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A NEW APPROACH TO THE ROOM IMPULSE RESPONSE SIMULATION

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The most important problem of room acoustics is the evaluation of the acoustic quality of projected and modernized rooms. To this end it is necessary to work out a calculating impulse response algorithm for a digital room model. The virtual rooms impulse response manners of modeling are described in the article. The impulse response synthetic models have been evaluated on the basis of selected numerical factors and a simple auditory test. Thanks to the impulse response calculation it is possible to evaluate the subjective and objective room acoustic properties and hence to run the room simulation tests (the sound sources configuration, the room planning) in order to reach optimal solutions, which meet a given criterion. The problems described are related to the "virtual sound space" concept. Intuitively, we mean by this a sound reality model, which enables, on the basis of the model and measurement data, an audio impressions simulation of the existing and being under projection acoustic systems. A basic process, which enables the sound reception, is the test signal and the impulse response convolution.

1. Introduction

The problem of room acoustic properties is very controversial. Generally, there are two approaches to the problem:

- Objective evaluation – based on the numerical factors,
- Subjective Evaluation – based on the audio monitoring "quality" comparison.

An acoustic pressure level or a reverberation time determination are examples of the objective evaluation. A subjective evaluation is, most often, a detailed description of the esthetic impressions related to audio monitoring within the given room. A room acoustic properties evaluation based on the single objective parameters is not sufficient, particularly referring to lecture halls, theaters and concert halls, and taking more factors into consideration causes the necessity of determination their scale coefficients [3, 1].

As the result, the evaluation process is not unambiguous and the scale coeffcients criterion entering causes that the process becomes subjective. It seems that the processes of evaluation of the room acoustic properties, the objective and subjective ones, are based on separated assumption sets.

The proposal to evaluate the room acoustic quality on the basis of the room impulse response seems to be unique. The room impulse response provides the determination of some objective factors: C_{80} (*Clarity*), D_{50} (*Definition*), EDT (*Early Decay Time*), RT_{60} (*Reverberation Time*), and STI (*Speech Transmission Index*). There is a possibility to use the room impulse response for a subjective evaluation on the basis of the audio tests for a given site of the room.

The RT_{60} **reverberation time** is a base parameter which is determined on the basis of the room impulse response. Until now, the most common method of the reverberation time determination has been the decay curve slope reading method.

The *method of integrated impulse response* is much more convenient. The method was introduced and applied for the first time by SHROEDER [23]. The method is based on the following relationship between $\langle h^2(t)\rangle$, the mean for all possible decay curves (for a given place and noise signal band), and the appropriate impulse response:

$$\langle h^2(t)\rangle = \int_t^{\infty} [g(x)]^2 dx = \int_0^{\infty} [g(x)]^2 dx - \int_0^{t} [g(x)]^2 dx. \tag{1}$$

The above equation (1) means that the impulse response second power integration is equivalent to all decay curves averaging for the curves for a given room point, when the noise signal is induced. It is clear that the integrated impulse response method is the most effective one because it is free of any accidental interference.

The EDT **early decay** time is used more frequently than the reverberation time. The EDT is defined as the time of an energy level falling from 0 dB down to -10 dB. The factor is very useful when the room acoustic properties are described subjectively.

The Thiele criterion [9], so-called D_{50} **definition**, is one of the earliest attempts to define an objective room acoustic properties factor. The definition is defined as follows:

$$D = \frac{\int_0^{50\text{ms}} [g(t)]^2 dt}{\int_0^{\infty} [g(t)]^2 dt} 100\%. \tag{2}$$

The lower integration limit $t = 0$ coincides with the reaching of a direct sound observation point. Bore [13] found a relationship between the D definition and the "syllable comprehensibility". He ran some subjective testes within various rooms with and without speaker systems. A tone test pulse with a duration time of 20 milliseconds was applied to read the impulse response. The D "definition" was calculated by means of a mean computing for the $340 - 3500$ Hz signal band. The results show that there is a "good" relationship between the D definition and the "syllable comprehensibility".

The C_{80} **clarity** resembles formally the definition. It is used for defining the music reception clarity in the concert halls. Reichardt [13] has defined the C_{80} clarity as follows:

$$C = 10\log_{10}\left\{\frac{\int\limits_{0}^{80\mathrm{ms}}[g(t)]^2 dt}{\int\limits_{80}^{\infty}[g(t)]^2 dt}\right\} \quad \mathrm{dB}. \tag{3}$$

In the Eq. (3), there is a 80 milliseconds delay time. The time is greater than the appropriate one in Eq. (2) (50 milliseconds) because in the case of music reception it is harder to detect a reflection in comparison with a speech signal. The preferred C value for various kinds of orchestral music was determined by means of subjective tests. It has been found that for $C = 0$ dB, even fast musical phrases sound clearly under subjective evaluation, and $C = -3$ dB is still acceptable.

The STI **speech transmission index** and $RASTI$ **rapid speech transmission index** are used for the evaluation of the lecture halls speech comprehensibility. The factors are calculated on the basis of the MTF modulation transfer function, which is determined by means of the impulse response [24, 11]. The greater the STI, the better audio reception conditions.

It is easy to evaluate a room acoustic quality on the basis of the factors, which are calculated by means of the impulse response.

2. The linear system impulse response

The room impulse response is a kind of a "short cut". In fact, an impulse response is related to two points: the source and the receiver. Therefore, within a given room, there are infinitely impulse responses depending on the positions of the two points.

In fact, an *impulse response of a transmission track* between the source and the receiver is a more correct term.

The linear system properties are included in its impulse response or, adequately, in its transfer function. The function is a Fourier transform of the impulse response. A room considered as an acoustic system can be described as a linear system. Therefore its impulse response gives a complete description of the acoustic signal changes on its way between two points within the room.

If a sound transmission track is introduced as a linear system, as shown in Fig. 1, where $x(t)$ – an input signal as a function of time, $y(t)$ – output signal as a function of time, $h(t)$ – linear system impulse response, $X(j\omega)$ – input signal spectral form, $Y(j\omega)$ – output signal spectral form, $H(j\omega)$ – linear system transfer function.

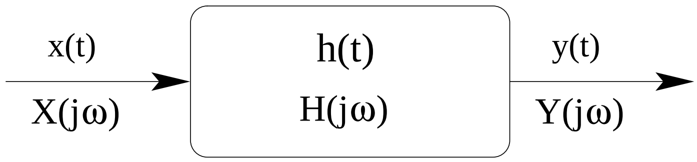


Fig. 1. Linear system model.

The relationship between the input and output signals is as follows:

$$Y(j\omega) = H(j\omega)X(j\omega). \tag{4}$$

If we operate the transfer function by means of a reverse Fourier transform, it follows:

$$h(\tau) = \mathcal{F}^{-1}\left[H(j\omega)\right] = \frac{1}{2\pi}\int\limits_{-\infty}^{\infty} H(j\omega)e^{j\omega\tau}d\omega \tag{5}$$

the $h(\tau)$ function is the so-called system impulse response; the function is the system response at instant t for a δ function at the $(t-\tau)$ instant.

The fundamental term of the linear systems theory is a *convolution* (6) [2]:

$$y(t) = \int\limits_{-\infty}^{\infty} x(t-\tau)h(\tau)d\tau. \tag{6}$$

The linear system output signal is a two functions convolution: an input signal and an impulse response.

The discrete convolution [2] is defined as follows:

$$y_m = \frac{1}{N}\sum_{k=0}^{N-1} h_k x_{m-k}, \qquad m = 0, 1, \ldots, N-1. \tag{7}$$

The signal and the sound transmission track impulse response convolution adds all changes to the signal as it was emitted in the given source and received in the given audio reception point. On the basis of the above consideration, it is obvious that it is possible to create an acoustic field model which is related to hearing – the sense which the acoustic field is dedicated to.

