Evaluation and Design of Small Rooms

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Summary
Small rooms in the acoustical sense, are the bottlenecks of classical room acoustics. There are practical guidelines for the design and evaluation of such rooms, but still no widely accepted objective numerical criteria and general design method are at hand. This paper presents a uniform evaluation and design approach, based on conventional measurement and computer aided modelling methods of impulse responses or energy decay curves.

1. Introduction

In classical room acoustics several studies discuss the acoustical properties of large rooms, concert halls and theatres. For the objective qualification there are also well adopted objective parameters and measurement methods for such rooms. However, in most of the cases, one has to deal with rooms that are rather small in the acoustical sense. It means, that the linear sizes of obstacles inside and of the room are commensurable, or of the same order of magnitude, with the wavelength at the frequencies in question.

For the practising acoustician it is hard to find useful, numerically expressed and widely accepted results and methods to base onto, especially if the room is to be heavily treated with special materials and surfaces, and the effect of furniture and equipment is not negligible.

In room acoustics there are four aspects one can speak about: subjective impression, design, prediction and measurement. These four elements are directly connected to each other with processes like qualification, verification and optimisation or the aim of the design. These are all summarised in the "design circle of room acoustics", shown in Figure 1.

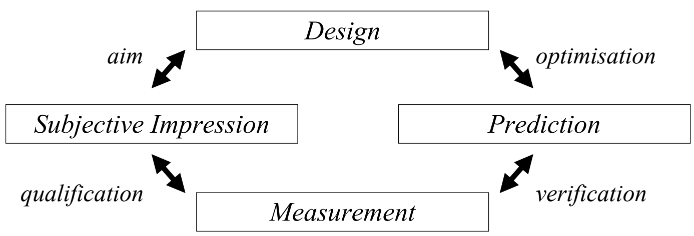

Figure 1. The design circle of room acoustics

In our investigations the goal was to find a set of scaleable objective parameters, that describe the subjective impression well. Having these parameters, their meanings were examined from the point of view of modelling, so that the design process can be more straightforward and automated in optimisation for the given numerical criteria.

2. Overview of the traditional approach

The conventional method of designing room acoustics is mostly a trial and error process, where different configurations of sound sources, positions and surface types are considered for a given room geometry. For the comparison, objective parameters can be predicted using different methods ranging from manual calculations to the measurement of scale models. Even if this approach is fairly well established, there are still questions to answer, like “what are the objective parameters to design for?”, “what are the numerical criteria for the parameters?”, or “how do parameters of prediction and of materials relate?”, and the like.

Particularly if the room is small, only conventions and first hand experience are helping this optimisation process. In order to see, how prediction results and objective parameters are related, an overview is given of the conventional approach in this part of the paper.

2.1. Qualification of rooms

Although the perception of sound in rooms has been of interest to acousticians for many decades, the interaction between the acoustical space and the listener is still not well understood. There are some objective parameters to characterise the acoustical properties of the sound field. The goal is to find those, which correspond to the subjective ones.

The reverberation time of a room used to be the predominant indicator of its acoustical properties. Whilst reverberation time continues to be a significant parameter, it is generally believed that more specific acoustical parameters are needed for a more complete evaluation of the perceptual quality of rooms.

As a result of extensive research work and subjective tests, several objective parameters have been derived in concert hall acoustics from impulse responses, like early or late energy ratio, definition or lateral energy fraction [1]. These are adequate for characterisation of large rooms but it is apparent, that with the same settings they are not directly applicable for small rooms. Therefore, no numerical criteria are proposed generally, either.

2.2. Methods and errors of prediction

The study of sound energy propagation within rooms has a long history. The beginning of modern scientific approach dates back to the end of the nineteenth century with the theoretical work of Lord Rayleigh and the practical methods of Sabine. Sabine assumed in his simple model of reverberation an ideally diffuse field in large, fairly reverberant rooms [2]. From that time reverberation time became one of the most important criterion for designing a room acoustically. It is obvious, since this phenomenon is perceived and measured easily.

Because of the complexity of analytic solutions, many of the related works have centred on the treatment of the statistics of hypothetical, randomly distributed sound energy [3]. It has long been recognised that this simplification is invalid when the size of the enclosure is not much larger than the wavelength.

In practice, sound energy emitted by a source does not travel far, before its path is obstructed by some solid object. In small rooms this interaction may happen within one or two milliseconds. What happens then is still very complicated. Because of this very complicated behaviour of the energy propagation in small rooms, a complete analytic approach of the description of the sound field, even in a simple real room, is not feasible.

