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REVERBERATION TIME ESTIMATION ALGORITHM ACCURACY

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Abstract*—The task of estimating the reverberation time is relevant in the acoustic examination of premises for both civilian purposes (kindergartens, educational institutions, concert halls, etc.) and military purposes (control centers). Reverberation time measurement is usually performed by the method of inverse integration of the room impulse response. However, due to the existence of background noise, the problem of choosing the moment of time (truncation time) from which integration should begin arises. In this paper, the accuracy of the algorithm for calculating the reverberation time, where the truncation time is defined as the moment of approach to zero of the derivative of the "squaring - moving averaging" system output signal, is studied. Estimates of the bias, standard deviation, and total error for the values of the signal-to-noise ratio and reverberation time typical for classrooms are obtained. At a signal-to-noise ratio of 45 dB, for measurements in a wide frequency band of 80 Hz – 11 kHz, the total relative error of the reverberation time estimation does not exceed 6% for reverberation time values of 0.6–1.2 s. When measuring reverberation time in octave frequency bands, the error reaches 20% for the frequency band in the vicinity of 125 Hz, decreases to 10% for the frequency band in the vicinity of 500 Hz, and does not exceed 3% for the frequency band in the vicinity of 8 kHz.*

Keywords—Reverberation time; truncation time; room impulse response; squaring – moving averaging; inverse integration method.

I. INTRODUCTION

The task of assessing the reverberation time T_{60} in rooms is relevant, as it allows one to draw a conclusion about the suitability of rooms for speech communication. For example, the reverberation time should not exceed 0.6–0.8 s in the premises of kindergartens and primary grades of secondary schools [1] – [4]. For high school and university students, these requirements are somewhat relaxed, but the reverberation time should not exceed 1–1.2 s [5], [6]. The issue of the quality of acoustics in military premises, an example of which are various control centres, is also important. In all these cases, compliance with a certain reverberation time is a necessary condition for ensuring high speech intelligibility. When measuring the reverberation time, the noise interruption method or the method of analyzing the room impulse response (RIR) $h(t)$ is used. The envelope of the signal $h^2(t)$, which is necessary for the calculation of the reverberation time, is obtained by one of two methods [7] – [10]. According to the first method, the signal $h^2(t)$ is subjected to moving averaging.

$$D_1(t) = \int_{-\infty}^{t} w(t-\tau)h^2(\tau)d\tau, \qquad (1)$$

where $w(t)$ is the impulse response of the filter implementing moving (exponential or linear) averaging, T is the effective averaging time, which should be at least 3 times smaller than the predicted reverberation time [7].

Today, preference is given to another method of calculating the envelope of the signal $h^2(t)$, namely, the calculation by the "inverse integration method" [10]

$$D_s(t) = N\int_t^{\infty} h^2(\tau)d\tau \approx N\int_t^{T_i} h^2(\tau)d\tau, \qquad (2)$$

where N is proportional to the spectral power density of the test noise signal in the measurement frequency range, T_i is the right boundary of the informative part of the impulse response of the room $h(t)$ (the left boundary $t = 0$ corresponds to the $h(t)$ maximum).

Regardless of the calculation method, the resulting envelope is subjected to logarithmization, as a result of which the exponential law of decay of the signal level $h(t)$ is converted into a linear one. Next, the envelope is checked for exceeding the thresholds at the levels of minus 5 dB, minus 15 dB, minus 25 dB and minus 35 dB, after which the corresponding T_{10}, T_{20}, T_{30} estimates of the reverberation time T_{60} are calculated. The thresholds of 0 dB and minus 10 dB are used to estimate the Early Decay Time (EDT).

II. PROBLEM STATEMENT

A generally recognized problem when using the relation (2) is the presence of background noise $n(t)$, which necessitates the preliminary determination of the boundary T_i of the informative part of the RIR $h(t)$. There are various proposals regarding the algorithm for determining the parameter T_i. In article [11], it is proposed to choose T_i in the interval $0.5T_{60} < T_i < T_{60}$, taking into account the value of the dynamic range D of the estimate $h(t)$. However, the value of a systematic measurement error (bias) of T_{60} estimate is not taken into account. In article [12], this drawback is eliminated and it is proposed to take into account, in addition to T_{60} and D, the maximum relative bias 7% was accepted. In article [13], it is recommended to avoid finding the value T_i at all, subtracting the mean square of the background noise from $h^2(t)$. Although this idea was supported in [14], later the same authors [15] proposed an iterative algorithm for estimating T_i, the idea of which is to perform calculations according to (1) with different values of the averaging interval T. The mentioned algorithm is quite complicated for practical implementation. In addition, its resistance to background noise has not been studied sufficiently.