3. The synthetic impulse response model

As mentioned above, the calculation of an impulse response is crucial to the room acoustic "climate" subjective evaluation. The impulse response of the existing real rooms can be measured. How to calculate the response for the rooms, which are being projected, and of course do not exist? It seems that the answer is obvious, when the linear system properties are taken into consideration: it is necessary to create a room sound transmission *model* on the basis of the existing acoustic field models [7].

Because the existing impulse response models do not meet the virtual room requirements related to a realistic sound [15], the manner of creating an impulse response model has been introduced. A *signal envelope* and a *fill* have been used in the model. A synthetic impulse response (Fig. 2c) is achieved if the envelope (Fig. 2b) is put onto the fill (Fig. 2a):

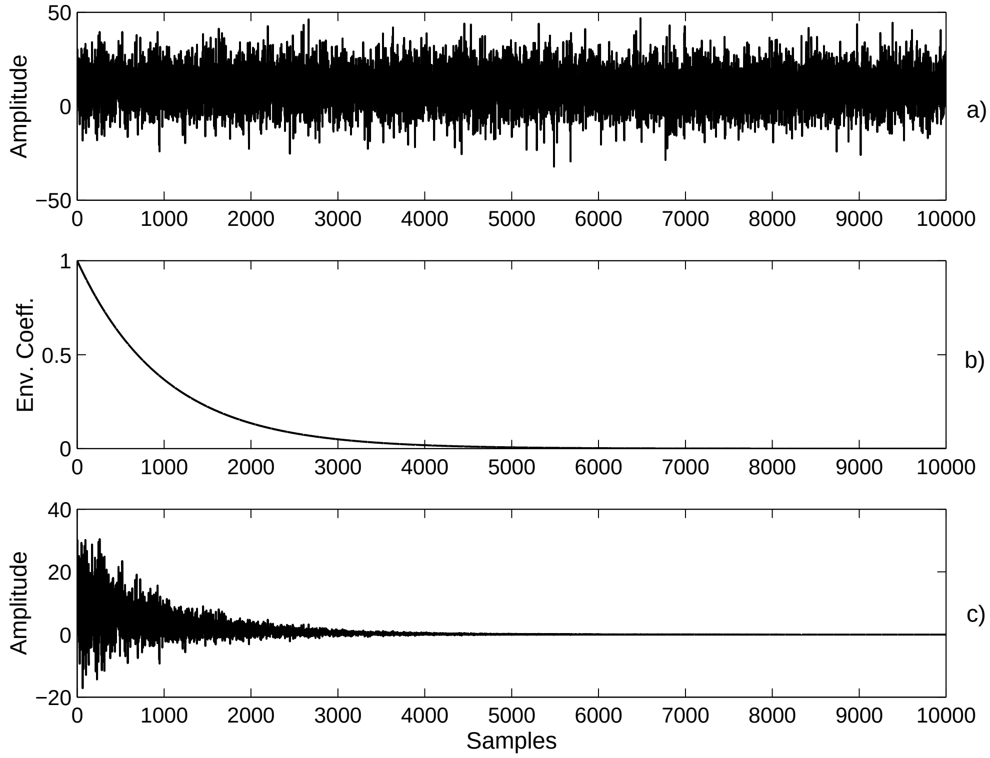


Fig. 2. The modeling of a synthetic impulse response.

In order to create a synthetic model of the virtual room impulse response it is necessary to model an acoustic field. An acoustic field geometrical model has been used because of its simplicity and the computation promptness. An implementation included in the *Raynoise* package of Numerical Integration Technologies has been used. The reflectograms are one of the resulting forms. The diagrams describe the time of reaching the observation point by an energy quantum against its energy, which is a function of the second power of the acoustic pressure (Fig. 3).

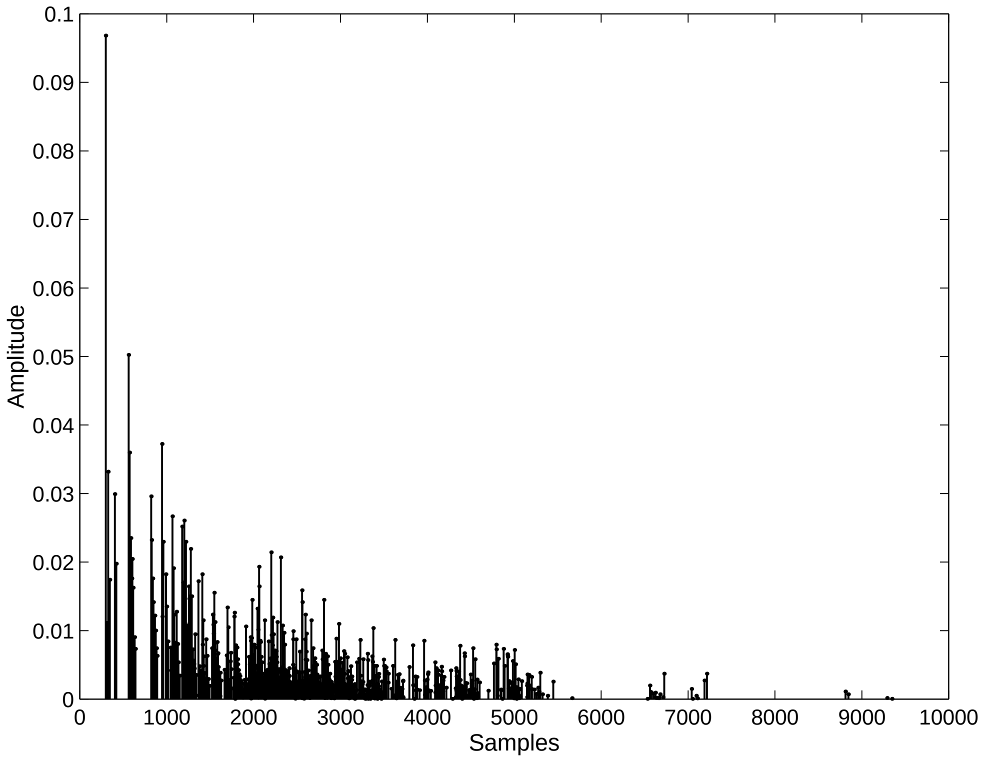


Fig. 3. A reflectogram (a one of the determined ones).

An reflectogram in its original form shows an energetic impulse response which is not suitable for using in the convolution operations. Anyway, it is possible to model an impulse response *envelope* on the basis of the reflectogram.

3.1. The impulse response envelope models

The form of an reflectogram is too discrete for using it in the envelope modeling. In order to “smooth out” the diagram an interpolation process can be applied. Of course, it is inadmissible to apply the process for the entire diagram. The initial diagram values are stripes which represent the direct sound wave and the first reflections; their averaging changes the room properties completely because the lines are responsible for the spatial audio impressions.

It is necessary to determine an interpolation initial point. It is suggested to replace the first echo diagram stripes in the impulse response model with the direct wave models (which are related to the source impulse response) and the reflected wave models [16, 17]. It is recommended to select an appropriate value of the dependence of the room’s properties and its dimensions.

The following interpolation methods have been tested:

1. Stripes interpolation by means of the rectangles.

2. Straight line interpolation.

3. Exponential averaging.

The interpolation has been operated by means of the IIR filter, which is described by the following recurrent equation:

$$X_{i+1} = \frac{n-1}{n} X_i + \frac{1}{n} X, \tag{8}$$

where X – averaged value, $i = 1, 2, \ldots, N$ – number of sample, n – quantity of samples used for the calculation of the next value.

4. The Savitzky–Golay filter approximation.

The least squares method is applied in the Savitzky–Golay smoothing filters. The filters are a generalization of the FIR filter, which assure a better preservation of the high frequencies at the cost of removing worse noises than the FIR filter [19].