In order to predict the most important sound field parameters, several modelling methods are developed. Methods like finite element (FEM), boundary element (BEM) or finite differences (FDM) give numerical solutions of the linear wave equations by introducing discretisation of space, surface or time, respectively. These are accurate, but are limited by the computational efforts to the relative low frequencies only. Other methods model wave propagation using the simplifications of geometrical acoustics [5]. These include for example the mirror image source (MISM) and the ray-tracing (RTM), beam-tracing (BTM) or particle-tracing methods, but these are valid at relatively high frequencies or are rather statistical because of the simplifications [4].

To cover the whole audible frequency range, numerical and geometrical methods are combined sometimes [6]. In the remaining part this paper we consider to use the beam-tracing method for prediction, because this method is one of the most popular and efficient ones in designing room acoustics.

A common problem of any prediction method is, that in addition to the limitations of the method itself, the results will be subject to uncertainty caused by the imprecision of input data. A first order approximation of the total prediction error may be expressed by [7]:

$$\textit{Total Error} = S \cdot \Delta H + \Delta S \cdot H \tag{1}$$

where S is the vector of input data with error ΔS and H is the matrix of transfer functions with error ΔH, comprising all geometrical information but excluding any other parameters, assumed to be part of the vector S. In a beam-tracing process, apart from the errors induced by the simplifications of the modelling method, there are two main groups of parameter errors. These are the directional data of sources and receivers, and the data describing the reflecting properties of surfaces, structures and materials. These data have generally limited resolution in frequency or space and are usually estimated from table lookups, therefore they do not represent real features accurately enough.

Such a lack of accurate data prompts the question of how far it is worth pursuing any method of prediction, and with these limitations in mind, how valid are the results in a trial and error process with estimated modelling parameters? Unfortunately we cannot discuss the parameter-sensitivity problem of prediction errors analytically, because it largely depends on the geometry and the type of the algorithm itself.

Discussions on the validity of approximations in a beam-tracing algorithm are beyond the scope of this paper. However, our experiences show, that parameter errors are usually greater than the errors of the method itself ($\Delta S \cdot H > \Delta H \cdot S$). Hence we assume the validity of a classical beam-tracing procedure below and focus our attention on the effects of parameter errors only.

3. Towards a better approach

To improve the design process, in the first step new objective parameters are needed, that are proved to correlate with the subjective impression well, and are easy to measure and predict. Finally, the parameters should be scaleable with global parameters of rooms (for example volume). Easy prediction of these parameters means, that prediction methods should result these parameters efficiently, such that a direct link is achieved between prediction and qualification.

3.1. Developing new objective parameters

Subjective studies have shown that several objective quantities, obtained from measured impulse responses, correlate well with particular subjective aspects of the acoustical character of an auditorium [8], and that ratios of received energy in the response may correlate with perceived quality. Hence, besides reverberation times, the most commonly investigated property of an impulse response is the balance of received energy before and after a given time point, called the early time limit (t_e) [9].

One can calculate the ratio of early-to-late energies from the impulse response, where the value t_e is chosen accordingly. This is called the early-to-late index and is denoted by C_{t_e}:

$$C_{t_e} = 10 \cdot \log_{10} \frac{E_0^{t_e}}{E_{t_e}^{\infty}} = 10 \cdot \log_{10} \frac{\int_0^{t_e} p^2(\tau) d\tau}{\int_{t_e}^{\infty} p^2(\tau) d\tau} [dB] \tag{2}$$

where $p(\tau)$ is the pressure at the receiver point (microphone). A fairly common usage of this parameter for large halls or rooms is called "clarity" (C_{80}), where t_e=80ms.

It is also possible to measure the early-to-total sound energy ratio, denoted by D_{te}:

$$D_{t_e} = \frac{E_0^{t_e}}{E_0^{\infty}} = \frac{\int_0^{t_e} p^2(\tau)d\tau}{\int_0^{\infty} p^2(\tau)d\tau}[\%]. \tag{3}$$

For qualifying speech conditions, “Definition” or “Deutlichkeit” is an accepted measure, where t_e=50ms (D_{50}), but again, this parameter is developed for large halls. Since C_{te} and D_{te} can be expressed from each other, it is not necessary to measure both quantities:

$$D_{t_e} = \frac{E_0^{t_e}}{E_0^{\infty}} = \frac{E_0^{t_e}}{E_0^{t_e} + E_{t_e}^{\infty}} = \frac{10^{C_{t_e}/10}}{10^{C_{t_e}/10} + 1}. \tag{4}$$

Using t_e as a parameter, these quantities may be used for smaller rooms as well. An example is the M-factor, where C_{te} is used with two time constants, t_{e1}=5ms and t_{e2}=20ms:

$$M = C_{20} - C_5 \, [dB] \tag{5}$$

A different approach to describe the temporal distribution of received energy is to calculate the centre time, denoted by t_S, which is the time of the centre of gravity of the squared impulse response:

$$t_S = \frac{\int_0^{\infty} \tau \cdot p^2(\tau)d\tau}{\int_0^{\infty} p^2(\tau)d\tau}[s] \tag{6}$$

These objective parameters are proposed in several publications even using band limited filtering to describe frequency dependent changes, but their subjective meaning and value range of interest is still under investigation. For our experiments, we used these parameters as starting points and tried to use different values for the early time limit.