In articles [16], [17], a similar two-stage algorithm for estimating T_i is proposed, where at the first stage calculations are performed according to (1) and the preliminary value of T_i is found, and at the second stage this value is refined. A significant difference of the proposed algorithm is the use of the estimate of the derivative $dD_1(t)/dt$ of the function $D_1(t)$. In this case, the preliminary value of T_i is found as the time point where the curve $dD_1(t)/dt$ crosses the $thr \approx 0$ threshold. The disadvantage of the proposed algorithm is the complexity of substantiating the choice of the threshold value thr.

This paper attempts to eliminate this drawback, and instead of the threshold method, another approach is proposed, according to which the value T_i is defined as the time point where the derivative $dD_1(t)/dt$ changes sign for the first time.

III. PROPOSED ALGORITHM

The proposed algorithm consists of the following steps:

- the function $h^2(t)$ is corrected by subtracting the mean square of the background noise $n(t)$;
- the corrected function $h^2(t)$ is normalized by the maximum and the envelope $D_1(t)$ is calculated, according to (1);
- the sampling frequency f_s of the envelope $D_1(t)$ is reduced to $1/T$, resulting in a discrete sequence $D_1(iT)$, $i = 1,\ldots, M$;
- the discrete derivative $d_{1i} = D_1\left((i+1)T\right) - D_1(iT)$ is calculated and normalized by the maximum
- the estimate T_i^* of the parameter T_i is calculated as the time point where the maximum-normalized discrete derivative d_{1i} changes sign for the first time;
- the estimate T_i^* is corrected: $T_i^{**} = T_i^* - T$;
- EDT, T_{10}, T_{20}, T_{30} estimates are obtained using the curve $D_s(t)$ constructed according to (2) using the refined value of T_i^{**}.

IV. SET UP OF THE STUDY

To assess the accuracy of the proposed algorithm, a model of a non-stationary random process in the form of an additive mixture of the RIR $h_M(k)$ and discrete white noise was used [18]

$$y(k) = h_M(k) + n(k),\ k = 0,1,\ldots, K,\ K = T_M F_s, \tag{3}$$

$$h_M(k) = A_r v(k) e^{-\rho k T_s},$$

where $v(k)$ is a white Gaussian noise with zero mathematical expectation (ME) and unit variance; $\rho = 6.908/T_{60}$; T_{60} is the reverberation time; $T_s = 1/F_s$ is the sampling period; F_s is the sampling frequency, the noise $n(k)$ is a white Gaussian process with zero ME and variance σ_n^2; T_M is the duration in seconds of the model (3).

It is convenient to generate realizations of the model (3) according to the expressions

$$y(k) = v(k)\sigma(k),\ k = 0,\ldots, K, \tag{4}$$

$$\sigma(k) = \sqrt{A_r^2 \cdot e^{-2\rho k T_s} + \sigma_n^2}.$$

To control the level of background noise, it is advisable to use the concept of "signal-to-noise ratio". For model (3) this value is $\mathrm{SNR}_M = 20\lg\left(A_r/\sigma_n\right)$. Since

$$A_r/\sigma_n = 10^{0.05\cdot \mathrm{SNR}_M},$$

under the condition $A_r = 1$ we have $\sigma_n = 10^{-0.05 \cdot \mathrm{SNR}_M}$, and an expression convenient for modeling instead of (4)

$$y(k) = v(k)\sqrt{\exp\left(-13.8k/\left(F_s T_{60}\right)\right) + 10^{-0.1 \cdot \mathrm{SNR}_M}}. \quad (5)$$

Figure 1 shows an example of the realization of model (5) for the case $T_M = 4\,\mathrm{s}$, $F_s = 22050\,\mathrm{Hz}$, $T_{60} = 0.8\,\mathrm{s}$, $\mathrm{SNR}_M = 45 \cdot \mathrm{dB}$.

