INVESTIGATION OF LISTENING CONDITIONS FOR MULTICHANNEL SOUND SYSTEMS

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In sound recording and broadcasting to test the different multichannel formats by listening tests calls for investigations of listening conditions. Well controlled sound is important and critical in reference listening rooms and recording rooms. In order to find new objective parameters for room acoustical evaluation of these spaces, several subjective tests were carried out. New objective parameters derived from measured impulse response are compared with the results of the subjective investigations. An additional investigation had to be done when pictures are accompanying multichannel audio programmes. In order to predict the most important sound field parameters accurately enough, a series of modelling has been performed. Reference listening rooms are small, special materials and structures are applied in them and the effect of furniture and equipment is not negligible. Since traditional finite element techniques cannot be used effectively in these situations, the validity of the geometrical approach is verified. The goal of our investigations is to find a high and medium frequency modelling process that predicts the acoustical parameters of the sound field more precisely.

INTRODUCTION

The introduction of multichannel audio technology into production and reproduction process in sound recording and broadcasting needs careful study. The study should cover besides the originating, recording, postproduction, transmission and reproduction process the listening conditions also.

The definition "listening condition" means the acoustical properties of the sound field of the room which influences the listener at the listening position. Although several international standards and recommendations [1], [2], [3], [4] give methods for conducting listening tests and some of them [1], [2] give recommendations for listening conditions, more research is required to determine the exact relationship between the sound field properties and the sound quality of reproduction in a room.

Recommendations define parameters like direct sound, early reflections, reverberant field and operational room response curve and they are used and accepted factors to qualify listening conditions. Nevertheless, experiences show that new parameters are needed.

Predicting these parameters is also an important task. Therefore a modelling method is needed also, which is able to calculate these features reliably.

In the first part of this paper the subjective test and their result are presented, and new objective parameters are proposed which correlate with the subjective side. In the second part the modelling process, a method to reduce the effect of the main error sources, finally the modelling results are discussed and described.

1. DEALING WITH SMALL ROOMS

Listening and control rooms are usually small rooms in the acoustic sense. In classical concert hall acoustics several objective parameters are known determined from impulse response like clarity, definition or lateral energy fraction. These are adequate for characterisation of large rooms but they are not directly applicable for small rooms.

The form of physical laws of sound propagation does not depend on room size. However, there are several important difference in methods of investigations used for small or for large rooms. Basically the propagation of sound in small enclosures is more strongly influenced by wave effects than it is in large rooms. In small spaces the sound energy does not travel as far before being reflected from the room surfaces than in large rooms.

2. OBJECTIVE PARAMETERS

There are different methods to find new parameters. The easiest one seems to be calculating parameters from room impulse response [5], [6]. Other proposed methods calculate parameters from binaural impulse responses [7].

In our work we determined new parameters based on the energy time domain integrals, and these are the following:

2.1 Parameters k_1 and k_2

These parameters are similar to the clarity used for concert hall characterisation. In small rooms the t_1 time-interval must be shorter.

$$k_1(t) = \log_{10} \frac{\int_0^{t_1} p^2(t)dt}{\int_0^{\infty} p^2(t)dt} [dB] \tag{1}$$

$$k_2(t) = \log_{10} \frac{\int_0^{t_1} p^2(t)dt}{\int_{t_1}^{\infty} p^2(t)dt} [dB] \tag{2}$$

2.2 Centre time: t_S

This parameter gives the middle time, that is "the centre of gravity" of the energy distribution.

$$t_S = \frac{\int_0^{\infty} t \cdot p^2(t)dt}{\int_0^{\infty} p^2(t)dt} [ms] \tag{3}$$

2.3 M-factor

The M-factor gives information about the ratio between the direct sound and the early part of the arriving energy.

$$M = k_2(20ms) - k_2(5ms) \, [dB] \tag{4}$$

3. SUBJECTIVE TESTS

To find relationships between objective and subjective parameters several subjective tests were performed in different rooms. Since most of the selected rooms were listening rooms for two-channel, two-channel stereo demo records were used in the work.

