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COMPARISON OF METHODS FOR MEASURING REVERBERATION TIME

The article presents the results of an experimental evaluation of the reverberation time of the lecture auditorium in the Lviv Polytechnic National University. For their further analysis a sub-system of batch processing of audio files with a registered response of the room for excited impulse noise has been developed. To determine the bypass signal, Schroeder's method was used and the least squares method was improved by pre-processing of the signal using the MM method. The influence of the number of neighboring points of the MM method on the accuracy of the time of reverberation method by the least squares method is investigated. A comparison of the Schroeder method and the least squares method was made, which made it possible to establish that Schroeder method is more precise. The average error of this method for 6 experiments against the Dirac system was 0.02 sec, whereas for the least squares method it was 0.06 seconds. The developed subsystem of the batch analysis of the registered response of the premises for pulsed noise made it possible to increase the speed and efficiency of processing the results of experiments.

Key words: reverberation time, lecture auditorium, impulse noise, Schroeder's method.

ПОРІВНЯННЯ МЕТОДІВ ВИМІРЮВАННЯ ЧАСУ РЕВЕРБЕРАЦІЇ

Наведено результати експериментальної оцінки часу реверберації лекційної аудиторії Національного університету “Львівська політехніка”. Для їх подальшого аналізу розроблено підсистему пакетного опрацювання аудіофайлів із зареєстрованим відгуком приміщення на збудження імпульсним шумом. Для визначення обвідної сигналу використано метод Шредера та удосконалено метод найменших квадратів попереднім обробленням сигналу методом змінного середнього. Досліджено вплив кількості сусідніх точок методу змінного середнього на точність визначення часу реверберації методом найменших квадратів. Здійснено порівняння методу Шредера та методу найменших квадратів, яке дало змогу встановити, що метод Шредера є точнішим. Середня похибка цього методу для шести експериментів відносно відомої серед акустиків системи Dirac становила 0,02 с, натомість для методу найменших квадратів 0,06 с. Розроблена підсистема пакетного аналізу зареєстрованого відгуку приміщення на імпульсний шум, яка дала змогу збільшити швидкодію та ефективність опрацювання результатів експериментів у декілька разів.

Ключові слова: час реверберації, лекційна аудиторія, імпульсний шум, метод Шредера.

Introduction

Reverberation time is one of the important criteria that determines the acoustic quality of any room. According to ISO 3741: 2010 [1], reverberation time is the time required to reduce the averaged volume of sound energy density in a closed environment at 10 ^ (n / 10) times, that is, at n dB after switching off the noise source. Nowadays there are a lot of systems for determining the reverberation time based on the analysis of the registered impulsive response of the room to excitation impulse noise, but they only allow to process one audio file. To conduct research to improve methods for determining reverb time scientists need to handle large amounts of files. The task to develop a system that allows automating the process of batch processing of a large number of audio files is important, and for this purpose it is necessary to choose the optimal method or to improve the existing ones for the purpose of implementation in the system being developed.

Scientists Wallace Sabine, Manfred Schroeder, Leo Beranek, Tim Mellow and others [2–3] made a significant contribution to the creation of methods for calculating reverberation time. Each of these methods allowed to improve the accuracy of calculations, but at the same time each of them has its advantages and disadvantages. In this regard, the efforts of modern researchers are aimed at clarifying the calculated formulas of reverberation time [4], as well as the study of methods of determining of the signal bypass for finding the reverberation time and developing systems in which these methods are implemented [5].

Registering a room response to pulse noise

According to ISO 3382-2 [6] reverberation time is measured with a pulse transition characteristic, which is to change the sound pressure at a certain point of the room as a result of radiation of the Dirac pulse at another point in the room. In order to reduce the impact of direct radiation the minimum calculated distance is 2.14 m, in our case, the closest distance from the source of noise to the measuring point is 2.8 m, which meets the requirements of the standard.

The layout of the measuring points and the source of impulse noise is shown in Fig. 1. At each measuring point, the maximum sound pressure level in dB is specified. As a source of impulse noise, petards were used. Shots were carried out on the spot where the lecturer is usually placed during the lecture. The microphone was placed at a height of 1.2 m from the floor where the ears of the listeners sitting at the desk are located.