3.2. The energy decay curve (EDC)

As we have mentioned earlier, the temporal behaviour of received energy plays an important role in shaping the subjective quality. Most of the above parameters can be expressed in a more general form by the energy decay curves:

$$EDC(t) = 10 \cdot \log_{10}\left[1 - \frac{\int_0^{t} p^2(\tau)d\tau}{\int_0^{\infty} p^2(\tau)d\tau}\right][dB]. \tag{7}$$

Instead of the ∞, usually the constant of 1s is chosen for calculating the total energy, because of the noise problems during the measurement.

Some features of the EDCs are advantageous. For example, EDCs are normalised, so they are level independent in a linear approach. Also, they are integrated in time which makes the comparison and matching of different responses easier. In addition, most of the objective measures are in direct connection with the EDC.

The relationship with reverberation and decay times is evident. For the theoretical reverberation time (RT_{60}) we get:

$$EDC(RT_{60}) = -60dB \tag{8}$$

In practice, reverberation time ($RT_{5\text{-}35}$) is usually derived from assuming dB-linear decay between the -5dB and -35dB points, or the early decay time (EDT_{10}) is measured from the -10dB decay time, but the relationship is simple again.

More interesting is the relationship with the values of early-to-total and early-to-late energy ratios:

$$EDC(t_e) = 10 \cdot \log_{10}\left(1 - D_{t_e}\right) = -10 \cdot \log_{10}\left(10^{C_{t_e}/10} + 1\right) \tag{9}$$

or with the M-factor:

$$EDC(5ms) - EDC(20ms) = 10 \cdot \log_{10}\left(\frac{1 + 10^{C_{20}/10}}{1 + 10^{C_5/10}}\right) = 10 \cdot \log_{10}\left(\frac{1 + 10^{(C_5 + M)/10}}{1 + 10^{C_5/10}}\right). \tag{10}$$

These relations determine several points or their relative distance on the EDC, these are illustrated in Figure 2.

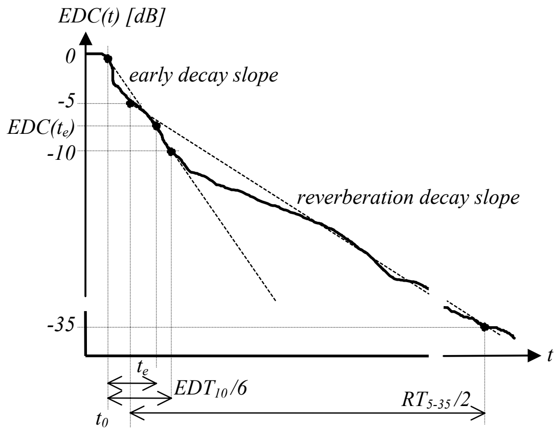

Figure 2. Relationship of objective parameters and the EDC

4. Subjective tests

After selecting the objective parameters, one has to determine how their parameter values correspond to real subjective qualities, and how well they correlate. These results can be derived from several subjective tests only, even for larger halls.

To find the relationships between objective and subjective parameters, several subjective tests were performed in concert halls worldwide. Comparing the objective and subjective results showed, that some of the objective parameters relate to perceived clarity, the balance between clarity and reverberance, and timbre. Based on these experiences, similar relations were examined in small rooms, too.

4.1. Subjective tests

Control rooms of broadcast and recording studios are acoustically controlled environments and are the object of the most critical acoustical design at the same time. This is why our subjective tests were carried out in reference listening rooms and control rooms of the Hungarian Radio.

Subjects of the tests were experts and students [10]. After an ear-training of the subjects, a 40 minutes long demo record was selected from different sources. In the first step this record was played back in the reference listening room of the Hungarian Radio to have a reference impression. Further tests were performed in other rooms, and after listening the tests records, the subjects filled out a questionnaire with questions like

- stereo accuracy,
- timbre,
- spatial impression,
- transparency,
- frequency response,
- room modes,
- other resonances,
- noise from outside,

- noise from equipment,
- main impression, and
- comfort impression.

The subjects were asked to qualify these impressions on a 5-point scale.

4.2. Evaluation of the subjective tests

In the same rooms where the subjective tests took place, acoustical measurements were carried out using the MLSSA analyser of DRA Laboratories.

Parameters specified by the EBU recommendations on measurements are describing [11]:

- direct sound,
- early reflections,
- reverberation field,
- operational room response curve, and
- background noise.