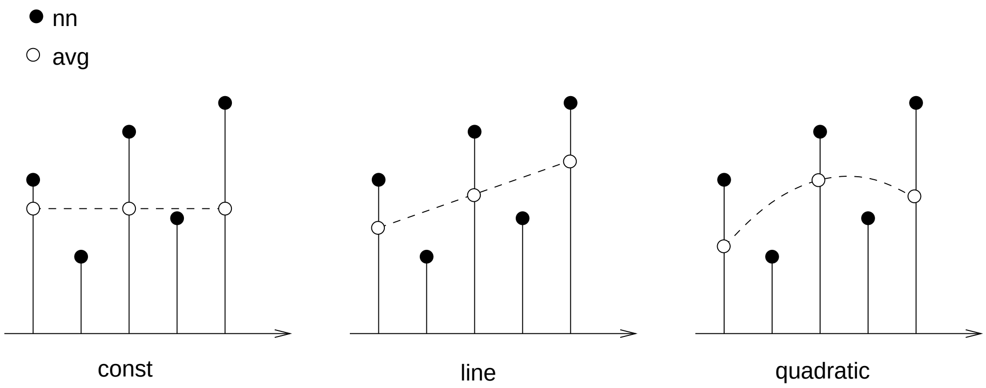


Fig. 4. Data approximation by means of the order $d = 0, 1, 2$ nn – the samples with noises, avg – approximated samples.

The data approximation by means of multinomials of various degrees is shown in Fig. 4. The appropriate smoothed values have been calculated by means of the 0, first and second order multinomial approximation for $m = -2, 1, 0, 1, 2$:

$$\begin{aligned} \hat{x}_m &= c_0 && \text{(constant)}, \\ \hat{x}_m &= c_0 + c_1 m && \text{(line function)}, \\ \hat{x}_m &= c_0 + c_1 m + c_2 m^2 && \text{(square function)}. \end{aligned}$$

The factors c_i for all the multinomial orders should be determined appropriately in order to fit best the data set. Applying the method of least squares can do this. The method enables the determination of the c_i factors with a minimal root-mean-square error. For example, there is an output factor minimization process applied for the square function:

$$\mathcal{J} = \sum_{m=-2}^{2} e_m^2 = \sum_{m=-2}^{2} (x_m - (c_0 + c_1 m + c_2 m^2))^2 = \min, \tag{9}$$

where the approximation error is determined as follows:

$$e_m = x_m - \hat{x}_m = x_m - (c_0 + c_1 m + c_2 m^2), \qquad m = -2, -1, 0, 1, 2.$$

Generally, the Savitzky–Golay filter of length N and order d, which smoothes the $x(n)$ sequences, can be introduced as follows (10):

$$y(n) = \sum_{m=-M}^{M} b_0(m) x(n+m) = \sum_{m=-M}^{M} b_0(-m) x(n-m). \tag{10}$$

The hypothetical impulse response envelopes calculated for $f = 1$ kHz by means of the described methods are shown in in Fig. 5–8.

Beside the interpolation methods described above, there is another manner of determining the reflectogram intermediate elements: it is possible to use an impulse response of a low-pass filter with parameters similar to the envelope spectral characteristic for a given frequency band.

The reflectograms show how the energy amounts, detected at given instants, decay with relation to the room dimensions. Beside the exponentially decreasing trend, there are in general some signal dynamic changes, especially in the initial phase of the model.

Because the impulse response energy decay for the given frequency bands are decreasing not monotonously exponential (Fig. 9), it is possible to determine the manner of the envelope modulation.

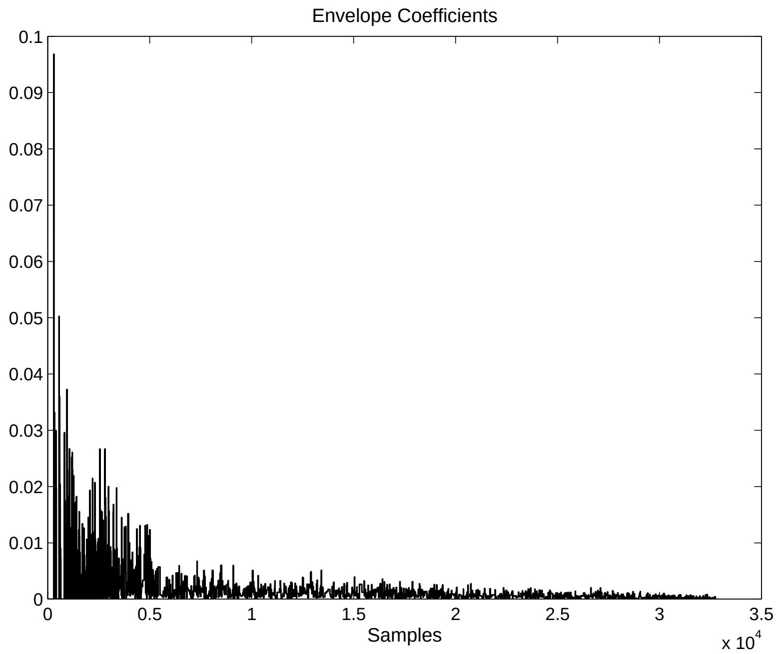
It is well known that such integrating filter smoothes rapid changes, which are characteristic of the echo diagrams. In this way their shapes remain similar to the time characteristics of the individual stages of the DWT (Discrete Wavelet Transform) obtained from the real measured impulse responses [18].

The envelope filter parameters database can be received as the results of the previous envelope time chart spectral analysis obtained after DWT of the real impulse responses from the objects with similar audio properties [15]. The stages of the envelope spectral parameters determination are as follows:

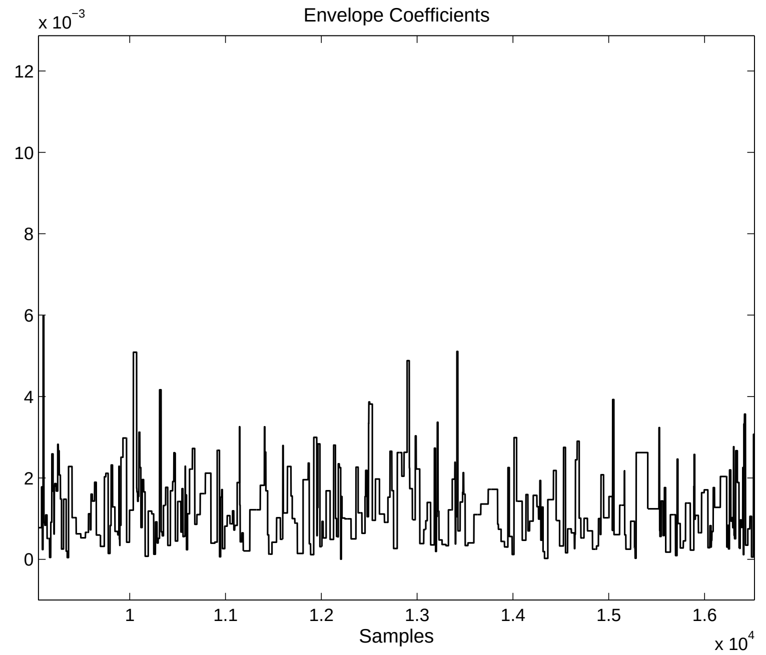
- response $h_R(\tau)$ time chart for B band:

$$h_{RB}(\tau) = \mathcal{F}^{-1}(H_B(j\omega) H_R(j\omega)), \tag{11}$$

where $H_R(j\omega) = \mathcal{F}(h(\tau))$, $\mathcal{F}$ – Fourier transform, $H_B(j\omega)$ – spectral characteristic of the B band-pass filter.

