Evaluation and Modeling of Small Rooms

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Summary: Technical rooms and reference listening rooms are usually small in the acoustic sense, and therefore their design and objective qualification need special care. Recommended criteria in the design of such rooms do not suffice in practice, hence new objective parameters are needed that describe the subjective impression, and can be used to optimize the design procedure for. Several listening tests and modeling were carried out, and with the new objective parameters found, an optimized approach in the design and modeling was developed.

INTRODUCTION

Small rooms were always a challenge for the acoustician, because classical and well-known methods used for larger spaces, like concert halls or theaters, are not always applicable. Similarly, objective parameters, trying to describe the subjective impression differ. However, objective parameters are crucial to be able to measure the perceived quality of the sound field, and because the ones found in recommendations are inadequate in practice, new parameters are needed. In this paper, after a short introduction of the subjective tests and their results, a new approach is presented in the application of the new objective parameters.

SUBJECTIVE TESTS AND OBJECTIVE PARAMETERS

Although there already recommendations on parameters calculated from binaural responses **(irodalom)**, in this paper only parameters that are derived from impulse responses, are investigated. Our experiences show, that the two main factors for a good perception of the sound field in a small room are, that:

- no distinct reflections or reflection patterns are preferable (diffuse field), and
- there should be no strong reflection within the first 15-20ms after the direct sound.

Trying to describe these features, the following objective parameters are proposed:

$$k_1(t) = 10 \cdot \log_{10}\left[\int_0^t p^2(\tau)d\tau \Big/ \int_0^\infty p^2(\tau)d\tau\right][dB] \tag{1}$$

$$k_2(t) = 10 \cdot \log_{10}\left[\int_0^t p^2(\tau)d\tau \Big/ \int_t^\infty p^2(\tau)d\tau\right][dB] \tag{2}$$

The parameters k_1 and k_2 express the ratio of received energy before and after a given time constant t, which is usually low (5-20ms) for small rooms and higher (50-80ms) for larger spaces. The two parameters are equivalent and can be expressed from each other. A combination of two ratios is the modified M-factor **(irodalom)**:

$$M = k_2(5ms) - k_2(20ms)\ [dB] \tag{3}$$

Finally, the well-known center time:

$$t_S = \int_0^\infty t \cdot p^2(t)dt \Big/ \int_0^\infty p^2(t)dt \, [s] \tag{4}$$

The subjective tests were carried out in 7 different listening rooms or technical rooms of the Hungarian Radio with the same 20 subjects. The subjects were trained before the tests. The evaluation **(Borsi et. al.)** of the questionnaires gave the correlation shown in Table 1.

TABLE 1: Correlation coefficients of subjective features and objective parameters

	k_1	k_2	t_S	M
spatial impression	0.74	0.74	0.78	0.69
timbre	0.43	0.43	0.75	0.81
transparency	0.45	0.45	0.88	0.75
stereo accuracy	0.79	0.79	0.71	0.68

CHOOSING THE MODELING METHOD

There are several modeling methods in use nowadays, but every technique is limited theoretically or computationally. For our investigations the triangular beam-tracing method (TBM) was chosen, because of its simplicity. However, this method is based on the laws of geometrical acoustics, and is limited especially at relatively low frequencies accordingly.

COMPARING MEASURED AND PREDICTED RESULTS

The accuracy of the chosen modeling method was examined using the measured room responses. For the comparison of the measured and predicted results the energy decay curve (EDC) was chosen, because it is hard to match wide band measured impulse responses and predicted monochromatic echograms, and because most of the parameters used for the objective characterization are in direct connection with EDCs. The EDCs show the temporal distribution of received energy and are calculated from the following formula:

$$e(t) = 1 - \left(\int_0^t p^2(t)dt \Big/ \int_0^\infty p^2(t)dt \right) \tag{5}$$

Global parameters, like reverberation times are also calculated from EDCs, and predicted and measured values agreed well. However, other parameters showed differences that are not negligible.

ERROR SOURCES OF TBM

There are two main groups of error sources in using the chosen method:

- parameter errors - discrepancy of parameters describing the properties of surfaces (absorption coefficients, diffusion coefficients, etc.), sources, receivers, and the medium.
- errors of the method - limitations of geometrical acoustics, imprecision of approximations

It is a well-known problem in practice to use the correct parameters in the model, therefore in the following part of the paper the effect of parameter errors are investigated. Let us assume that the errors due to the limitations are smaller than the errors introduced by the errors of parameter, i.e. a simple TBM can predict the temporal distribution of detected energy at a given receiver point accurately enough.

