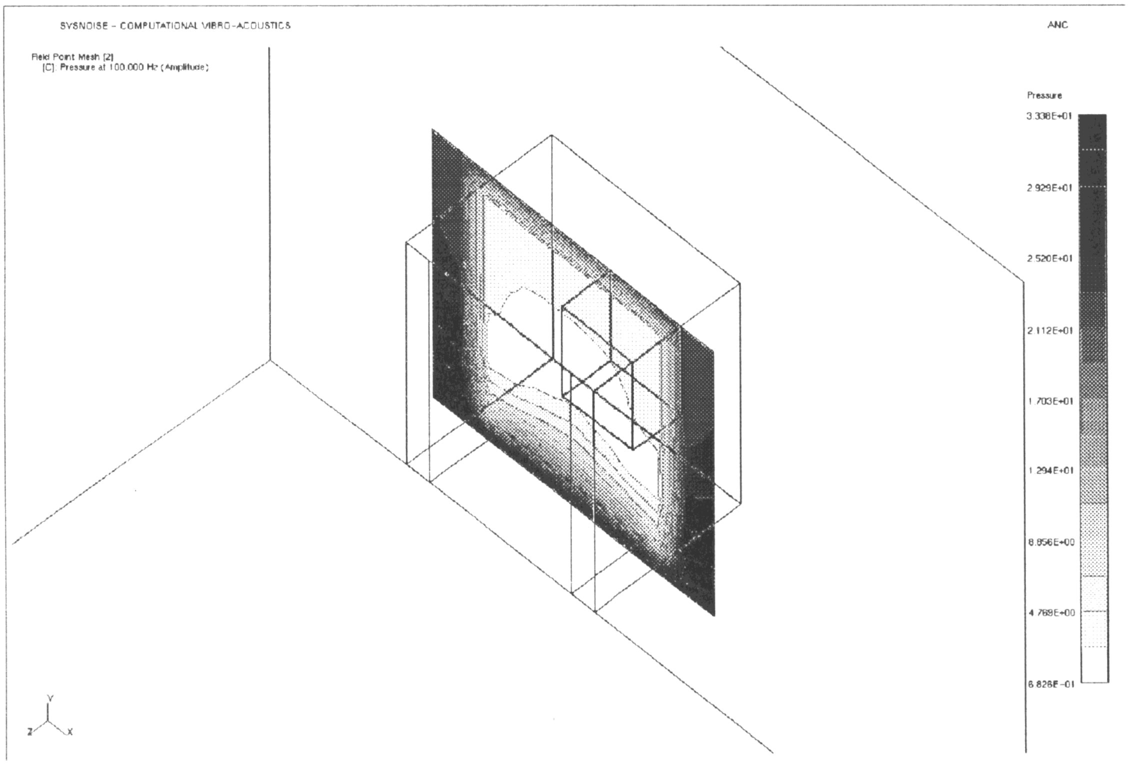

Figure 5 Pressure distribution in the plane of error microphone at 100 Hz (white indicates the pressure minimum)

However, one has to mention a number of practical problems here. In order to prevent the ANC system from being non-causal, the electric delay from the reference microphone to the secondary loudspeaker must be shorter then the acoustics delay. Thus the reference microphone must be placed far enough from the phone booth, or the sampling rate must be high enough. Unfortunately, the correlation between the undesired noise and the input signal from the reference microphone can decrease to unacceptable level if the reference microphone is located too far from the place where the cancellation is to be reached.

While the numerical ANC simulation generates approximately zero error values, the maximum decrease in the numerical predicted pressures is not more than 20 dB. This difference is caused by the numerical instability of the boundary element method. As a consequence, the theoretically conceivable minimum of the error cannot be reached in practice.

Finally, due to the varying geometry caused by the presence of the speaking person in the phone booth, on-line secondary path identification is an absolute must, which raises the convergence time significantly [Kuo and Morgan].

To summarise, we have established that in principle one can get a good estimation of the potential performance of an ANC realisation by using the developed hybrid method. Nevertheless, in case of the phone booth the realisation poses a number of principal and practical problems.

5. REFERENCES

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