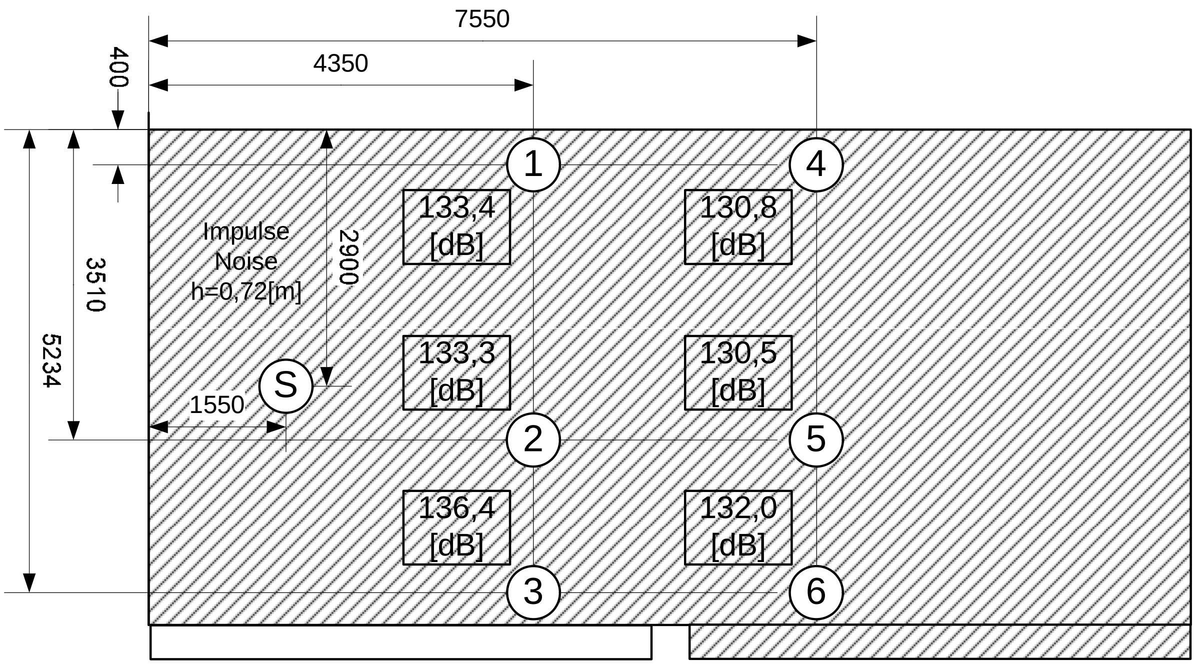

Fig. 1. Layout of measuring points

During the experiments the impulse source generated a peak sound pressure level in which the initial level of the downturn curve was in all cases greater than 45 dB from the background noise level in the corresponding band of frequencies when measuring T30 [6].

The following equipment was used for the experiment:

- four channel sound and vibration analyzer SVAN 958A;

- capacitor omnidirectional microphone SV22 1/2 ", measurement range from 10 Hz to 16 kHz, 25 mV / Pa sensitivity allows to measure high noise level;
- petards (source of impulse noise);
- Bosch GLM30X laser measure.

All experiments were carried out at room temperature of 22 °C and relative humidity of 45 %. Fig. 2 shows the process of registering an auditorium response to a pulse source of noise using a SVAN 958A analyzer.