Beside the recommended parameters [11] [12] [13], at every listening position in every room, values for C_{15} ,C_{20}, D_{15}, D_{20} , t_S, and M were calculated from the measured impulse responses. The latter parameters were added, because in the case of smaller rooms, due to their smaller early time limits, they were assumed to have a more direct connection to the subjective side than the recommended ones. A digest of our results for the calculated correlation are summarised in Table 1. Here, the parameters C_{15} and D_{15} were left out, since they did not show higher correlation in our cases.

	D_{20}	C_{20}	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

Table 1. Calculated correlation between subjective and objective parameters of the tests

As one can see, there is some clear correlation. However, these results should be handled with care, because only several other independent subjective tests could prove if these relations are definitive. Nevertheless, the evaluation of the results proved again, that characterisation of the sound field in control rooms with the previously recommended parameters is not exact enough. Looking for new objective parameters is still reasonable, and the ones formulated in our paper may be adequate or at least promising for these purposes.

5. A proposed design approach

As it was pointed out in the earlier part of this paper, energy decay curves can be used to compare different responses. Doing this comparison with predicted or measured responses can lead to useful results.

5.1. Theory

5.1.1. EDC fitting

Modelling methods based on the laws of geometrical acoustics, give their results in the form of echograms, which are the temporal distribution of detected reflections at the given frequency (Figure 3). Time smearing of reflections is usually neglected in these methods, so it would be hard to match broad-band real impulse responses with monochromatic echograms directly, even after a band limiting filtering. By means of integration, energy-decay curves are a more appropriate choice for doing this.

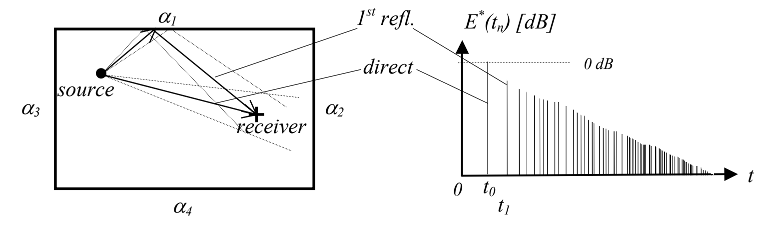

Figure 3. Echogram and its meaning

The idea behind the inverse procedure, called "EDC fitting" is, that from a given decay curve, based on the timing of modelled detection, one can calculate the amplitudes of each echo in the echogram. The result is an echogram with the same timing as the modelled one and with the same energy decaying properties as the required one. A scheme of this process is shown in Figure 4.

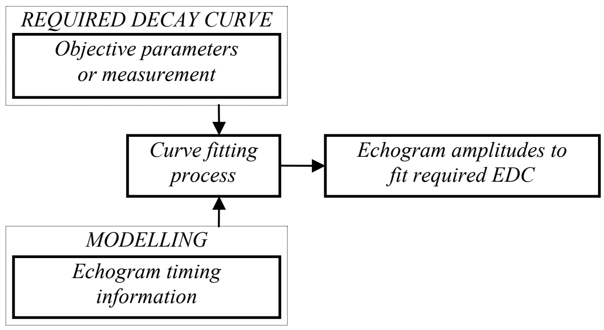

Figure 4. The EDC fitting process

For a more detailed explanation of this process, let us denote the timing of the n-th reflection by t_n (n=0,1,2,…N). Then, the modelled energy decay curve at a given frequency is:

$$EDC_f^*(t_n) = 10 \cdot \log_{10}\left[1 - \frac{\sum_{i=0}^{n} p^{*2}_f(t_i)}{\sum_{i=0}^{N} p^{*2}_f(t_i)}\right], \tag{11}$$

where N is the number of modelled reflections and f is the frequency where our monochromatic model was run at. The asterisk denotes the predicted values here. The normalisation with the total energy can be neglected, since our model is linear, so only the attenuation of the received reflections relative to the direct sound are important:

$$EDC_f^*(t_n) = 10 \cdot \log_{10}\left[1 - \sum_{i=1}^{n} \overline{p}^{*2}_f(t_i)\right] = 10 \cdot \log_{10}\left[1 - \sum_{i=1}^{n} \overline{E}^{*}_f(t_i)\right]. \tag{12}$$

For the best possible match, the modelled curve should be the sampled version of the required one:

$$EDC_f^*(t_n) = EDC_f(t_n), \tag{13}$$

where $EDC_f(t_n)$ is the required curve, sampled with the timings of reflections. This means, that after this sampling, the modelled EDC looks only stepped, but has the same decaying properties (Figure 5).