Subjects of the tests were experts and students. After an ear-training, a 40 minutes long demo record was selected. 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 test 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.

4. OBJECTIVE RESULTS

In the same rooms where the subjective test took place, acoustical measurements were carried out using the MLSSA analyser of DRA Laboratories. From the measured impulse responses k_1, k_2, t_S and M-factor were calculated. The k_1, k_2 were calculated with t_l=20msec. The results are given in the Table 1.

	St1	St6	St8	St22	St23	St24
k_1 L (dB)	-0.23	-0.80	-0.47	-0.29	-0.26	-0.17
k_1 R (dB)	-0.25	-0.61	-0.56	-0.31	-0.28	-0.28
k_2 L (dB)	12.60	6.94	9.39	11.63	12.09	14.03
k_2 R (dB)	12.20	8.24	8.58	11.25	11.75	11.72
t_S L (ms)	10.46	18.93	17.32	12.09	10.59	9.10
t_S R (ms)	10.53	16.69	20.90	12.98	11.40	10.17
M L (dB)	0.51	0.30	0.21	0.31	0.12	0.13
M R (dB)	0.49	0.38	0.14	0.30	0.22	0.11

Table 1: Objective parameters

In the reference listening room the parameters were calculated for the all five channels:

	FL	FR	FC	SL	SR
k_1 (dB)	-0.17	-0.28	-0.39	-0.44	-0.26
k_2 (dB)	14.03	11.77	11.21	20.83	19.12
t_S (ms)	9.1	10.17	12.83	4.07	5.26
M (dB)	0.13	0.11	0.24	0.34	0.33

Table 2: Objective parameter for the 5 channels

5. COMPARISON

To find the relations, the correlation between the different parameters were calculated. Table 2 shows some of the calculated correlation values.

From the correlation results the following conclusion can substracted:

- t_S centre time is correlates good with *transparency* and *spatial impression,*
- M-factor relates to the *timbre*, finally
- k_1, k_2 show connection to *stereo accuracy*.

	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

Table 3: Correlation of subjective and objective parameters

6. MODELLING OF ACOUSTICS IN GENERAL

In practice, properties of the sound field in rooms are generally modelled using algorithms based on the laws of geometrical room acoustics. These are proved to be reliable enough where the modelled room is not small in the acoustic sense, i.e. the linear sizes of obstacles found in the room are much larger than the wavelength at the frequencies in question [8].

However, there are cases where this condition is not fulfilled, so other methods, for example finite element modelling must be used. Unfortunately while designing room acoustics, because of the computationally intensive algorithms, the use of numerical methods is worth in special cases or at extreme low frequencies only.

Therefore the geometrical method was chosen, and this way we could study the effect of the equipment and furniture on the subjective impression of the sound field also. Nevertheless, the validity of geometrical acoustics is limited at lower frequencies, so in order to have a reliable yet efficient modelling method, we carried out a number of measurements and modelling, finally compared the results. For the comparison we chose the parameters that we found to have the best match with the subjective results.

7. MODELLING PROCESS

In geometrical room acoustics the simple idealisation of a sound ray is used rather than the concept of a wave. The sound ray is a small portion of the wave front originating from the sound source, propagates in straight lines with the velocity of sound and reflects optically [9]. From these postulates different algorithms have been developed.

7.1 Modelling Method

We used the well-known and simple triangle beam tracing method, where rays represent an exact triangular shaped piece of the wave front so that receiving points can have an exact intersection with the beam surface (Fig. 1), thus the timings of detected reflections are correct.

For the modelling we used the RAYNOISE simulation software package from LMS. Edge diffraction was not considered, absorption coefficients were taken from lookup-tables.