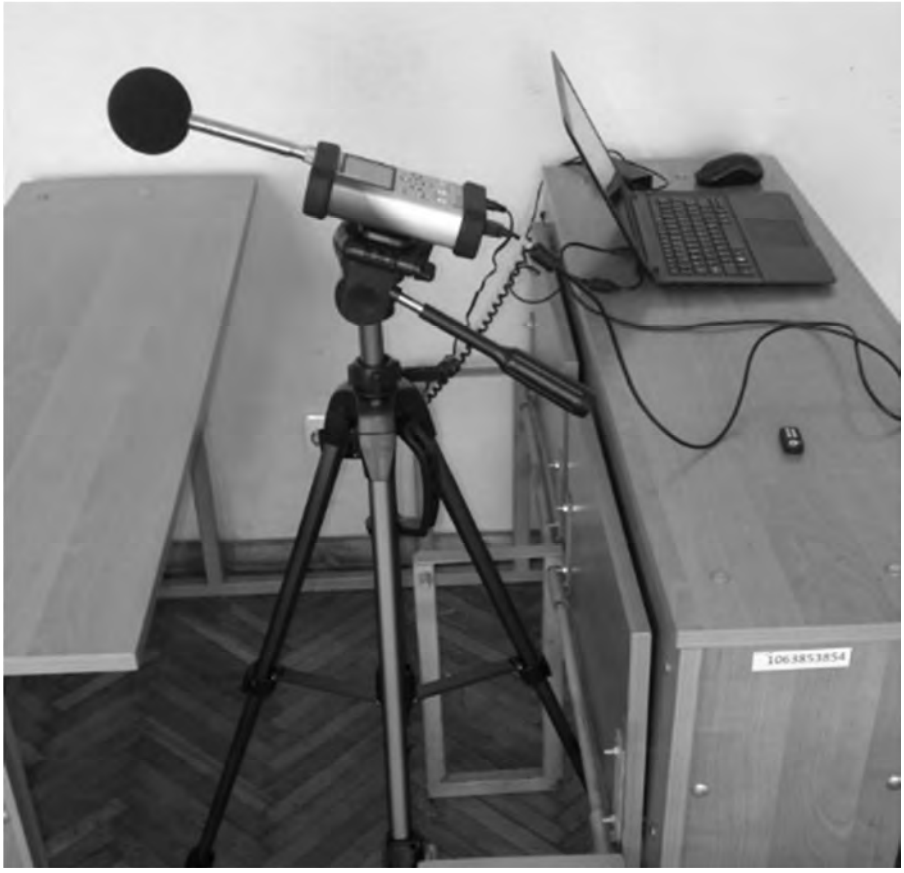

Fig. 2. Registration of the auditorium response to impulse noise source

Design of a subsystem

The subsystem for analyzing the registered audio files with the response of the room to impulse excitation was developed taking into account all the requirements of the standard ISO 3382-2 [6].

When developing the subsystem it was taking into consideration that it is necessary to shift by 5 dB from the peak value, and if the difference between the peak value and the background noise is less than 35 dB, the program will issue an error that the file does not meet the requirements. The subsystem was developed in the MatLab environment, because this system has sufficient means to solve such tasks. The development takes into account that audio files may have an incorrect checksum and signalizing an error when opening in MatLab. Therefore, a function was developed that fixes these files and allows to process them. The algorithm of the work of the developed subsystem is presented in Fig. 3.

Method of Moving Mean

In the Moving Mean (MM) method, the output data is smoothed by the following rule [7]:

$$ys_i = \frac{1}{2N+1}\sum_{k=-N}^{N} y(\mathrm{i}+k), \qquad (1)$$

where *2N+1* – the number of points that are chosen for smoothing. That is, to the left and to the right of the current point is N points are selected. The data located at the points close to the boundaries of the segment are not smoothed, because there are not enough points to the right or left of the current point at which smoothing is being done.