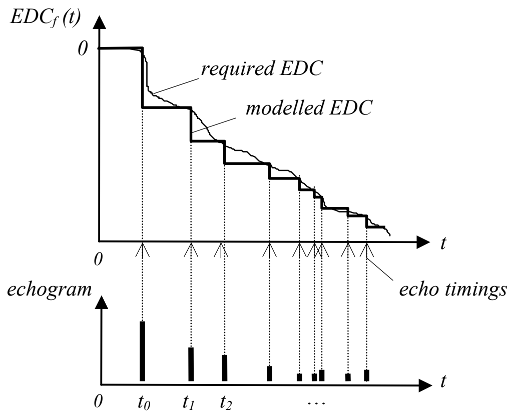

Figure 5. Sampled and original EDC in the EDC fitting process

The amplitude of the direct sound in the echogram plays an important role in this process, so we need an extra parameter which tells the balance of the direct sound and the total sound energy. This can be expressed by the aforementioned early-to-total energy ratio (or here, the "direct-to-total ratio"):

$$D_{t_0} = \frac{E(t_0)}{E_0^\infty} \quad [\%] \tag{14}$$

where t_0 is the timing and $E(t_0)$ is the energy of the direct sound in the echogram. Note, that in a real decay curve the slope around the direct sound is not as abrupt, because the direct sound is smeared in time and the real response is band limited. Parameters, like EDT_{10} or C_5 are meant to describe the role of the direct-to-reverberant behaviour of the room, since the direct sound arrives in the very first 2ms…6ms part of the response.

Using the value of direct-to-total ratio and a numerical differentiation, the normalised energy of the direct sound, and the subsequent reflections is:

$$\begin{aligned} &\overline{E}_f^*(t_0) = D_{t_0}, \\ &\overline{E}_f^*(t_n) = \sum_{i=0}^{n} \overline{E}_f^*(t_i) - \sum_{i=0}^{n-1} \overline{E}_f^*(t_i) = \left[1 - 10^{EDC_f^*(t_n)/10}\right] - \left[1 - 10^{EDC_f^*(t_{n-1})/10}\right], \quad n = 1,2,\ldots N. \end{aligned} \tag{15}$$

This recursive calculation is relatively simple to do, and depending on the original EDC to fit to, several aspects of the model can be derived.

5.1.2. Calculation of modelling parameters

Using the results from an EDC fitting process, one can calculate the most important modelling parameters that are contained in the echograms. In order to discuss a simplified calculation process in detail, we introduce a number of restrictions first:

- sources and receivers are supposed to be omnidirectional and point-like;
- diffuse reflections are neglected because they make EDCs just smoother, but they do not affect their decay slope much (although they are subjectively preferable);
- we are interested only in the absorption coefficients, non-specular phenomena are neglected;
- geometry and positions are supposed to be exact.

Note, that even if curves fit with minimal error, there is always a difference. The fitted curve looks more stepped, because non-specular phenomena are neglected in our model. It is obvious, that by involving other phenomena e.g. diffusion, angle-dependent absorption, etc., their parameters give more degrees of freedom in this process, so other parameters could be calculated also, and greater accuracy can be achieved.

If we assume the validity of the simplified algorithm, the receiver point detects the sum of reflections, where each reflection path has its own response function:

$$H_{rec}(f) = H_{src}(f) \cdot \sum_{i=1}^{\infty} H_{path,i}(f) \cdot \frac{e^{-j\omega t_i}}{c(f)t_i} \tag{16}$$

where H_{rec} is the resulting response function at the receiver position, H_{src} is the response function of the source, t_i is the time (length) of the reflection path, c is the propagation speed and $H_{path,i}$ is the response function of the i-th path:

$$H_{path,i}(f) = H_{air,i}(f) \cdot \prod_{k=1}^{K} H_{surf,i,k}(f) \tag{17}$$

where $H_{air,i}$ is for the air absorption, finally $H_{surf,i,k}$ is the response function of the k-th reflecting surface on the i-th reflection path.

After correcting the amplitudes of the reflections due to air absorption, we may write the following equation system:

$$\prod_{k=1}^{K} H_{surf,1,k}(f) = H_{path,1}(f)$$
$$\vdots \tag{18}$$
$$\prod_{k=1}^{K} H_{surf,N,k}(f) = H_{path,N}(f)$$

Considering the simplifications, a surface only attenuates incident energy, and the transfer function of a path is a product of simple attenuations:

$$(1-\alpha_1)^{M_{1,1}} \cdot (1-\alpha_2)^{M_{1,2}} \ldots (1-\alpha_K)^{M_{1,K}} = H^2_{path,1}(f)$$
$$\vdots \tag{19}$$
$$(1-\alpha_1)^{M_{N,1}} \cdot (1-\alpha_2)^{M_{N,2}} \ldots (1-\alpha_K)^{M_{N,K}} = H^2_{path,N}(f)$$

where K is the number of surface types, α_k is the absorption coefficient of the k-th surface. Since a surface may take part in a reflection path more than once (or not at all), this multiplicity is expressed by the prime $M_{i,k}$, which can have a value of 0, 1, …, etc.