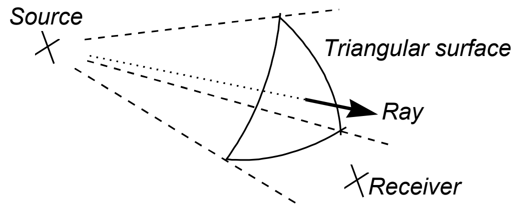

Figure 1: Triangular beam tracing.

7.2 Calculating the results

The results of the modelling are basically echograms at given frequencies, including information about the amplitude, phase and timing of each reflection. From these echograms one can calculate energy decay curves easily, which are the basis of the most important parameters, for example reverb time, clarity, etc. The proposed parameters in the first part of this paper are also based on energy-time integral ratios.

7.3 Comparing Measured and Predicted Results

As expected, the predicted value of statistical parameters like reverberation time, etc. matched fairly good, and even the furniture or the equipment did not influence the results (Fig. 2).

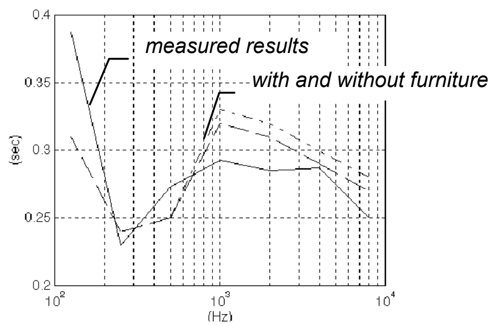

Figure 2: Global reverberation times (RT_{60}).

However, examining different acoustical quantities and their change around the listening positions showed clear difference between the room with and without furniture, but the agreement between the measured and modelled results was not so trivial (Fig.3).

To inspect the causes of deviations, the possible error sources had to be investigated.

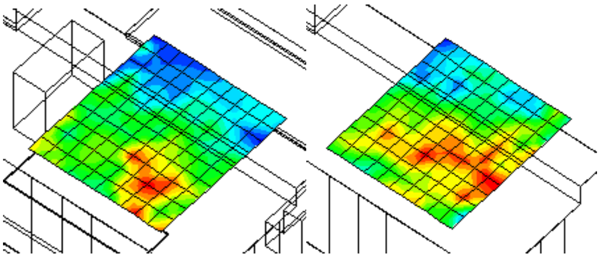

Figure 3: Distribution of central gravity time around the listening point, with and without furniture at 1000Hz

8. CHECKING THE VALIDITY

8.1 Error sources of modelling

Our modelling experiences show, that errors during a modelling process are mostly due to the incomplete knowledge about the data describing the properties of materials. These include absorption coefficients, diffusion factors and their directionality.

Other error sources are the imprecise geometry description and/or source and receiver positions, their directional data, and last but not least, the limitations of the modelling method itself, of course.

In other words, if we assume that our geometrical room acoustics modelling algorithm is valid so that only the parameters are incomplete and incorrect, we have to study the effect of parameter errors on the modelling results.

Unfortunately we cannot discuss this parameter-sensitivity problem analitically because it largely depends on the geometry and the type of the algorithm itself. For this reason in the following part of this paper we assume the validity of the modelling process and study the parameter values only by comparing measured and modelled results.

8.2 Parameter calculation

If the modelling method is valid and parameters of the model are imprecise, we should calculate parameter values for the model in order to gain modelling results matching the measured ones.

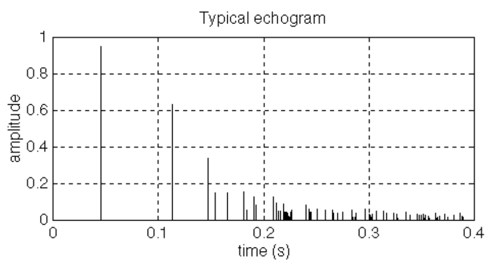

Figure 4: A typical echogram (e.g. at 1kHz)

As stated above, direct results from the modelling process are typically echograms showing the temporal distribution and the amplitudes of reflections from reflecting and diffusing surfaces around the receiver position at a given frequency (Fig. 4).