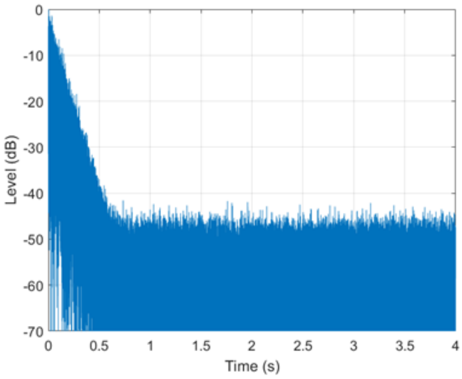


Fig. 1. Realization of model (5) for $T_M = 4\,\mathrm{s}$, $F_s = 22050\,\mathrm{Hz}$, $T_{60} = 0.8\,\mathrm{s}$, $\mathrm{SNR}_M = 45\,\mathrm{dB}$

V. RESULTS OF THE STUDY

Figure 2 shows the results of the preliminary verification of the simplified algorithm, which is obtained from the proposed algorithm as follows: step 1 was removed, steps 2–4 were performed under the condition $T = 0.1\,\mathrm{s}$, the estimation of T_i at steps 5–6 was performed by the threshold method under the condition $thr = -0.0001$. The calculations were performed for the case $T_M = 4\,\mathrm{s}$, $F_s = 22050\,\mathrm{Hz}$, $T_{60} = 0.8\,\mathrm{s}$, $\mathrm{SNR}_M = 45 \cdot \mathrm{dB}$.

As can be seen, at $thr = -0.0001$ the results of the reverberation time estimation are in good agreement with the given value $T_{60} = 0.8 \cdot s$. However, experiments show that the results of the calculations significantly depend on the choice of thr value. For example, at $thr = -0.0001$ the estimate of T_{30} is overestimated on average by 0.04 s, and at $thr = -0.0001$ the estimate of T_{30} is underestimated on average by 0.17 s. The results of the calculations are also sensitive to the values of T, SNR_M and T_{60}, but our special attention to the choice of thr is due to the fact that this parameter is key in the proposed algorithm. The problem of choosing thr can be circumvented to some extent if $thr = 0$ is set, fixing the change in the sign of the derivative $dD_1(t)/dt$.

Since this approach should lead to a systematic overestimation of T_i estimates and, as a consequence, an overestimation of reverberation time estimates, it is appropriate to consider the following ways to eliminate this effect:

- adjust the estimate of T_i, reducing it by the value T;
- subtract the mean square of the background noise $n(t)$ from the function $h^2(t)$ before calculating the envelope (2);
- perform both methods together.

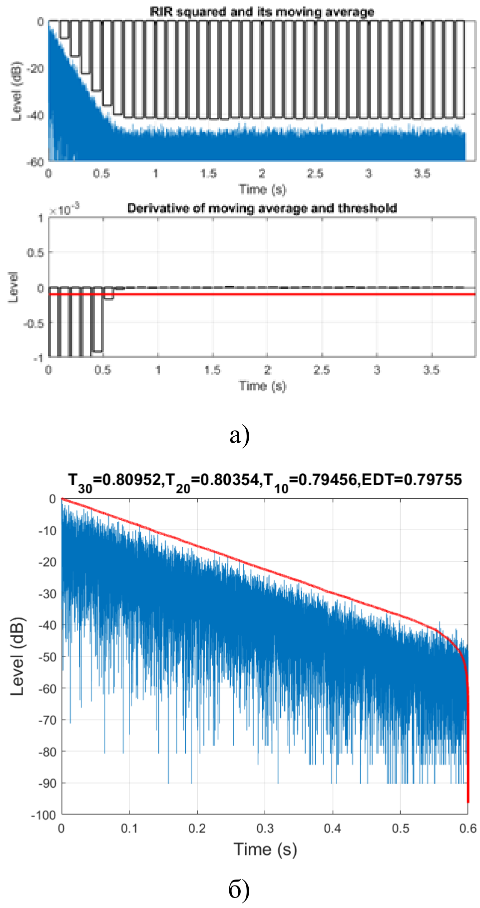


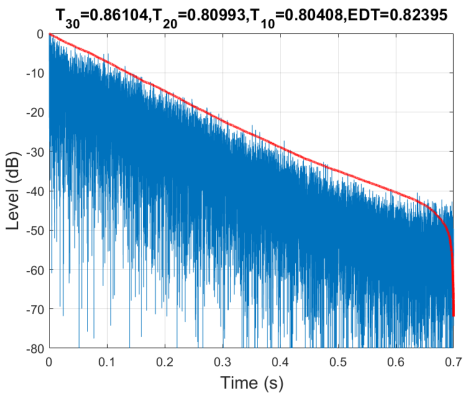
Fig. 2. Reverberation time estimation for $thr = -0.0001$: calculation of T_i (a), calculation results (b)