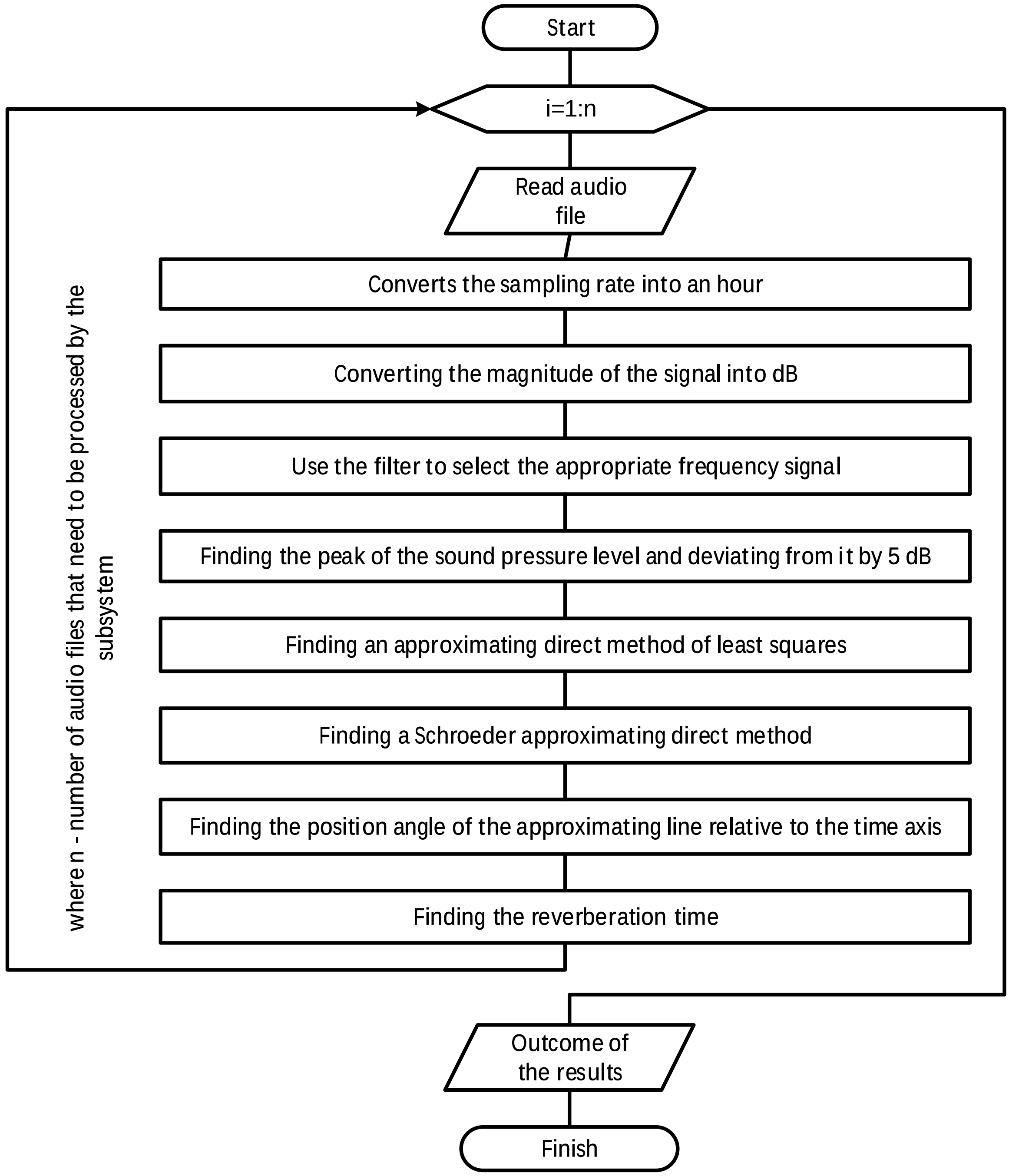

Fig. 3. The algorithm of the subsystem for determining the reverberation time for audio files

That is why for analysis in the developed system the time interval bigger than necessary is analyzed so that the results are not distorted. The whole signal next to the peak value is also cut off since the average value between the background noise and the peak values greatly distorts the results. The MM method is used before using the least squares method to find the optimal time interval for analysis. Results of the determination of the reverberation time had an absolute error 5 times greater without using the pre-processing of the signal by the MM method results. This results are unacceptable. The conclusion can be made that the method of least squares to determine the bypass signal in order to determine the reverberation time can only be used after signals pre-processing.

In Fig. 4 the results of the smoothing of the signal by means of the MM for 10, 100 and 1000 smoothing points are presented. As one can see from the graph the signal becomes more acceptable for further processing with the increase of points of smoothing.