After taking the logarithm of Eq. (17), we get a simple linear equation system, where only the absorption coefficients are unknown.

$$\underline{\underline{M}} \cdot \log_{10}(1-\underline{\alpha}) = \log_{10} \underline{H}^2_{path}(f) \tag{20}$$

where $\underline{\underline{M}}$ is the "multiplicity" matrix. A sample of a typical $\underline{\underline{M}}$ matrix is shown in Figure 6.

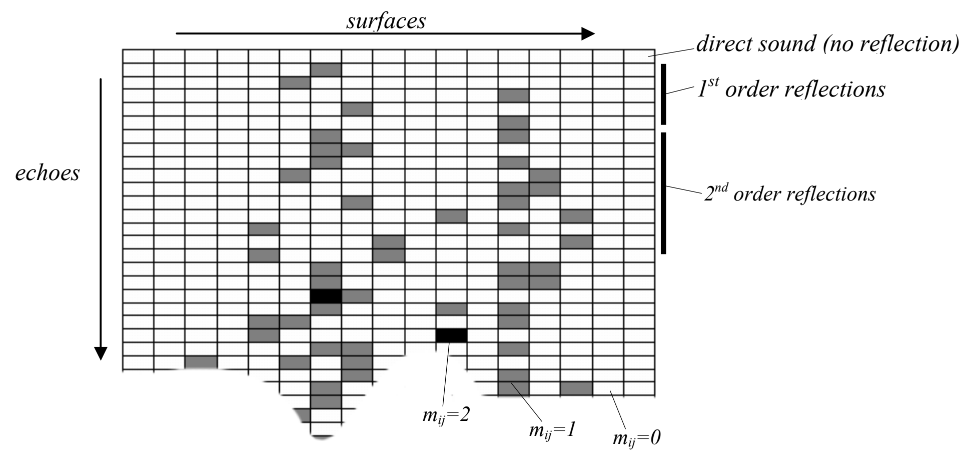

Figure 6. An example for the M-matrix

The problem one is facing with is, that from an ordinary echogram one can have detections in the order of 10000, but there are usually about just some 10-50 surface types. In other words, the equation system is rather overdetermined. Obviously, the very first equations are the most important ones. Therefore an iterative procedure is suitable to approximate the parameters, where the amplitude of the correction vector is decaying in order to express this precedence:

$$\underline{\alpha}_{k+1} = \underline{\alpha}_k + \gamma_k \cdot \underline{\delta}_k^T \tag{21}$$

where γ_k is the decaying term causing the attenuation of the correction vector of error $\underline{\delta}_k$.

5.1.3. Optimisation of modelling parameters

Our initial goal was to find a value set of modelling parameters, that results in the required objective parameter values. The idea is to use an "EDC template", where only those points of the EDC are prescribed, which relate to the objective parameters. The points (i.e. slope) of the EDC between these points are assumed to be unimportant. The differences between the determining points of the EDC describe the ratio of the received energy between the time points and the total energy remaining:

$$EDC(t) - EDC(t + \Delta t) = 10 \cdot \log_{10}\left(\frac{E_t^{t+\Delta t}}{E_{t+\Delta t}^{\infty}} + 1\right) . \qquad (22)$$

Since the inherence of modelling parameters and modelled results is too complex, instead of an analytical approach, a stochastic optimisation is proposed. The error function is the sum of the differences between the modelled and the required values at the determining points of the curves in dB:

$$e = \sum_{l=1}^{L} \left[EDC(t_l) - EDC^*(t_l)\right]^2 \cdot w_l \ , \qquad (23)$$

where e is the total error of the approximation, L is the number of determining points at times t_l, and w_l is the weighting of the errors respectively.

This may be a more practical approach, because the acoustician can decide, which objective parameters are, and which are not so important to design for. In addition, the optimisation process usually results in several different solutions that fulfill the requirements equally. One may even use different weighting functions for different receiver points in the room, in order to differentiate between different seating positions of the audience in a larger hall, for example.

5.2. Application

In this part of the paper examples are presented for using the EDC fitting process, the calculation and the optimisation procedure respectively.

5.2.1. EDC fitting

There are two ways of using the EDC fitting process, depending on the origin of the curve to fit:

- if the curve is from a measured and band-limited response, the results of the fitting can be used for validation and measurement, or
- if the curve is synthesised from some objective parameters, the results of the fitting can be used for design (one may construct a synthesised EDC assuming exponential decay between determining points of the curve for example).

In both cases, after correcting for the attenuation of air absorption and propagation the resulting echogram can indicate if there is something wrong with the model or the requirements. Normally the direct sound has the greatest amplitude, since all echoes are much more attenuated due to the propagation and absorption on reflections. Exceptions include for example the extreme directional character of the source or the receiver, and the focusing features of reflecting surfaces.