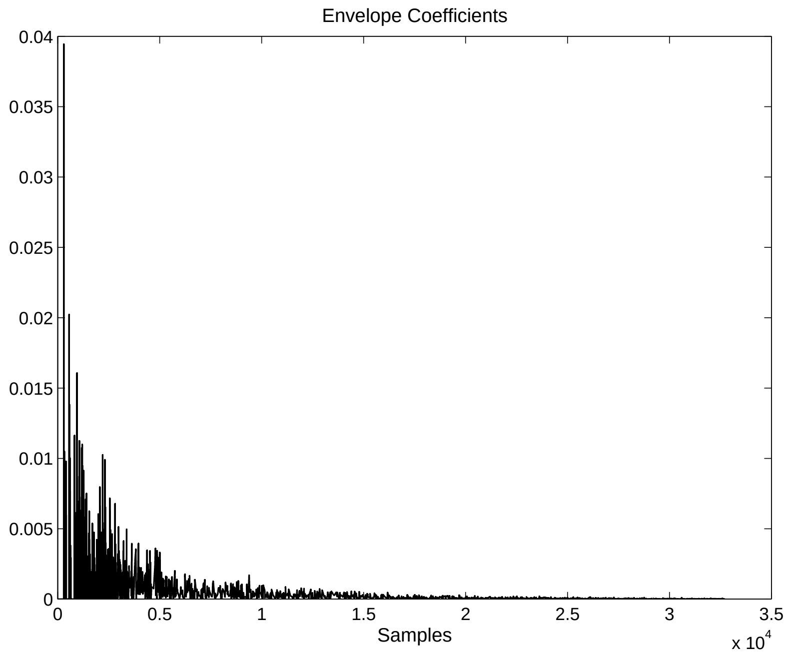


(a) The envelope.

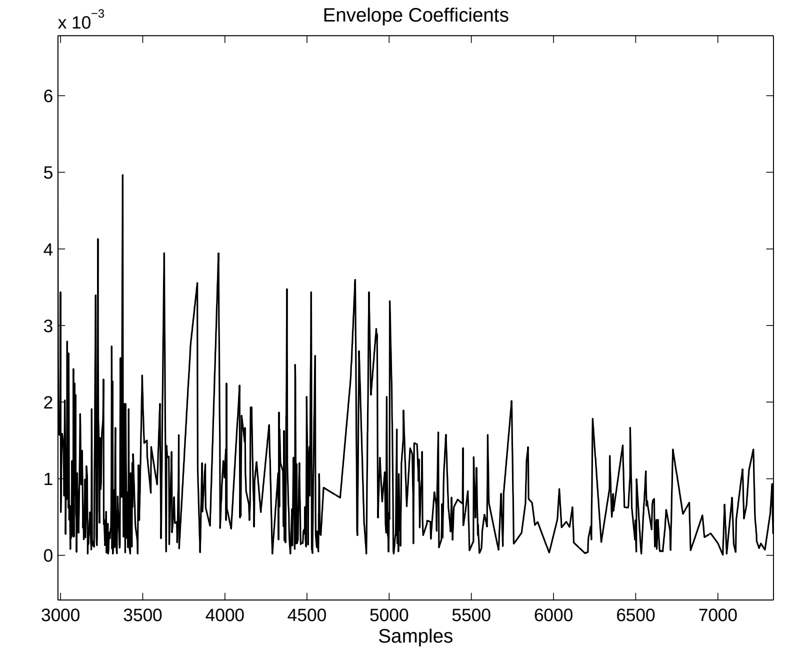


(b) The envelope magnified part.

Fig. 5. The envelope determined by means of an approximation with squares.

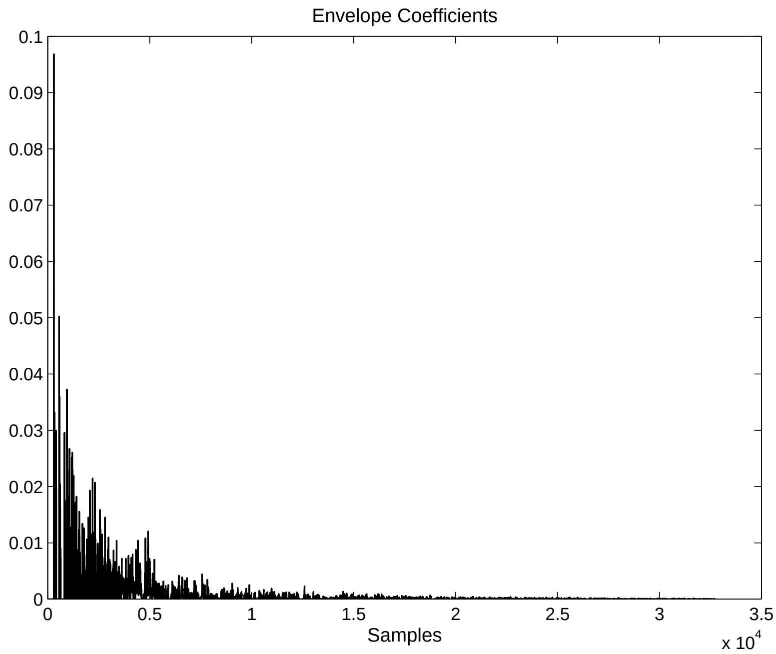


(a) The envelope.

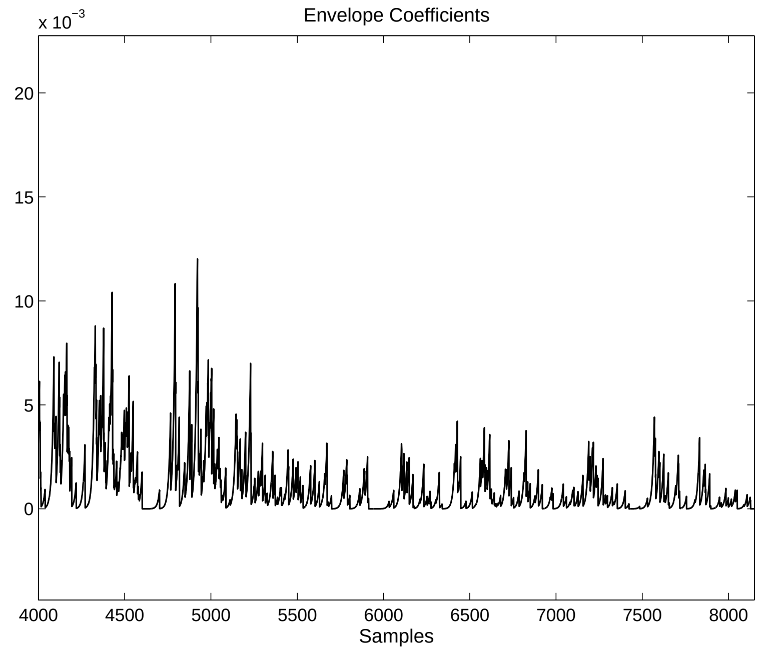


(b) The envelope magnified part.

Fig. 6. The line approximation envelope.

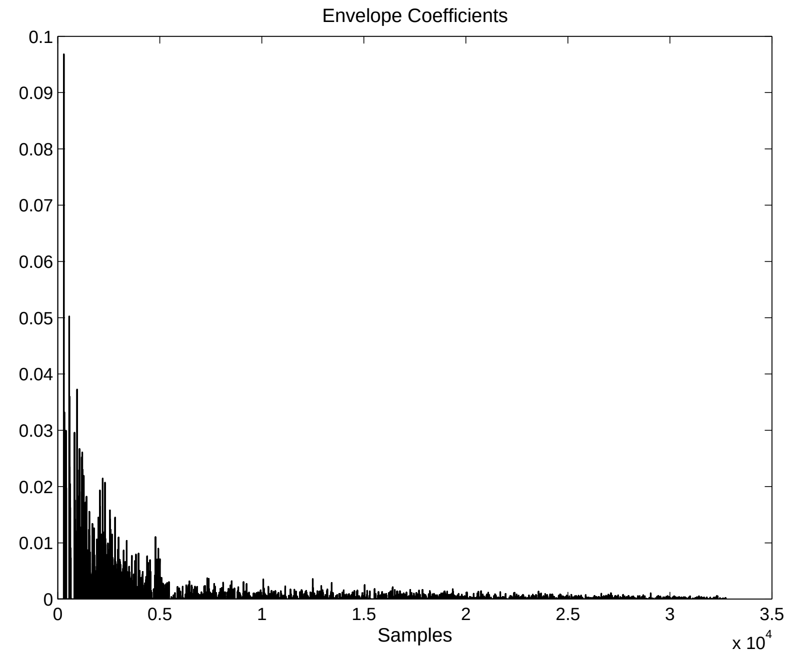


(a) The envelope.