Echograms also contain information about the history of a given reflection: what was the path of the reflection, which surfaces took part in the reflections, and so on.

The direct results from a measurement are usually wide-band impulse responses (Fig. 5).

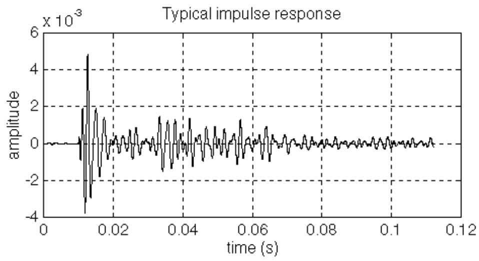

Figure 5: A typical room impulse response

As we can see, it is rather difficult if not impossible to fit monochromatic echograms to wide-band impulse responses directly in order to calculate the required parameters. Even if we calculate wide-band impulse responses from the echograms based on the assumption that the linear system of the room is causal, in practice the resulting ‘continuous’ time-domain impulse responses differ too much, mainly because of the lack of exact simulation of the phase relations.

Therefore the fitting process should be based on energy-time functions. Note, that this is reasonable also, since the reverberation quantities and most of the acoustic quality factors (Clarity, Definition, etc.) and also the proposed ones (k_1 and k_2, M) are based on energy-time integrals and ratios.

8.3 EDC fitting

For the fitting process the energy decay curve (EDC) is chosen:

$$EDC(t) = 1 - \frac{\int_0^t p^2(t)dt}{\int_0^\infty p^2(t)dt} \tag{5}$$

where *p(t)* is the pressure at the receiver point (microphone). Instead of the ∞ usually the constant of 1s is chosen because of the noise problems during the measurement.

The fitting is based on the information of the time intervals of incoming reflections (modelled echogram) and the energy content ratios of the successive reflections (measured EDC).

The result is an echogram with the same timing as the modeled one and with the same energy decaying properties as the measured one (Fig. 5).

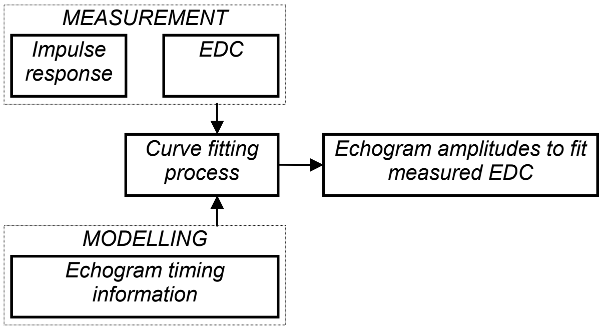

Figure 6: EDC fitting

In order to discuss the process we introduce a number of simplifications:

- sources and receivers are supposed to be omnidirectional;
- diffuse reflections are neglected because they make the 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
- geometry and positions are supposed to be exact.

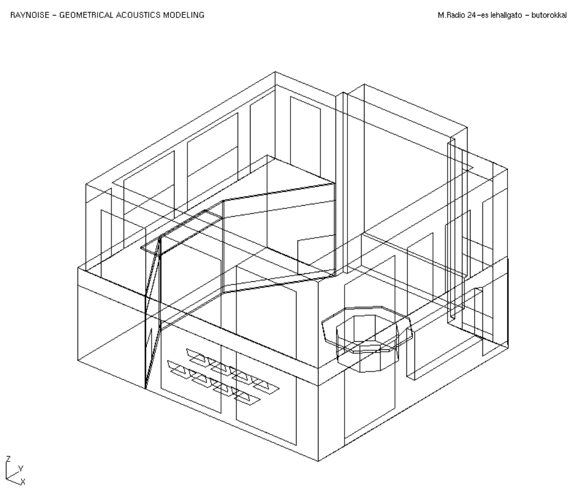

Figure 7: Geometry of the room

For the experiments a well documented reference listening room in the Hungarian Radio was chosen (Fig. 7) having sound sources with fairly smooth and even directional characteristics and with very good transient response.