Excluding these possibilities, echoes with higher amplitudes indicate the invalidity of the modelling method (or the simplifications). Similarly, aberrations of this kind may indicate, that the required objective criteria in the synthesised EDC cannot be fulfilled using the given geometry or source-receiver position and directional characteristics. Note that in practice, these errors are important in the calculations only in the early part of the response, because of the attenuation of propagation.

In the examples in Figure 7 two different requirements are prescribed for the same configuration using a synthesised EDC. Objective parameters in both examples are: D_{0ms}=50%, RT_{60}=0.2s. In the first example, C_{20ms}=3dB was specified. It is clear, that using this geometry and source-receiver configuration, the requirement can be fulfilled only, if the directional characteristics of the source or the receiver is not omnidirectional. Using omnidirectional sources and receivers, this configuration can produce about C_{20ms}=4.55dB with special reflectors. A more reasonable requirement specifies C_{20ms}=8dB…12dB, to have a more uniform energy distribution.

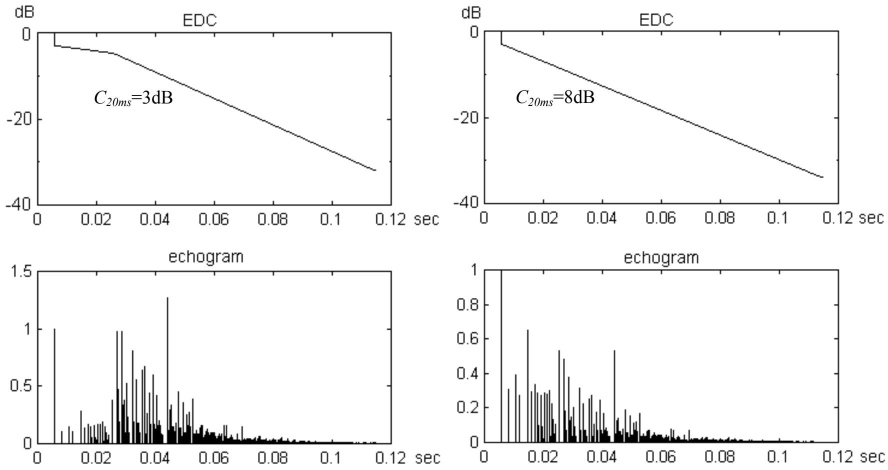

Figure 7. Examples for the discussion of EDC fitting

5.2.2. Calculation of modelling parameters

Having the amplitudes of the echoes from an EDC fitting, one can calculate the parameters of the model. As it was mentioned above, if the EDC to fit was synthesised from some objective parameters, the process can be used to design room acoustics. However, in our next example, the original EDC is a measured one.

The original goal was to examine the validity of a simplified beam-tracing algorithm, and to test the inverse method. For the experiments a well documented reference listening room in the Hungarian Radio was chosen having sound sources with fairly smooth and even directional characteristics and with very good transient response.

For the calculations, measured impulse responses were band limited to the standard frequencies from 31.25Hz up to 16kHz by using phase-linear FIR filters, and stored as raw floating point data with the sampling rate of 75kHz. From the data using the procedure outlined in section 5.1.2., the absorption coefficients of different surfaces were calculated.

The geometry of the room, an example for the calculated and measured EDC at 62.5Hz, and a digest of absorption coefficients of different materials is shown in Figure 8.

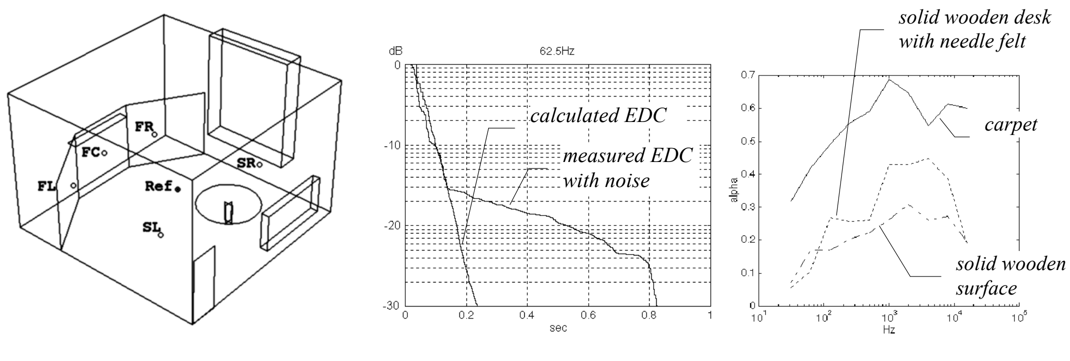

Figure 8. Geometry and results for the discussion of the calculation procedure

As one can see, the two curves do not fit in the late part. This is due to the noise during the measurement. The absorption coefficients are fairly reasonable, even at 31.25Hz.