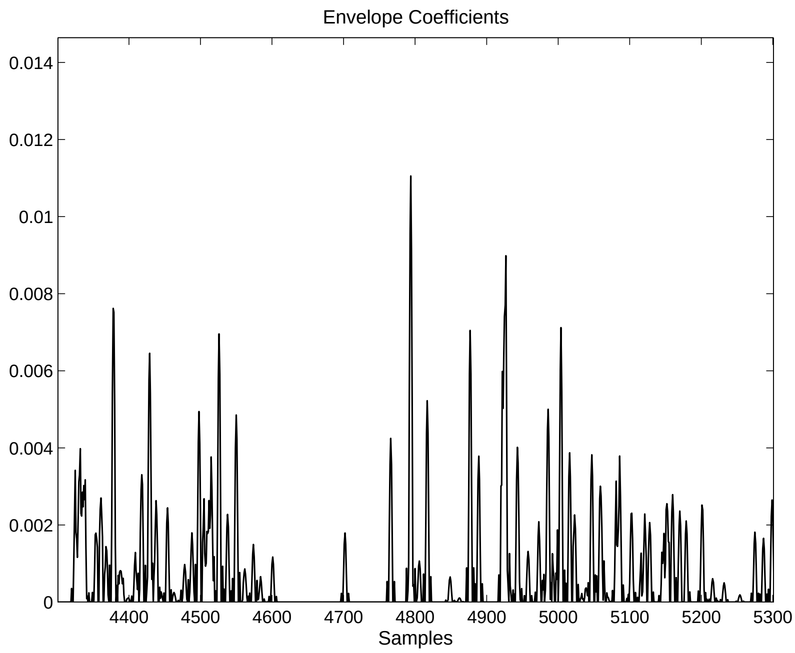


(b) The envelope magnified part.

Fig. 7. The exponential averaging envelope.



(a) The envelope.



(b) The envelope magnified part.

Fig. 8. The Savitzky–Golay filter approximation envelope.

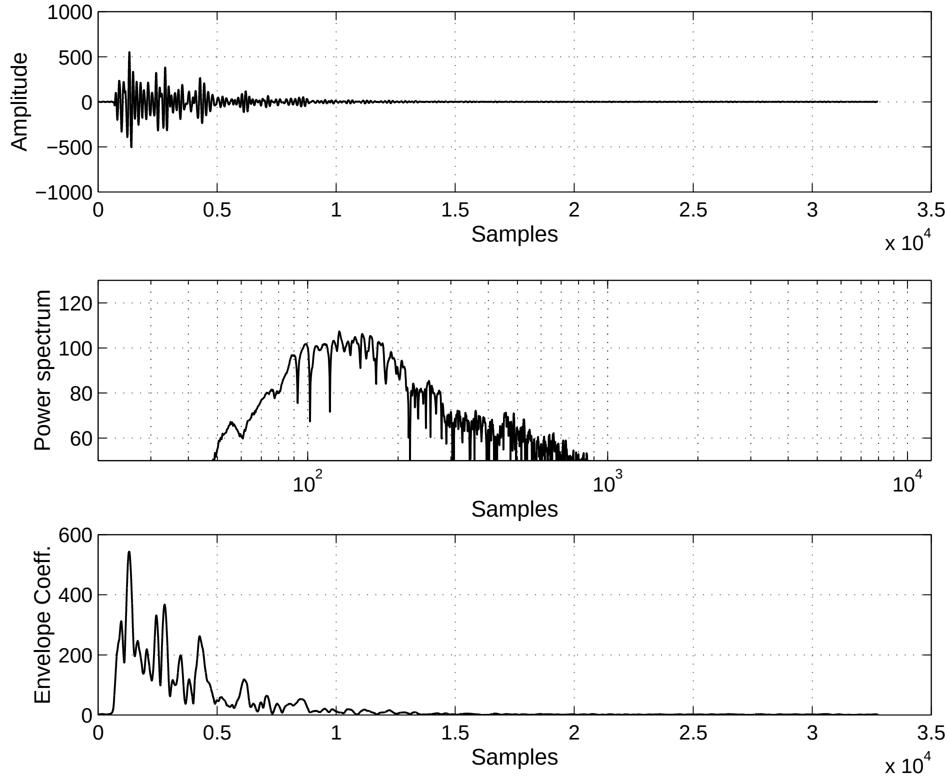


Fig. 9. Filtered Room Impulse Response: spectrum and envelope.

The following equation enables the determination of the envelope with quick-variable and slow-variable components:

$$env(\tau) = \sqrt{h_{RB}^2(\tau) + (\mathcal{H}(h_{RB}(\tau)))^2}, \tag{12}$$

where $\mathcal{H}$ – Hilbert transform.

The impulse response envelopes after the DWT (Discrete Wavelet Transform) process are shown in Fig. 10. The DWT process is equivalent to the octave filtration.

The envelope quick-variable filtered component arises after the high-pass filtering of the reversed initial envelope:

$$envi(\tau) = \int_{T}^{0} env(\tau - l) h_{HP}(l) dl. \tag{13}$$

The smooth envelope spectral characteristic is determined as follows:

$$O(\omega) = |\mathcal{F}(env(\tau)) * h_{av}(\omega)|. \tag{14}$$

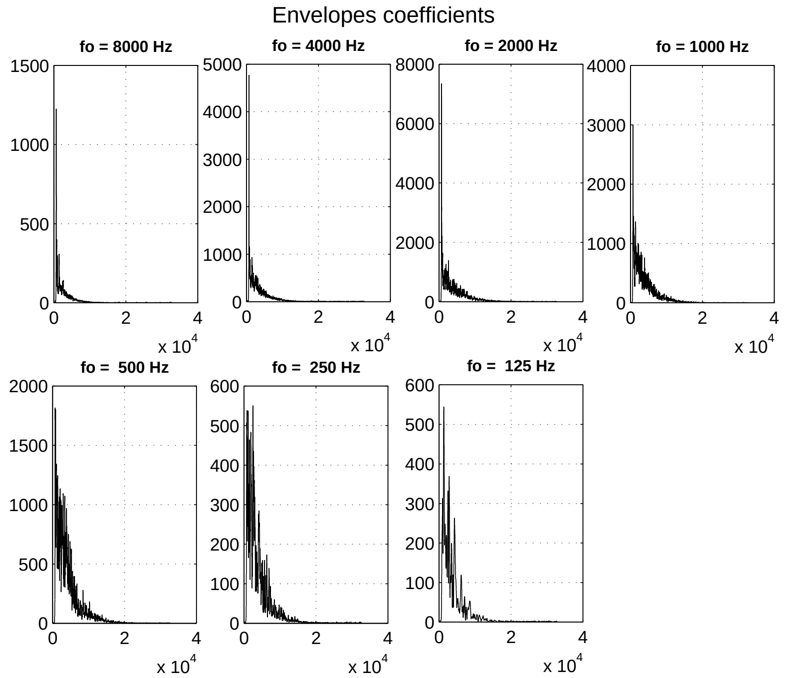


Fig. 10. The impulse response envelopes after the DWT process, f_0 – the octave band frequency.