CALCULATING THE PARAMETERS - AN INVERSE METHOD

In order to examine the effect of parameter errors, we developed a method for matching measured and predicted EDCs. The idea is that from the band-limited measured EDCs and from the timings of modeled reflections the amplitudes of these reflections can be calculated so that predicted and measured EDCs can fit with minimal error.

Echograms give the temporal distribution of detected reflections. A reflection in the echogram is a detection of energy arriving at a given time. Knowing the amplitudes of these reflections, parameters of different phenomena (e.g. reflection) can be calculated. The transfer-function (TF) of a received reflection is in general:

$$H_i(\omega) = H_{AIR}(\omega, r_i) \cdot H_{PROP}(\omega, r_i) \cdot H_{SRC}(\omega, \varphi_i, \nu_i) \cdot H_{REC}(\omega, \varphi_i, \nu_i) \cdot \prod_n^{N_i} H_{SURF,n}(\omega) \qquad (6)$$

where H_i is the TF of the i-th detected reflection, H_{AIR} is for the absorption of air, H_{PROP} is for the attenuation of propagation, H_{SRC} and H_{REC} are the TF of source and receiver, and $H_{SURF,n}$ is the TF of the n-th reflecting surface in the i-th path.

To try the theory in practice, a well documented room was measured and modeled. For the sake of simplicity, all non-specular phenomena were neglected, sources and receivers were supposed to be omnidirectional. With these simplifications, after correcting the amplitudes for the air and the attenuation of propagation, and writing in terms of energy, Eq. 6. gets the simple form of

$$E_i'(f) = \prod_n^N \left(1 - \alpha_n(f)\right)^{m_{i,n}} \qquad (7)$$

Here the prime on E_i denotes the correction of air and attenuation of propagation, α_n is the absorption coefficient, finally $m_{i,n}$ shows how many times surface n took part in path i. After taking the logarithm of the equations, we get a linear equation system, where only the absorption coefficients are unknown. Since the equation system is rather overdetermined (we can have about 20,000-50,000 reflections, i.e. equations, and only 10-40 surface types), the equation system was solved with an iterative procedure featuring exponentially decaying correcting vectors.

$$\overline{\alpha}_{k+1} = \overline{\alpha}_k + \overline{\gamma}_k \cdot \overline{\delta}_k \qquad (8)$$

Here α_k is the solution in the k-th step, γ_k is the exponentially term causing the exponential decay of the error δ_{k+1}. Some results are shown in Fig. 1 and 2.

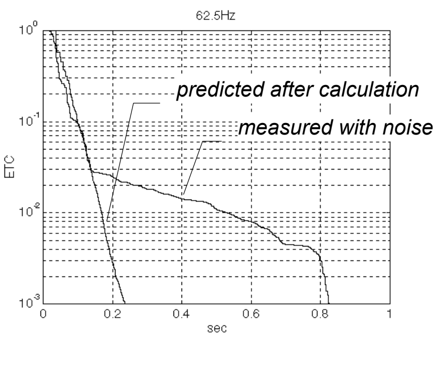

FIGURE 1: Measured and calculated EDC

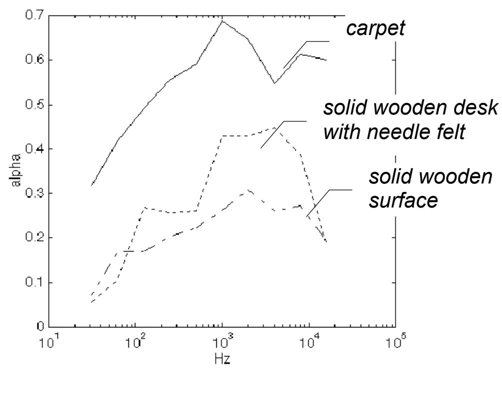

FIGURE 2: Calculated absorption coefficients

It can be seen, how the slope of the predicted EDC fits the measured one even with this very simplified model. However, the predicted EDC is much more coarse, and looks stepped. This is, because non-specular reflection was neglected, so reflected (not absorbed) energy arrives as an ideal impulse, which is not the case, of course in the real world. Also, the exponentially decaying feature of the iterative solution helped to attenuate the effect of the noisy part in the measured EDC. The calculated absorption coefficients look reasonable.

The problem with this inverse calculation procedure is that timing is crucial, especially in the early part of the impulse response, and it is hard to determine for example, where the direct sound is in a band limited impulse response at low frequencies. For the experiment above, the time of the absolute maximum was supposed to be the time of the direct sound.

It is obvious, that by involving other phenomena e.g. diffusion, angle-dependent absorption etc., their parameters give more degree of freedom in this process, so other parameters can be calculated also, and greater accuracy can be achieved.

