*inter.noise 2000*
***The 29th International Congress and Exhibition on Noise Control Engineering***
***27-30 August 2000, Nice, FRANCE***

**I-INCE Classification:** 7.6

# SIMULATION OF A LOW FREQUENCY NOISE INSIDE A CAR CABIN BASED ON ANALYTICAL-NUMERICAL SOLUTION OF THE WAVE EQUATION

**A. Kovtonyuk*, V. Galanenko**, A. Kalyuzhny***

* Scientific-Production Enterprise, 27, acad. Krymsky Str., 03142, Kiev, Ukraine

** National Technical University of Ukraine, Apt. 29, N 27, Bol'shaya Vasil'kovskaya, 01004, Kiev, Ukraine

Tel.: (+380 44) 452-0160 / Fax: (+380 44) 452-0160 / Email: aleko@delta-net.kiev.ua

**Keywords:**
NOISE, CAR CABIN, SIMULATION, ANALYTICAL-NUMERICAL METHOD

**ABSTRACT**
The mathematical algorithm and the respective software for simulation of speech and noise sound fields inside a car cabin are developed. The software is examined experimentally. Application of this software to the analysis of the effectiveness of a small microphone array is demonstrated.

## 1 - INTRODUCTION

Speech communication and speech recognition systems as the parts of a car hands-free interface work in presence of noise which appears when a car moves. The real effectiveness of the algorithms these systems are based on depends significantly on noise level, noise field spatial and spectral structure. Surely the examination of these systems performance in presence of noise is an obligatory stage of their development. At a final stage of designing process one can examine the system effectiveness experimentally in a really moving car or in the special laboratory [1] by using the pilot setup. But at the earlier stage (when algorithms are just developed) the computer modeling seems to be more expedient. The main purpose of the presented work is to develop convenient mathematical model of sound field within a car (or another vehicle) and the respective software to simulate speech signal at noise background. This software is expected to be useful tool for examination of different kinds of speech processing algorithms.
The software has been verified experimentally: the engine noise spectrum within the car cabin has been measured and then has been matched to the spectrum of the simulated noise. In the last part of the paper an example of the developed software application is demonstrated: the effectiveness of small microphone array is examined and discussed.

## 2 - GENERAL DESCRIPTION OF THE SIMULATION ALGORITHM

The model consists of the rectangular prism which approximates the cabin geometry, of the set of point sources which approximate a distributed source of noise initiated by the cabin surface vibration and of unique point source as an approximation of speech signal source. The height of the prism is equal to the width of the cabin and the prism base is a polygon which approximates the cabin profile (see fig. 1).
Acoustical properties of the prism boundaries are defined by the frequency dependent sound absorption coefficients of the covering materials. The number of noise sources, their disposition and cross correlation (dependent on frequency) are defined by user. We start from the experimentally measured (or defined phenomenologically) spectrum of vibrations of the respective part of a cabin surface. Then the program generates N statistically dependent random sequences of the same averaged spectrums. When simulating speech signal we start from a speech sample obtained by use of microphone and sound card.
For each frequency the point source sound field is the Green's function $S(\omega, \mathbf{r}) = S_0(\omega) * G(\omega, \mathbf{r}/\mathbf{r}_0)$ of the respective 3D boundary problem for the wave equation. The program computes Green's functions by the analytical-numerical method (presented below) for low frequencies (< 1000 Hz) or by imaginary sources method (modified in such a manner to account for possible shadings) for high frequencies. After