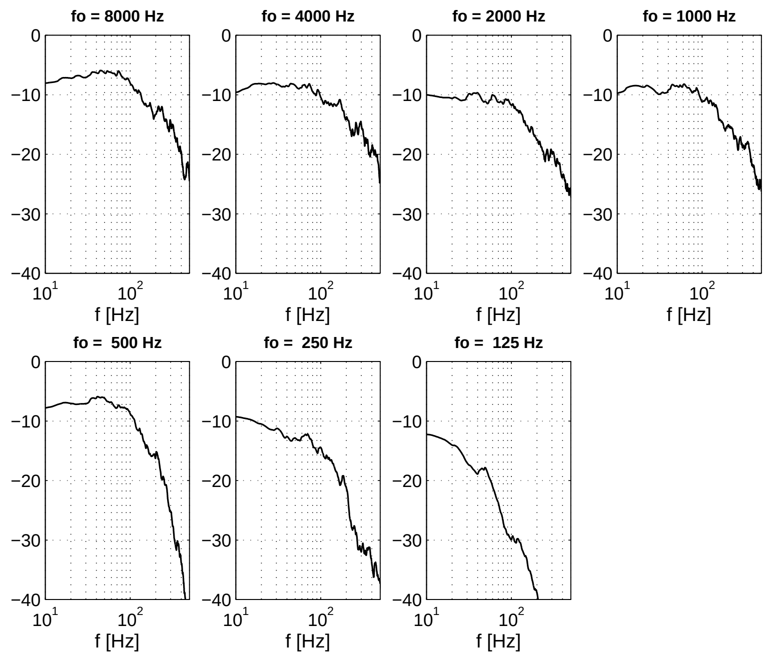


Fig. 11. The envelope spectral characteristics of the filters.

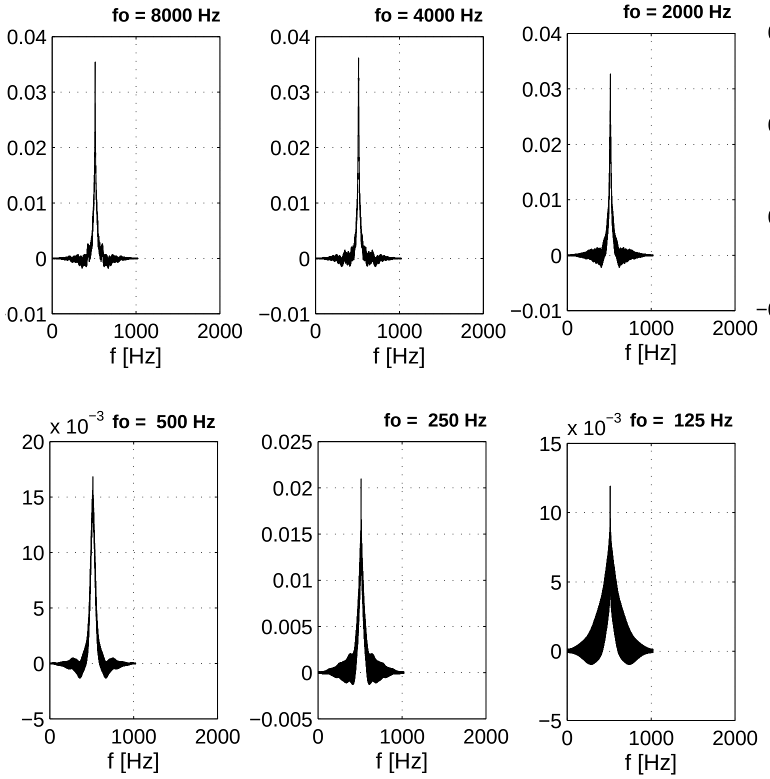


Fig. 12. The envelope impulse responses of the filters.

The averaging-linking filter impulse response is determined as follows:

$$h_{av}(\tau) = \frac{1}{2\pi} \int_{F1}^{F2} O(\omega) H_d(j\omega) d\omega, \tag{15}$$

where $H(j\omega) = \mathcal{F}(\delta(\tau - T/2))$, $F1$ and $F2$ – the filter limit frequencies.

The envelope time characteristic, determined by the Eq. (14), can be the base for creating a model in the form of the low-pass filter impulse response. The characteristic determines the f_{gr} limit frequency and the increasing high frequency-attenuation factor.

The time chart of the entire impulse response model is determined as follows:

$$h_R(\tau) = \sum_{n=1}^{K} h_{RB}(\tau), \tag{16}$$

where K – the frequency bands quantity.

3.2. Impulse response synthesis

The *wavelet transform* is used for creating the impulse response synthetic models. In order to analyze the signals, the wavelet transform uses the so-called *wavelets*, a family of functions like the Fourier transform uses the sine and cosine functions. The wavelets are functions which meet the given mathematical requirements and are used to represent data and/or other functions [14, 4].

The wavelet analysis assumes a basic wavelet function, the so-called *mother wavelet*. The time analysis is operated by means of the mother wavelet narrowed high frequency version; the frequency analysis uses the low frequency shifted version of the wavelet.

The wavelet continuous transform is determined as follows:

$$C(k,h) = \int_{-\infty}^{\infty} f(t)\Phi(k,h,t)dt, \tag{17}$$

where $f(t)$ – the analyzed signal, $C(s,k)$ – transform factors, $\Phi(k,h,t)$ – mother wavelet.

The mother wavelet scaling is necessary in order to include the entire data range (18):

$$W(x) = \sum_{k=-1}^{N-2} (-1)^k c_{k+1}\Phi(2x+k), \tag{18}$$

where $W(x)$ is the Φ mother wavelet scaling function. If the c_k wavelet factors meet the following conditions:

$$\sum_{k=0}^{N-1} c_k = 2, \qquad \sum_{k=0}^{N-1} c_k c_l = 2\delta_{l,0}$$

the *Discrete Wavelet Transform* is obtained (*DWT*).

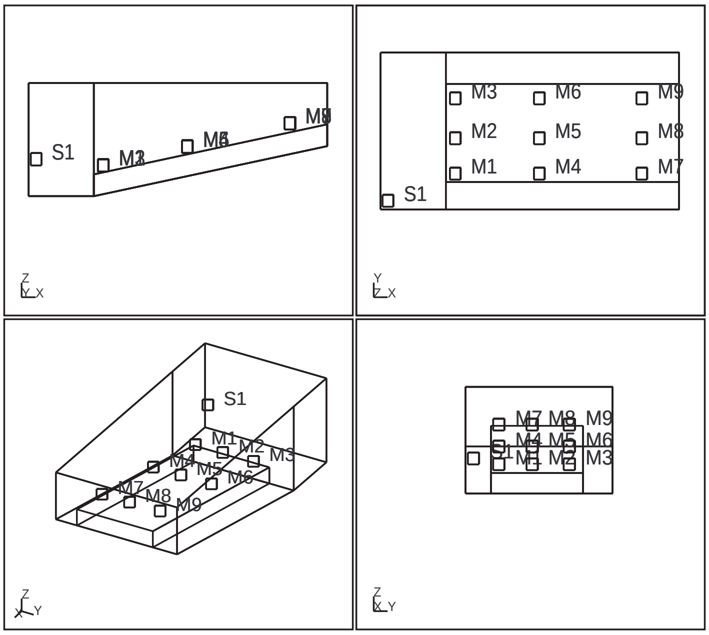
The $c_0 \dots c_n$ factors are ordered accordingly to two categories: the first one operates like a smooth filter and the second one gives information on the signal "details". These ordered filter factors are called the *quadrature mirror filter pair* QMF [19].

The factors posses the perfect reconstruction features, thus the wavelet transform become an invertible one.