THE STRAIGHTFORWARD DESIGN PROCEDURE

The common practice in the design of a room is to start with an estimation of the global parameters like reverberation time, and then to create a model and try different configurations of the source and the receiver, and try different distribution of given surface types with estimated parameters, finally run the model and see whether the results are satisfactory or not.

Using the inverse idea, outlined above, the design and modeling can be much more straightforward. One has to specify only the required objective parameters, derive an idealized EDC matching this criteria, and finally calculate the parameters of the model that are necessary to get these results.

THE IDEAL EDC

For the sake of simplicity, the ideal EDC is supposed to consist of exponentially decaying parts, each of which is in direct connection with a specified objective parameter, like reverberation times, k_1 or M, etc. Our experiences show, that apart from the very beginning of the EDC and the noisy late part, this assumption is not far from reality.

For example, if we specify k_1 and T_{60} only, we can calculate the parameters of an ideal EDC consisting of 3 parts, with 2 unknown slopes (Fig. 3).

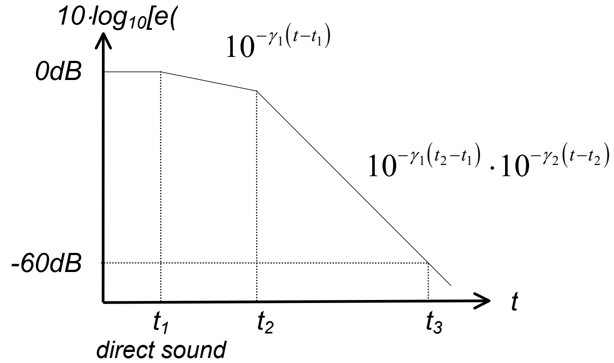

FIGURE 3: Ideal EDC for k_1 and T_{60}

With simple calculations, one can derive the parameters of this particular EDC:

$$\gamma_1 = \frac{\log_{10}\left(1-10^{k_1/10}\right)}{\left(t_1 - t_2\right)}, \quad \gamma_2 = \frac{6-\gamma_1\left(t_2 - t_1\right)}{T_{60} + t_1} \tag{7}$$

EXAMPLE OF APPLICATION

As an example of this direct design approach, we took a simple shoebox room with 3×5×8m size, where the position of the source and the receiver is fixed, and the surfaces consist of 1×1m pieces (Fig. 4). The goal is to determine, what absorption should these surface pieces have in order to achieve a k_1=-1dB and T_{60}=500ms response. The source and the receiver is supposed to be omnidirectional, non-specular phenomena are neglected.

With the EDC fitting process, this direct approach gave the result in Fig. 5.

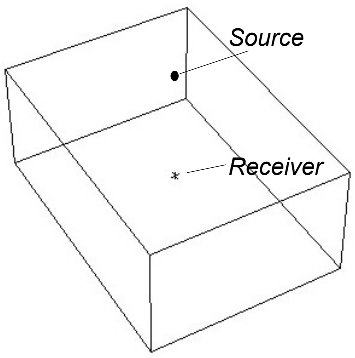

FIGURE 4
The shoebox example

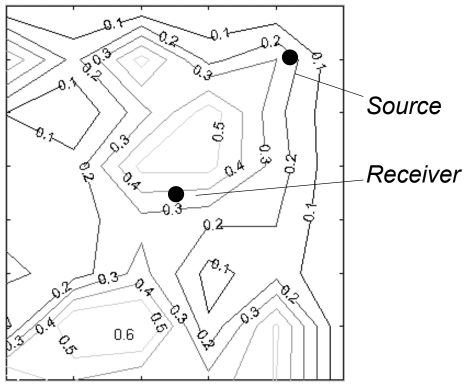

FIGURE 5
Calculated absorption coefficients on the floor

Of course, this procedure can be extended with partially specified absorption coefficients (e.g. there is a fixed window in the room), and since the idea of EDC fitting is not limited to only specular reflections, approximations of non-specular phenomena may be taken into account if necessary.

Besides the properties of surfaces, the properties of source and receiver may be also calculated. Or, the results can be optimized not for just one receiver point. Moreover, if we get strange results (e.g. negative absorption), it shows that the specified values of the parameters cannot be achieved with the given geometry and source-receiver configuration.

CONCLUSIONS

Based on the subjective tests, new objective parameters were developed. A new direct inverse approach was also introduced that helps the straightforward design of room acoustics, optimized for the most important parameters.

In our future works we have to examine this approach in other applications and verify their validity for larger halls as well. Also, the effect of introducing more degrees of freedom in the form of detailed description of reflection and source-receiver should be investigated. Having a more accurate measurement system, a new measurement technique could be also developed in the future.

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