8.4 Calculating parameters from the echogram

Generally spoken, if we assume the validity of the algorithm, with a point-like source and a receiver, where directivity is not taken into account, the receiver point detects the sum of reflections, where each reflection path has it own response function:

$$H_{rec}(\omega) = H_{src}(\omega) \cdot \sum_{i=1}^{\infty} H_{path,i}(\omega) \cdot \frac{e^{-j\omega t_i}}{c(\omega) t_i} \tag{6}$$

where H_{rec} is the resulting response function at the receiver position, H_{src} is the response function of the source, ti 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:

$$H_{path,i}(\omega) = H_{air,i}(\omega) \cdot \prod_{n=1}^{N} H_{surf,i,n}(\omega) \tag{7}$$

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

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

$$\begin{gathered} \prod_{n=1}^{N} H_{surf,1,n}(\omega) = H_{path,1}(\omega) \\ \vdots \\ \prod_{n=1}^{N} H_{surf,K,n}(\omega) = H_{path,K}(\omega) \end{gathered} \tag{8}$$

For a given frequency in terms of energies:

$$\begin{gathered} (1-\alpha_1)^{M_{1,1}} \cdot (1-\alpha_2)^{M_{1,2}} \ldots (1-\alpha_N)^{M_{1,N}} = E_1 \\ \vdots \\ (1-\alpha_1)^{M_{K,1}} \cdot (1-\alpha_2)^{M_{K,2}} \ldots (1-\alpha_N)^{M_{K,N}} = E_K \end{gathered} \tag{9}$$

where N is the number of surface types, α_n is the absorption coefficient of the n-th surface, E_i is the calculated and corrected energy of the i-th reflection of the echogram. Since a surface may take part in a reflection path more than once (or not at all), this multiplicity is expressed by $M_{i,n}$, which can be 0, 1, …, etc.

After taking the logarithm of Eq. (9), we get a simple linear equation system. A part of the coefficient matrix is shown ‘from above’ in Fig. 8.

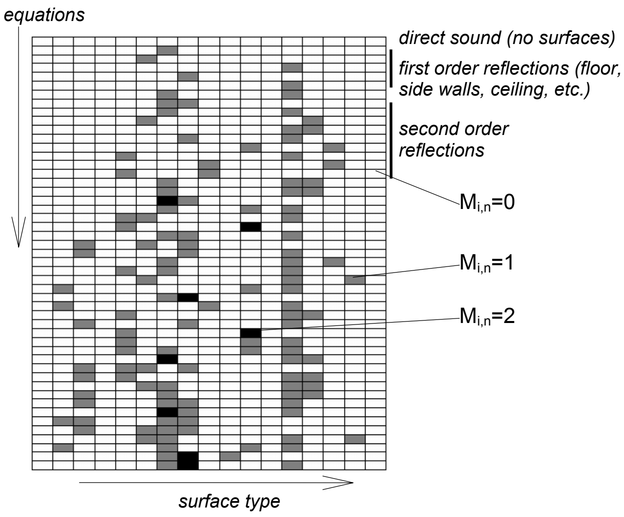

Figure 8: Part of the coefficient matrix

This equation system is strongly overdetermined, thus a special iterative procedure was created (Eq. 10).

$$\overline{\alpha}_{k+1} = \overline{\alpha}_k + \gamma \cdot \overline{\delta}_k \tag{10}$$

where the correction vector decays exponentially due to the coefficient γ, according to the following considerations:

- the first reflections are the most important,
- the decay slope of the ETC is important,
- the later part of the measured ETC is not as important because of noise problems during the measurement.