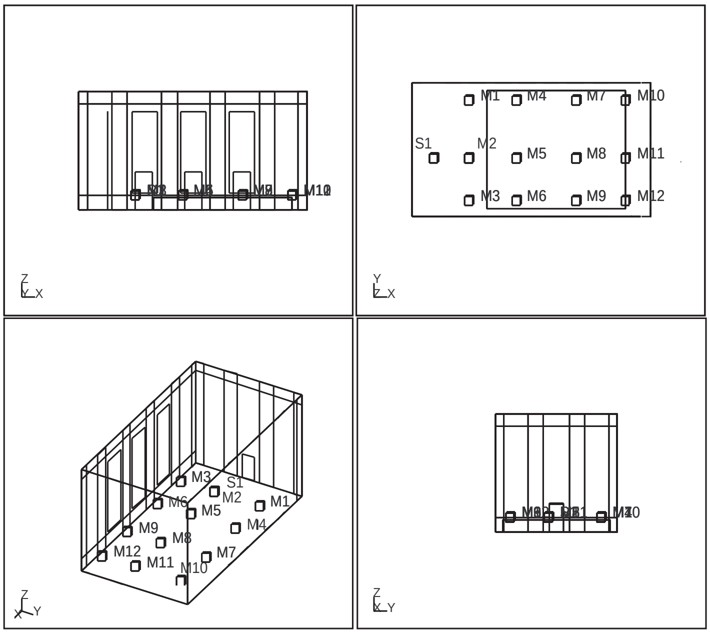
3.2.1. Acoustic field distribution analysis

Two rooms were selected for the tests: an amphitheatre lecture hall and the AGH – University of Science and Technology Hall.

The testing points (receivers) and the sound source were placed in the room geometrical models in a manner reflecting the real measurement points' positions exactly. The rooms' geometry and the sound sources and receiver positions are shown in Fig. 13(a) and Fig. 13(b).

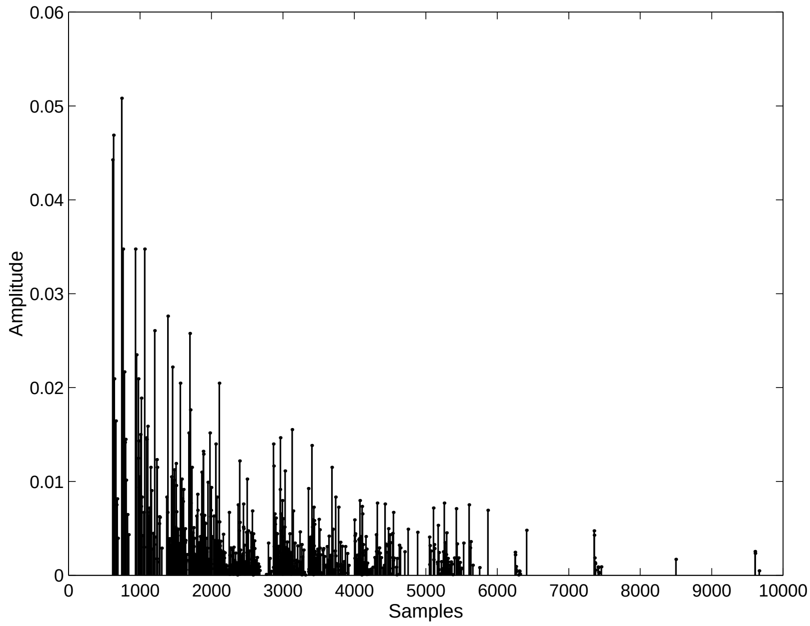


(a) The Hall No. 104 geometry with the sound sources and receivers positions.

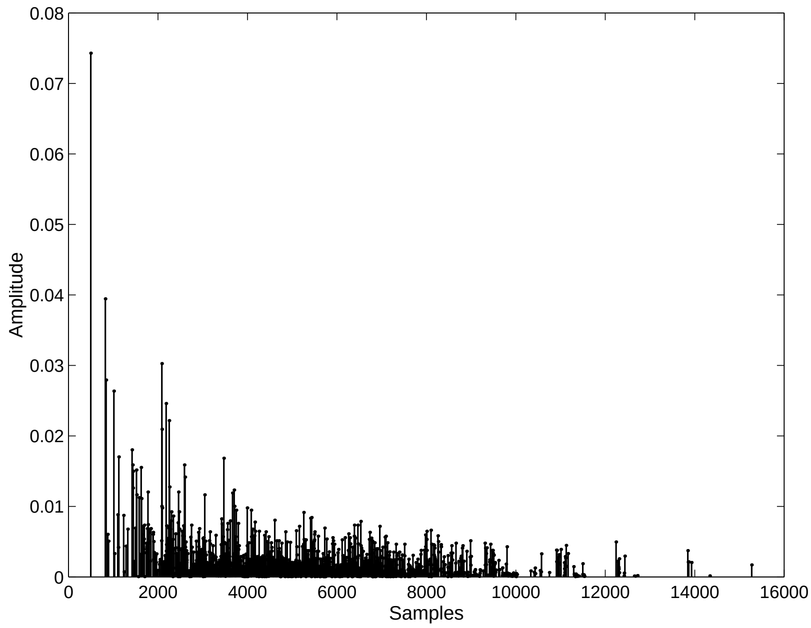


(b) The AGH Hall geometry and the sound sources and receivers positions.

Fig. 13. The 1 kHz octave reflectograms.



(a) Hall No. 104.



(b) AGH Hall.

Fig. 14. The 1 kHz octave reflectograms.

The surface materials absorption coefficients were determined on the basis of [21, 22] and the Raynoise package database. The table below includes the average absorption coefficients for the simulation models:

Table 1. Mean materials absorption coefficients.

	63 Hz	125 Hz	250 Hz	500 Hz	1kHz	2kHz	4kHz	8kHz
Hall No. 104 α_{av}	0.146	0.136	0.145	0.164	0.174	0.178	0.173	0.173
AGH Hall α_{av}	0.027	0.031	0.049	0.054	0.123	0.129	0.149	0.178

The reflectograms and the parameters C_{50}, D_{80}, early decay time EDT, reverberation time RT_60, and the speech transmission index STI were determined from the acoustic field analysis by means of the results of the geometrical methods. Selected reflectograms determined at the audio reception points are shown in Figs. 14(a) and 14(b).

3.2.2. Impulse Response Synthesis

During the first stage of the synthesis process, white noise sequences were generated. The generated signal was operated by means of the seven-level wavelet decomposition. In this way 8 noise bands were obtained (appropriately to 8 frequency bands, for which the echo diagrams were determined). If N_w is a white noise signal, its wavelet decomposition can be described as follows:

$$N_w(j,k) = \sum_{n \in Z} N_w(n) g_{j,k}(n), \tag{19}$$

where j – decomposition level, $N_w(j,k)$ – the j decomposition white noise signal, $g_{j,k}(n)$ – the wavelet used in the analysis.

The appropriate envelope was put onto each noise band:

$$h_i = N_w(i) \cdot Env_i, \tag{20}$$

where h_i – the *i* partial impulse response, $N_w(i)$ – a white noise signal after the *i* decomposition, Env_i – the *i* band envelope.

In this way the partial impulse responses of the given points were determined for the octave bands. A complete synthetic impulse response is obtained by means of the inverse wavelet transform using:

$$h_s(t) = A_J + \sum_{j \leqslant J} D_j, \tag{21}$$

where

$$D_j(t) = \sum_{k \in Z} C(j,k) \Psi_{j,k}(t). \tag{22}$$

4. Evaluation of the impulse response synthetic model

The verification of the reasoning described above was done by means of a comparative evaluation of the modeled impulse responses and the measured responses. The evaluation was based on the following criteria:

- the objective criterion: the acoustic field parameters subset (EDT, C_{80}, D_{50}, STI);
- the subjective criterion: the audio tests of the signals obtained as the results of the impulse response and test signal convolution.

In order to verify the models, the following parameters were selected: the early decay time EDT [13], the clarity C_{80}, the definition D_{50} [7] and the speech transmission index STI [7, 11].

The parameters were determined on the basis of measured impulse responses and the synthetic models.

The tables below include the calculated parameters for the various models. The parameters were calculated on the basis of the reflectograms by means of the Raynoise package and the measured impulse responses.

Table 2. C_{80} values, Hall No. 104, point 1.