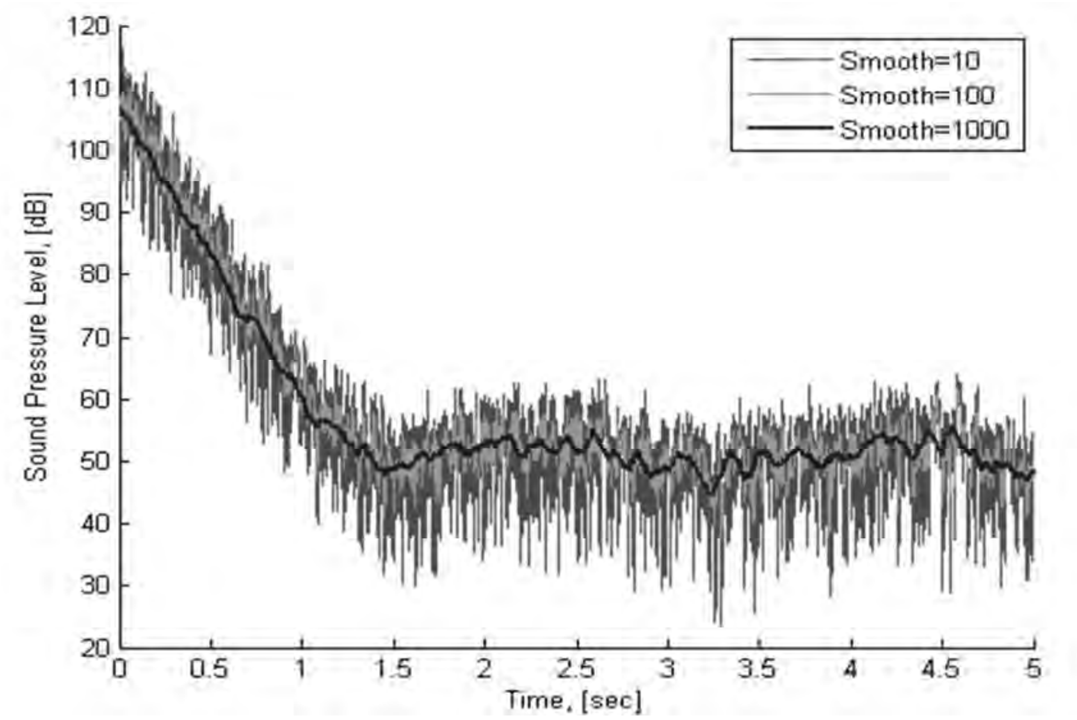

Fig. 4. Smoothing results for different number of smoothing points

The next step was to find the optimal number of points for smoothing. That is why the task was set to investigate how the total error of reverberation time would change for 6 experiments in comparison with the Dirac system. From the obtained calculations it turned out that the smallest error is obtained by smoothing the signal of 2000 neighboring points. The graph of the dependence of the number of adjacent points for smoothing (Fig. 5). In connection with this 2000 thousand adjacent points for smoothing the signal have been selected in the system.