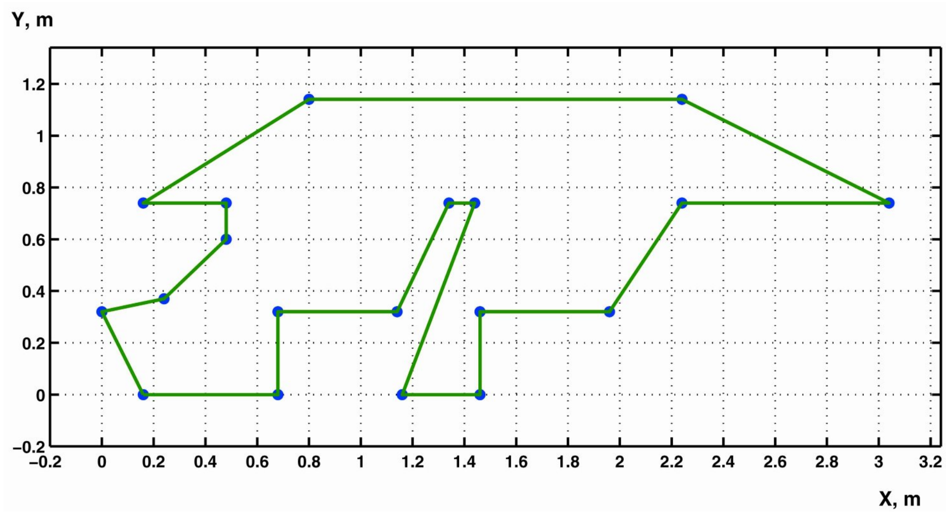


**Figure 1:** Cabin profile approximation.

multiplying noise spectrums by respective Green's functions, summarizing the incomes of all the sources and computing reverse Fourier transformation we obtain the simulated noise at specific point within the cabin. We obtain the simulated speech signal in analogous way. Optionally weighted mixture of speech and noise can be used further for examination of speech processing algorithms. The program allows the audition of simulated signals before and after processing.

## 3 - GREEN'S FUNCTION COMPUTING FOR LOW FREQUENCY RANGE

The peculiar feature of the discussed problem is that the speech frequency range is comparatively wide. We can use the ray acoustics approach for high frequencies but the small (or insufficiently big) dimensions of a cabin in the wavelength scale do not allow the ray approach for low and middle frequency subranges. In the case analytical computation of the sound field is not available due to complicated geometry of a cabin. Numerical solution of the wave equation for 3D region is difficult as well. The combined analytical-numerical method is proposed below.

For each frequency we formulate boundary problem for the wave equation with generalized Neumann boundary conditions. In accord with the cross sections method [2], [3] we can find the solution as an expansion in eigenfunctions $u_n(z)$ of the respective cross section operator. Then we derive:

$$G(\omega, \mathbf{r}/\mathbf{r}_0) = \sum_{n=1}^{\infty} \frac{ik}{4\pi} \frac{u_n(z_0)}{\|u_n\|} c_n(x, y, x_0, y_0)\, u_n(z) \qquad (1)$$

Each of the unknown functions $c_n(x, y, x_0, y_0)$ can be found by numerical solving of the equation

$$\frac{\partial^2 c_n}{\partial x^2} + \frac{\partial^2 c_n}{\partial x^2} + \gamma_n^2 c_n = \delta(x - x_0, y - y_0) \qquad (2)$$

with the generalized Neumann boundary condition using the finite elements technique. Here: $\gamma_n^2 = k^2 - \beta_n^2$ and $\beta_n$ is the respective root of the next equation:

$$q\cos(\beta H) - \beta\sin(\beta H) = 0 \quad - \text{for the symmetric modes or}$$
$$q\sin(\beta H) + \beta\cos(\beta H) = 0 \quad - \text{for the antisymmetric ones,}$$

where $2H$ is equal to a cabin width, $k$ is the wavenumber.

Under practical computing the series (1) was reduced into finite sum. The number of summands and required number of finite elements go up with frequency increase. Practically, the accuracy of computation is sufficiently big if we save all the summands for which $\mathrm{Re}(\beta_n) < k$ and 2-3 summands for which $\mathrm{Re}(\beta_n) > k$. Required number of finite elements was no more than 1500 for the configuration shown at fig. 1 within the frequency range down to 1000 Hz.

## 4 - COMPARISON OF THE SIMULATED AND REAL NOISES

Vibrations at the surface M (see fig. 1) initiated by the engine and sound noise at a point near surface A were measured and saved in step into computer memory. Then the vibrated surface M was approximated by the set consisted of 13 point sources with frequency dependent cross correlation (the higher was a frequency the smaller was correlation radius). The spectrum of each source was adequate to the measured spectrum of vibrations. The simulated sound noise was produced for the same point near surface A. Fig. 2 gives the averaged spectrums of real and simulated sound noises in the point mentioned above.