The performance of the calculation can be improved by increasing the degree of freedom in the model, i.e. by involving diffuse or directional reflection, directional characteristics of the source or the receiver, and so on.

8.5 Results

Measured impulse responses were band limited to the standard center frequencies from 31.25Hz up to 16kHz and stored as raw floating point data with the sampling rate of 75kHz. From these data using the procedure outlined above, the absorption coefficients of different surfaces were calculated.
Figure 9 shows some measured and predicted EDCs using the new absorption coefficients at low frequencies.
Some of the calculated absorption coefficients are shown in figure 10 with frequency.

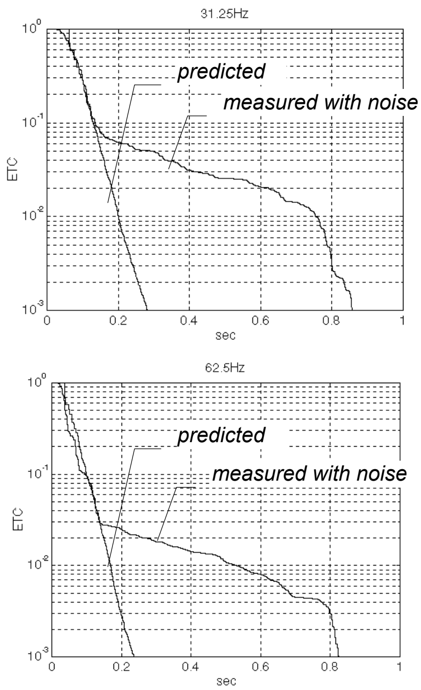

Figure 9: Measured and predicted EDCs with calculated parameters

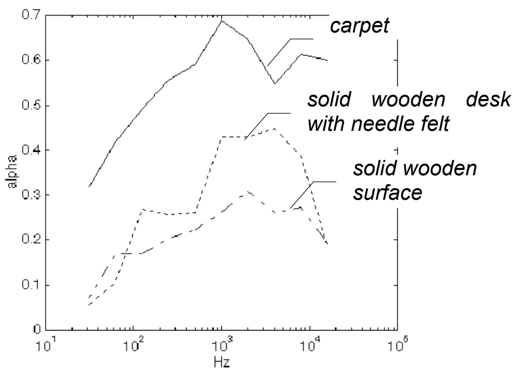

Figure 10: Calculated absorption coefficients

In figure 10 the solid line shows the absorption coefficients of surface type 7 (Fig. 8), which was the floor with a wall-to-wall carpet on it. The dashed lines below show absorption coefficients of solid wooden surfaces, where the upper one is a desk coated with needle felt.

9. CONCLUSIONS

To predict the quality of the sound field in a room, new objective parameters are needed especially in the multichannel production and reproduction process, where the well controlled sound is even more important and critical. Because listening and control rooms are usually small, the investigation and modelling process are focusing on small rooms.

In this paper the proposed objective parameters show promising correlation between the objective and the subjective side when listening to two-channel stereo recordings. At the same time, objective parameters relating the subjective impression in a multichannel listening environment can be developed based on the proposed approach.

Although the methods of geometrical room acoustics are only coarse simulations of what really takes place in the sound field of a room, our examinations showed they can be used even in the extreme situation of acoustically treated small rooms.

The accuracy of these algorithms may be significantly improved just by tuning the different parameters. To match the measured EDCs, a method for estimating the absorption coefficients was presented.

Results, calculated from measurements and predictions at different excitation and receiver points in the same room correlated well, which could be important in a room for multichannel reproduction. As one would expect, differences occured mainly at surfaces far from the receiver point and at higher frequencies. These errors could be explained here with the very fast decay rate and the directional properties of the sources.

Based on the idea of utilising the temporal distribution of reflections to determine their amplitudes and that predicted and measured energy-time curves can be matched almost directly, several other new 'inverse' method could be developed and proved to be reliable in the future.

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