C_{80}[dB] The Envelope Approximation Method							
f [Hz]	Line	Rectangles	Exponential	Savitzky–Golay Filter	Envelope Filters	Raynoise	Measurements
125	1.26	2.0	1.05	5.8	7.79	2.9	3.66
250	3.14	1.89	1.06	5.28	6.72	3.3	9.39
500	2.73	2.62	0.59	4.14	7.03	4.2	5.20
1k	3.60	6.62	1.07	4.36	7.38	4.7	6.49
2k	3.79	2.14	2.97	5.33	11.88	5.2	11.7
4k	2.56	3.24	4.99	6.82	12.14	5.6	15.47

Table 3. D_{50} values, Hall No. 104, point 1.

D_{50} [%] The Envelope Approximation Method							
f [Hz]	Line	Rectangles	Exponential	Savitzky–Golay Filter	Envelope Filters	Raynoise	Measurements
125	45.35	36.52	49.86	66.1	71.33	55.3	62.64
250	24.30	35.73	41.86	30.67	90.58	55.3	83.33
500	45.03	27.67	41.26	44.72	70.94	59.8	64.76
1k	57.22	12.05	45.17	57.54	71.30	62.4	74.89
2k	44.27	30.62	55.22	64.80	90.33	64.8	91.06
4k	31.82	40.72	62.75	65.30	92.76	67.0	96.31

Table 4. EDT values, Hall No. 104, point 1.

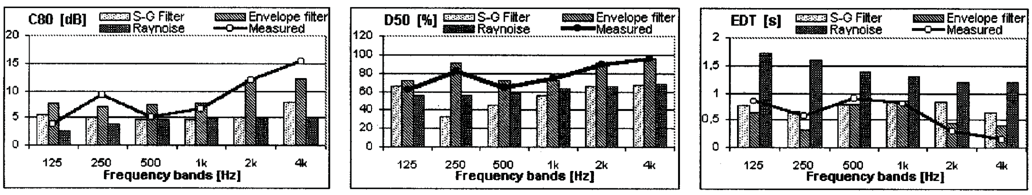
EDT [s] The Envelope Approximation Method							
f [Hz]	Line	Rectangles	Exponential	Savitzky–Golay Filter	Envelope Filters	Raynoise	Measurements
125	1.69	1.21	1.44	0.75	0.65	1.7	0.87
250	1.44	1.21	1.14	0.66	0.31	1.6	0.54
500	0.90	1.12	1.16	0.74	0.78	1.4	0.94
1k	1.02	1.15	1.33	0.89	0.51	1.3	0.79
2k	1.71	1.08	1.27	0.84	0.42	1.2	0.29
4k	1.7	1.01	0.96	0.62	0.14	1.2	0.12

Table 5. STI values, Hall No. 104, point 1.

STI [$-$] The Envelope Approximation Method							
f [Hz]	Line	Rectangles	Exponential	Savitzky–Golay Filter	Envelope Filters	Raynoise	Measurements
1	0.51	0.57	0.56	0.67	0.84	0.53	0.81
4	0.55	0.59	0.51	0.65	0.66	0.46	0.60
5	0.53	0.60	0.58	0.67	0.50	0.46	0.61
7	0.53	0.59	0.62	0.67	0.56	0.44	0.62

The comparison of the best models' parameters, measured responses and calculation results are shown in Figs. 15 and 16.

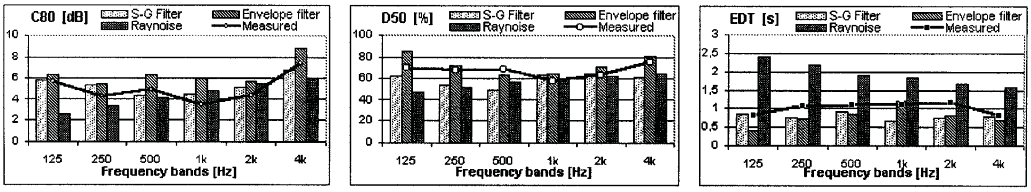
The C_{80}, D_{50}, EDT and STI parameter values, determined on the basis of the synthetic models are close to the appropriate ones determined by means of the real impulse responses. The differences are mostly less than in the case of the appropriate direct values calculated adequately on the basis of the echo diagrams and the real impulse responses.



(a) C_80 Hall No. 104, point 1 (b) D_{50} Hall No. 104, point 1 (c) EDT sala 104, p.1

Fig. 15. The parameters determined on the basis of the various models.

The quality of the models was evaluated on the basis of the subjective audio tests for a group of five individuals. The following evaluation scale was used:



(a) C_{80} Hall No. 104, point 4 (b) D_{50} Hall No. 104, point 4 (c) EDT Hall No. 104, point 4

Fig. 16. The parameters determined on the basis of the various models.

Table 6. Evaluation scale in subjective audio tests.

	Grade
5	Very good
4	Good
3	Satisfactory
2	Poor
1	Unsatisfactory

The main task of the listeners was to evaluate the convolution quality, the reverberation characteristic and some sound distortions for each model.

During the second series of the tests, the listeners evaluated the synthetic impulse response convolutions and the convolutions with the measured impulse response. A musical phrase recorded within the echo-free room was used as the test signal.

Summary of the listeners' impressions:

- there are significant tone differences between the various synthetic models;
- the convolutions with the models are significantly "brighter" in comparison with the real impulse responses, which is caused by the presence of high frequencies;
- the convolution with the model created by means of the echo diagram averaging with an exponential filter caused a distinctly metallic sound;
- the convolution with the model with the envelope filters caused sound distortions; the resonance could be heard;
- the convolution with the model with the Savitzky–Golay filter was most similar to the convolution with a real impulse response.

The individual results of the models evaluation are given in Table 7a. The maximum grade was 25 points.

Table 7. Results of the models subjective tests evaluation.

(a) Individual models

No.	Model envelope approximation method	Grade
1	Savitzky-Golay filter	23
2	Rectangles	19
3	Expotential	18
4	Line	15
5	Envelope filters	12

(b) Comparison

No.	Model envelope approximation method	Grade
1	Savitzky-Golay filter	24
2	Rectangles	19
3	Line	18
	Expotential	18
4	Envelope filters	16

As the results show, the models subjective evaluation is not unambiguous. The synthetic model with the Savitzky–Golay filter got the best grades in the two testes, but the differences between the remaining models were rather small.

5. Summary

In general, the impulse response modeling described enabled the obtaining of models good quality

- the Savitzky–Golay filter reflectogram approximation model;
- the envelope modeling by means of the envelope filter averaging.

The best results were obtained for the reflectogram interpolation by means of the filters envelope synthetic models. In this instance, almost all the calculated parameter values were close to the parameters determined on the basis of the real impulse responses.

The audio tests showed that the test signal and the impulse response synthetic models convolution caused the impression of a very natural reverberation and a room three-dimensional effect.

Nevertheless, a difference between the signal convolution and, adequately, the synthetic model and the real impulse response was heard.

Unfortunately, the manner of modeling based on the reflectograms is very sensitive to the “quality” of the geometrical model and hence to the reflectograms.

The impulse response synthetic models are very sensitive to the quality of the room acoustic field analysis. The object geometry mapping, the sound sources accuracy and, first of all, the acoustic wave propagation associated effects (reflection, dissipation) have a fundamental significance for the correct modeling.

The creating of an appropriate model of the sound source is very significant because, as it has been proved empirically, the echo diagrams accuracy and the impulse response synthetic models depend predominantly on the sound source model.

The algorithms of the room impulse response synthesis, based on the acoustic field model analysis by means of geometrical acoustics that is a unique achievement of the authors of this article, will enable the creation of an acoustic evaluation methodology during the projection and modernization stages.

An objective evaluation based on the numerical factors determining is possible, and a subjective evaluation, which should be particularly stressed, can be processed too on the basis of the audio tests.

It is necessary to run the next tests with relation to the described problems. A modification of the acoustic field model as well as taking into consideration the wave effects will enable the achievement of more precise impulse response models and hence the preparation of a more accurate methodology of evaluation of the room acoustic properties.

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