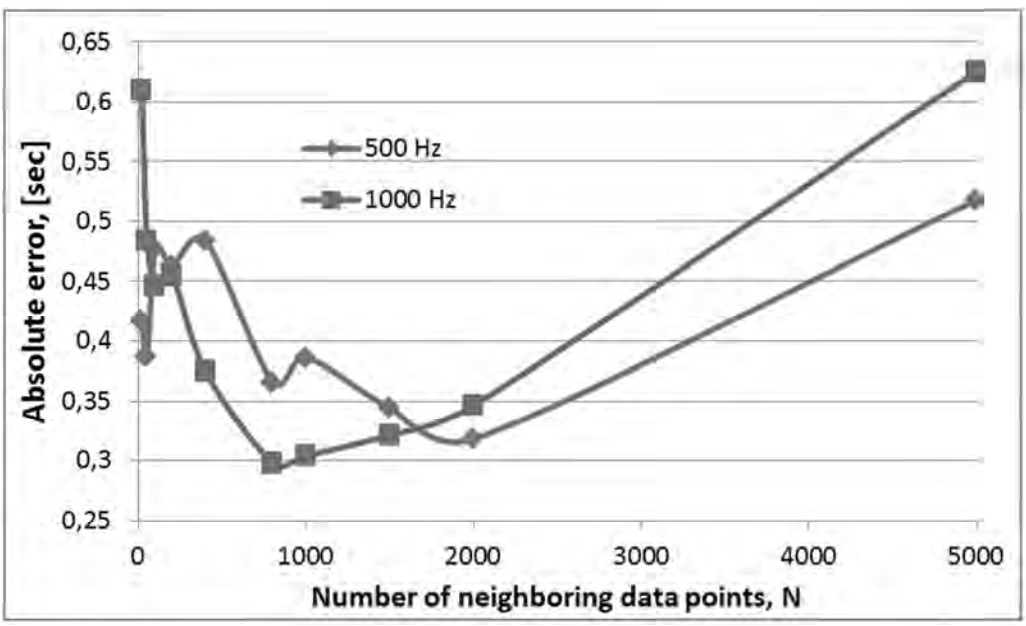

Fig. 5. Investigation of the influence of the number of adjacent points on the total error

Results and their analysis

To study the accuracy of the subsystem and compare the Schroeder method with the modified least squares method, the results of the reverberation time T30 for frequency f = 500 Hz and f = 1000 Hz were taken, which are most often used to assess the acoustic suitability of the room, and sometimes their mean value is taken. The results of the subsystem data processing and the results obtained from the program Dirac [8] are presented in Table 1 and Table 2.

Table 1

Time reverberation T30 for frequency f = 500 Hz, sec

Experiment #	1	2	3	4	5	6
Dirac system	1.2000	1.2600	1.1500	1.2500	1.2300	1.2300
Schroeder's method	1.2258	1.2345	1.1440	1.2455	1.2203	1.1820
Method of Least Squares	1.2491	1.1679	1.2273	1.3218	1.2513	1.2235

Table 2

Time reverberation T30 for frequency f = 1000 Hz, sec

Experiment #	1	2	3	4	5	6
Dirac system	1.1800	1.2000	1.1100	1.2200	1.1700	1.2300
Schroeder's method	1.1833	1.1593	1.1363	1.2395	1.1713	1.2633
Method of Least Squares	1.2138	1.2611	1.1727	1.2889	1.2435	1.2765

Since we work with roughly identical values, we use only absolute errors to evaluate the correctness of the work of the developed system and the comparison of the methods. The absolute errors obtained for the Schroeder method and the improved least squares method are presented in Fig. 6 for frequency 500 Hz and on Fig. 7 For frequency 1000 Hz.

