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A new method for auralisation of airborne sound insulation

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ABSTRACT

Auralisation techniques have quickly grown in popularity during the last few years, mostly due to their application in virtual reality, the gaming industry and room acoustics. The application of auralisation to sound insulation has also become a valuable tool to assess the sound insulation of buildings, and to suitably describe to the non-acousticians how is the sound experience hidden behind the swarm of acoustic quantities. Auralisation is used now in commercial software to show the relevance of some acoustic features that are often not well-understood by users and constructors until too late.

This paper presents a new method to compute the binaural sound transmission between rooms. As the established method, this approach makes use of the EN 12354 sound insulation predictions to build up the auralisation of the sound insulation predictions. In addition, it takes into account the spatial variation of the sound pressure field inside arbitrarily shaped rooms, enabling the user to experience binaural sound from arbitrary sound sources and varying the source and receiver position. Additionally, the method addresses the problem of synthesising the room impulse response from the one-third octave band values of the reverberation time obtained as per ISO 3382-2:2008.

The new method increases interaction with the simulated environment creating a deeper, more realistic and immersive scene and leading to more accurate subjective evaluations of sound insulation.

1. Introduction

The first applications of auralisation techniques to sound insulation were due to Schmitz and Zier [1], but they were Vorländer and Thaden who firstly settled the mechanisms of the auralisation of airborne sound insulation [2].

Starting from the EN 12354 calculation method [3] they developed an algorithm that, through binaural technology, made it possible to subjectively evaluate the sound insulation predictions on standard computers. With time, the auralisation of sound insulation results became more and more used, standing as the perfect link between the results from predictions models and the perception by the population [4]. In consequence, [2] was progressively implemented in the most relevant sound insulation software tools.

More that a decade after, some improvements have been made to the algorithm [5]. Different authors have proposed the use of diffusive methods [6] and ray-tracing techniques [7] to include the spatial behaviour of transmitted sound. However, these latter contributions do not meet real time constrains, due to their high computational load. This fact, and its perfect coupling to the sound insulation results of EN 12354, made [2] the unrivalled method for auralising sound insulation results.

In the present paper the established method is reviewed, comprising the contributions presented in [2,5,4,8]. Some aspects of the method are discussed and an alternative approach is introduced that preserves real-time constrains. In addition, it incorporates the spatial pattern of the sound field in arbitrary shaped rooms and the radiation patter of sound sources. The perfect coupling to EN 12354 is not only preserved, but improved, with the introduction of a technique that approximates the room impulse response from the reverberation time expressed in one-third octave band values.

2. The established method

The standardised sound level difference D_{nT} between two rooms is given by [3],

$$D_{nT} = L_S - L_R + 10\log\frac{T}{0.5\ \mathrm{s}}, \tag{1}$$

where L_S is the average level in the source room, L_R is the average level in the receiver room and T is the reverberation time in the receiver room. We can express D_{nT} in terms of the transmission coefficients $\tau_{ij} = 10^{-0.1R_{ij}}$ of all transmission paths ij [3],

$$D_{nT} = -10\log\sum_{\forall ij}\tau_{ij} + 10\log\frac{0.32V}{S_D}, \tag{2}$$

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where V is the volume of the receiving room in cubic metres and S_D is the area of the separating element in square metres. By Inserting (2) into (1), L_R can be expressed in terms of τ_{ij},

$$L_R = L_S + 10\log\sum_{\forall ij}\tau_{ij} + 10\log\left(\frac{S_D}{0.32\ \mathrm{V}}\frac{T}{0.5\ \mathrm{s}}\right), \tag{3}$$

which, in energetic notation, yields

$$p_R^2 = p_S^2\frac{S_D}{0.32\ \mathrm{V}}\frac{T}{0.5\ \mathrm{s}}\sum_{\forall ij}\tau_{ij}, \tag{4}$$

where p_R^2 and p_S^2 are the mean squared pressure values in the receiving and the source room respectively. Expression (4) makes it possible to calculate the one-third octave band spectrum of the sound pressure in the receiving room from τ_{ij}, the outcome of the EN 12354 method. However, in building acoustics, the transmission coefficients τ_{ij} are usually expressed in one-third octave bands. To convert (4) into a suitable audio filter, a 4096 points cubic spline interpolation is applied, as suggested in [2], to convert the 21 available values in the extended range to an appropriate frequency resolution for audio processing.