Note, that there are several problems, when calculating parameters after fitting for a measured EDC:

- timing of the direct sound is crucial, but it is hard to find a "direct" sound in a band limited response at around 125Hz, for example;
- great error is induced by the inaccuracy of directional behaviour of sources and receivers.

A common problem is, that the selection of the decaying factor in the iterative procedure has a great effect on the accuracy of fitting.

5.2.3. Optimisation of modelling parameters

In contrast to the deterministic approach in the previous section, here we search the optimum solution by using a conventional stochastic algorithm, and can compare different results that already fulfil the requirements.

For our example let us consider a room having a simple shoebox shape. The dimensions are 5×6×3m. The source and the receiver are omnidirectional. The side walls are divided vertically into 3 pieces. The EDC for the calculations was synthesised assuming exponential decay between the points, that were determined by the objective parameters C_{20ms}=5dB, D_{0ms}=50% and RT_{60}=0.4sec. Two additional points were selected on the curve at 40ms and 80ms. For the stochastic search, 5000 echoes were used.

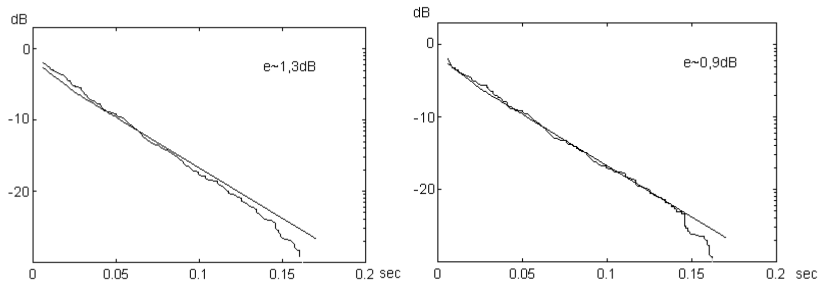

Figure 9. Different solutions approximating a synthesised curve (see Table 2)

Figure 9 shows how the synthesised and the predicted curves fit after the optimisation of absorption coefficients. Three sets of results are summarised in Table 2. On the explanation of calculating the error, see equation (23).

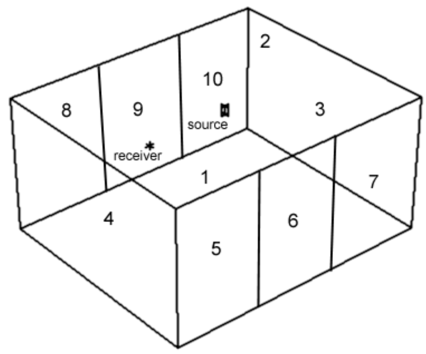

Figure 10. Geometry and surface numbers of the room, table with detailed results and errors

surface	α_I	α_{II}	α_{III}
1	0.64	0.96	0.1
2	0.54	0.81	0
3	0.04	0	0.78
4	0.09	0	0.8
5	0.63	0.53	0.46
6	0.34	0.24	0.62
7	0	0	0.16
8	0.4	0.39	0.66
9	0.6	0.16	0.9
10	0.18	0.1	0.02
error	1.3dB	0.9dB	0.7dB
RT_{Sabine}	0.292s	0.238s	0.336s

Table 2. Detailed results and errors

It is interesting to see, that totally different configurations of absorbers and reflectors can comply to the same specifications. The classical reverberation time is calculated for the different versions in the table, for comparison purposes.

6. Conclusions, further work

Different aspects, like prediction, measurement and subjective impression in room acoustics should be discussed together to examine their relationship.

In this paper the starting point was to use objective parameters showing good correlation with perceived quality in our tests. With the energy decay curves as general tools, prediction, measurement and subjective impression can be compared and analysed, based on those parameters. Inverse methods in geometrical acoustics are a novel topic, but they seem to be beneficial and expedient in the validation and qualification of different modelling algorithms, measurement of acoustical parameters in situ and, last but not least, the straightforward design of acoustics of both large and small rooms.

However, the correlation of objective and subjective features must be studied in more detail. Especially the question of dependence of the required numerical criteria on the global properties of the room calls for several independent subjective tests.

The inverse approach is suitable for larger rooms as well, but a common problem with tracing methods is, that spatial accuracy deteriorates with time, because the solid angle between adjacent rays or beams is constant. The model and the examples were rather simple in this paper, therefore extensive modelling and measurement experiments are needed for validation in more complex situations. For the very same reason, these methods should be tested with more degrees of freedom in the model, and the possibility of other application areas should also be investigated.

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