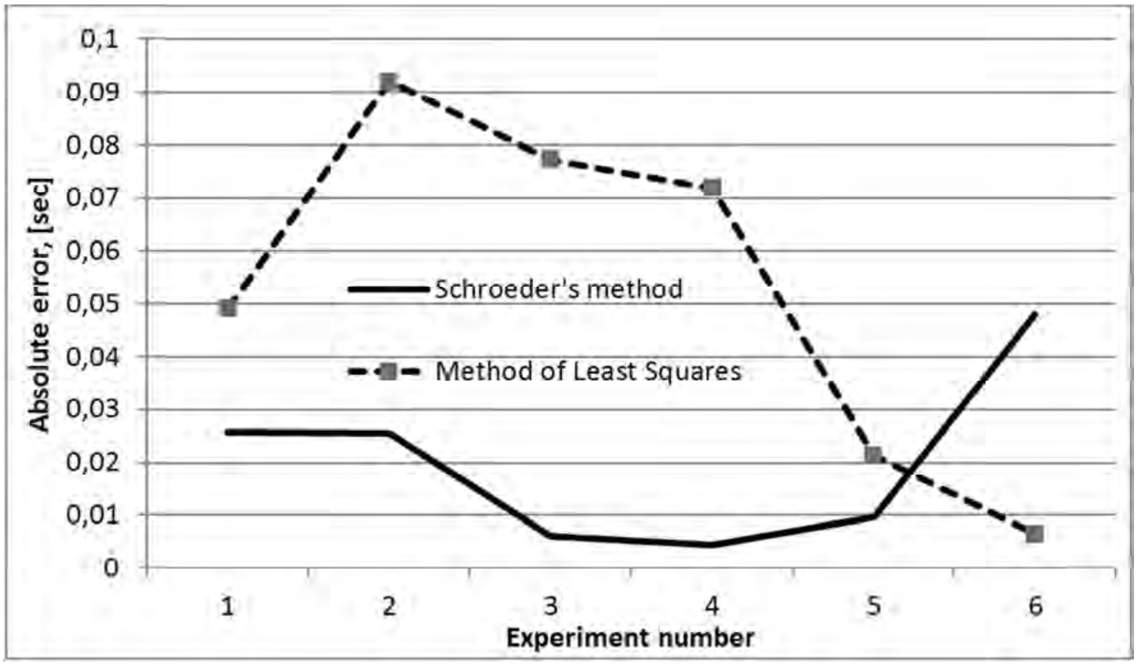

Fig. 6. Absolute error for frequency 500 Hz

As one can see from the graphs in Fig.6 and Fig.7 the Schroeder method has got the lesser error. To better present the difference we determine the sum of the absolute errors of all experiments. The results are presented in the Table 3.

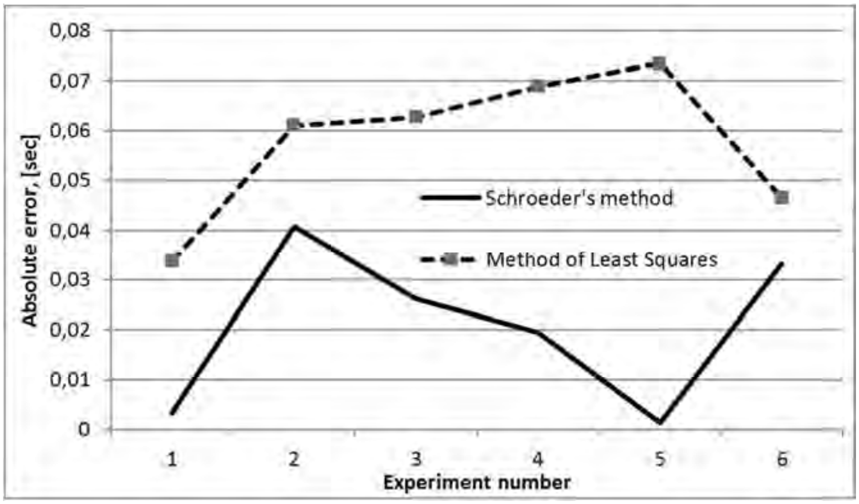

Fig. 7. Absolute error for frequency 1000 Hz

Table 3

The sum of absolute errors for 6 experiments in comparison with the results obtained from the Dirac system [9]

	The sum of absolute errors for 6 experiments, [сек]	
Frequencies, [Hz]	Schroeder's method	Method of Least Squares
500	0.1195	0.3180
1000	0.1243	0.3464

Analyzing the results of the total absolute error in all experiments, we see that the maximum total error for the Schroeder method is much smaller than the least squares method and for all 6 experiments is 0.12 sec. This small error is a very good result, because if you divide it into 6 experiments, then the average error will be 0.02 sec, which confirms the effectiveness of the developed subsystem. Application of the pre-processing of the signal in the method of least squares made it possible to achieve greater accuracy of the results. The average error in 6 experiments for the least squares method does not exceed 0.06 sec which is also a very good result.

Conclusions

A subsystem has been developed that allows to investigate the accuracy of methods of inverse integration and least squares on experimental data. Based on the results of the research, the method of least squares was improved by using the pre-processing of the signal by the method of the alternating average. However, even the improvements made it impossible to obtain the accuracy of the Schroeder method in relation to the Dirac system, which can be explained by the fact that the Dirac system also uses the Schroeder method. The mathematical and software of the subsystem of automatic estimation of acoustics of premises by impulse method has been developed.

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