This interpolated version of (4) provides loudness and colouration, but some important effects are lost, such as the temporal effect of the receiving room impulse response, the balance between direct and reverberant sound and the time delays experienced by the different transmission paths. In [8], radiating elements are approximated through point sources and the balance between reverberant and direct sound is obtained through the ratio of energies

$$\frac{E_{rev}}{E_{dir}} = \frac{16\pi r^2}{A}, \tag{5}$$

where E_{rev} and E_{dir} are, respectively, the energies of the reverberant and direct field at a distance r from a point-source, and A is the equivalent absorption area of the receiving room. Assuming that the reverberant and the direct field are uncorrelated, the contribution of the transmission path ij to the mean squared pressure can be written in terms of the reverberant and the direct contribution of path ij, respectively $p_{R,ij,rev}$ and $p_{R,ij,dir}$, as

$$p_{R,ij}^2 = p_{R,ij,rev}^2 + p_{R,ij,dir}^2. \tag{6}$$

Inserting (5) into (6), it follows that

$$p_{R,ij,rev}^2 = \frac{16\pi r_{ij}^2}{16\pi r_{ij}^2 + A}\ p_{R,ij}^2, \tag{7}$$

$$p_{R,ij,dir}^2 = \frac{A}{16\pi r_{ij}^2 + A}\ p_{R,ij}^2, \tag{8}$$

where r_{ij} is the distance from the radiating element j of the transmission path ij to the receiver position.

To include the temporal effect of the receiving room impulse response $h(t)$, [2] introduces a modified version of $h(t)$ where

1. the direct sound has been removed,
2. it has been equalised approximately to white spectrum, and
3. it has been normalised in energy.

In [2] this normalised impulse response, hereinafter denoted by $\bar{h}(t)$, is obtained from a measurement of the impulse response of a given room.

Now, (4) can be expressed in terms of the instantaneous sound pressure. Letting $p_S(t)$ be the instantaneous sound pressure in the source room, the contribution of the transmission path ij to the instantaneous sound pressure in the receiving room $p_{R,ij}(t)$ can be written as

$$p_{R,ij}(t) = \sqrt{\frac{S_D}{0.32V}\frac{T}{0.5\ \mathrm{s}}\frac{\tau_{ij}}{16\pi r_{ij}^2 + A}}\ p_S(t) * \left[\sqrt{A}\ \delta(t - r_{ij}/c) + \sqrt{16\pi r_{ij}^2}\ \bar{h}(t)\right], \tag{9}$$

where the symbol $*$ denotes the convolution, and $\delta(t - r_{ij}/c)$ is the Dirac's delta delayed r_{ij}/c seconds. Finally, the binaural sound is produced by substituting the Dirac's delta in (9) by the head related impulse response HRIR $(t - r_j/c, \theta_j, \phi_j)$ for an angle of incidence (θ_j, ϕ_j), and the total instantaneous sound pressure $p_R(t)$ perceived by the listener is obtained by superposition,

$$p_R(t) = \sum_{\forall ij} p_{R,ij}(t). \tag{10}$$

Some simplifications are implicit in this formulation, most of them inherit from the statistical approach in which the EN 12354 is based. In first place, the incident sound pressure is considered to be equal for all transmission paths, since the statistical energy analysis (SEA) assumes that energy is evenly distributed within the room. Congruently, the same incident sound power hits all elements, independently of the source position and room geometry. Additionally, as pointed out by Rindel [9], this method does not take into account the influence of the source room reverberation, the directionality of the sound source, or the ratio between direct and reverberant energy inside the source room.

Recent models have addressed some of these issues, based on finite element method [6] and on ray tracing techniques [7]. However, the computational cost of numerical and ray-tracing models prevents them to be used for real-time simulations.

3. An alternative method

Above the Schoereder frequency, the direct field can be assumed to decay as in free-field conditions, whilst the reverberant field is evenly distributed. This behaviour is conveniently modelled through the classical theory for sound propagation in rooms,

$$L_S = L_W + 10\log\left(\frac{Q_S}{4\pi r^2} + \frac{4}{A_S}\right), \tag{11}$$

where L_S is the sound pressure level inside the room, L_W is the source sound power level, Q_S is its directivity, r is the distance from the source to the evaluation point, and A_S is the equivalent absorption area of the source room. Notice that (11) automatically introduces the influence of the room reverberation, the directionality of the sound source, and the same ratio between direct and reverberant energy considered in [2].

3.1. Sound field in the source room

If (11) is expressed in energetic notation, the mean squared pressure at any point within the source room is given by

$$p_S^2 = 400P_a\left(\frac{Q_S}{4\pi r^2} + \frac{4}{A_S}\right), \tag{12}$$

where $P_a = 10^{-12}10^{0.1L_W}$ is the source acoustic power in watts. Let us define the instantaneous sound pressure, $s(t)$, corresponding to a recording of the sound source taken under free-field conditions that has been normalised in power. Using the normalised impulse response of the source room $\overline{h_S}(t)$ defined in [2], the instantaneous sound pressure $p_S(t)$ in at any point r inside the source room is given by

$$p_S(t) = 20\sqrt{P_a}\ s(t) * \left(\sqrt{\frac{Q_S}{4\pi r^2}}\ \delta(t - r/c) + \sqrt{\frac{4}{A_S}}\ \overline{h_S}(t)\right), \tag{13}$$