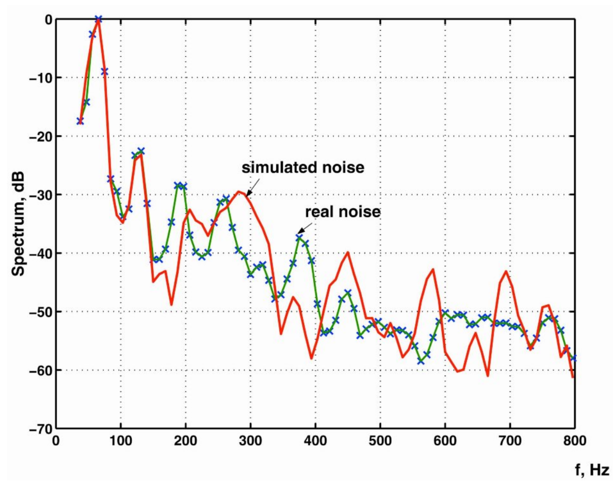


**Figure 2:** Spectrums of the real and simulated noises.

The compared spectrums are not identical but the spectrum of the simulated sound renders basically behavior of the real noise.

## 5 - AN EXAMPLE OF APPLICATION OF THE SIMULATION ALGORITHM

We have applied simulation algorithm presented above to the analysis of effectiveness of small microphone arrays in a car cabin which are expected to give more ample opportunities for selection of a voice call at noise background comparatively to a single microphone. Optimal processing of the microphone outputs consists in their weighted summation which is individual for each frequency component and dependent on spatial correlation at array aperture and further filtration. Adaptive form of optimal speech processing involves preliminary estimation of cross correlation and spectrums.

We investigated effectiveness of linear array of length 0.15 m consisted of 5 omnidirectional microphones using the algorithm of sound field simulation within a car cabin. Fig. 3 gives the waveforms of the initial (pure) speech signal (3a), this signal at noise background (3b) and cleared mixture of speech and noise at the output of optimal speech processing algorithm ( 3c). Adaptive algorithm demonstrates closed results.

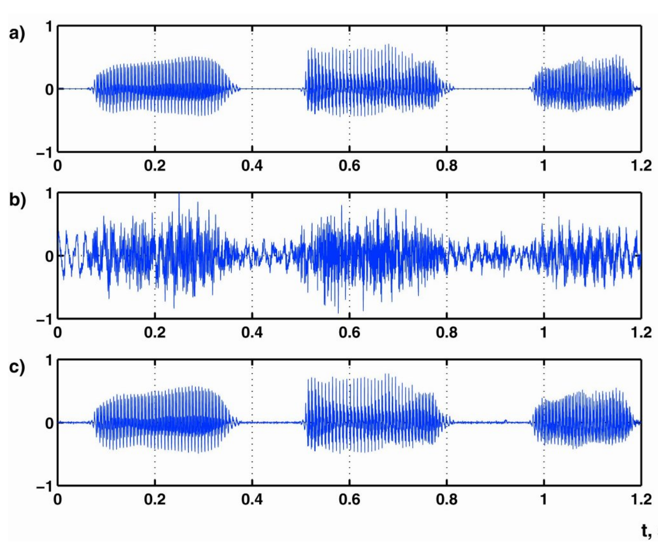


**Figure 3:** Waveforms of the speech signal: a – pure speech, b – speech at noise background, c- signal at the output of 5-elements microphone array after matched spatial processing.

**REFERENCES**

1. **F. Kettler, H.W. Gierlich, W. Krebber**, Hands-Free car Kits - A Performance Discussion of "State of the Art" - Solutions, In *137th meeting of the ASA and 2nd conv. of the EAA, Berlin*, 1999

2. **V. B. Galanenko**, On coupled modes theory of two-dimensional wave motion in elastic waveguides with slowly varying parameters in curvilinear orthogonal coordinates, *J. Acoust. Soc. Am.*, Vol. 103 (4), pp. 1752-1762, 1998

3. **H. Weinberg, R. Burridge**, Horizontal ray theory for ocean acoustics, *J. Acoust. Soc. Am.*, Vol. 55 (1), pp. 63-72, 1974