Experiments show that the first method is effective, but the results of the reverberation time estimation are sensitive to both the form of the signal $h(t)$ and the value of T. Figure 3 shows an example for T_{30} when for the same $h(t)$ realization satisfactory results were obtained at $T = 0.5\,\mathrm{s}$ (Fig. 3a) while at $T = 0.1\,\mathrm{s}$ the results are unsatisfactory (Fig. 3b). In this case, the calculations were performed for $T_M = 4\,\mathrm{s}$, $F_s = 22050\,\mathrm{Hz}$, $T_{60} = 0.8\,\mathrm{s}$, $\mathrm{SNR}_M = 45 \cdot \mathrm{dB}$.

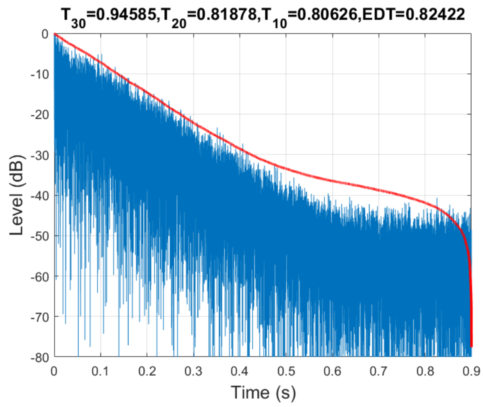
The results of calculations according to the second method are shown in Fig. 4. In this case, in accordance with the recommendations of the ISO 3382 standard, a variable duration of the parameter T was used:

$$T = \begin{cases} T_{60}/15, & \text{if} \quad T_{60} \quad \text{was predicted}, \\ 0.05, & \text{if} \quad T_{60} \quad \text{was not predicted}. \end{cases}$$

As can be seen, subtracting the average background noise level helped increase the accuracy of the results, the relative error of which does not exceed 10%.



a)



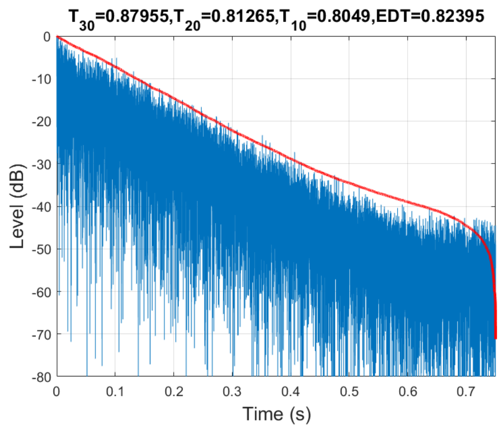
b)

Fig. 3. Correction of the estimate T_i: $T = 0.05\text{s}$ (a); $T = 0.1s$ (b)

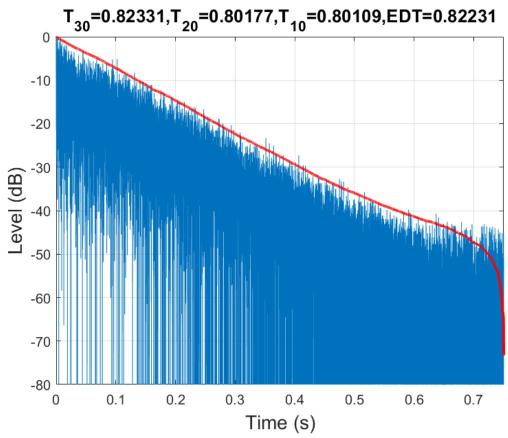
It should be noted that the results shown in Figs 3–5 were obtained for the same $h(t)$ realization. At the same time, among the different realizations of the signal $h(t)$, one was chosen for which the error in estimating the reverberation time was close to the maximum error, which allows one to draw certain conclusions regarding the capabilities of the applied algorithm. At the same time, it is advisable to assess the accuracy of the calculations using a statistical approach, estimating the bias Δ, the standard deviation σ and the total error $\prod = \sqrt{\Delta^2 + \sigma^2}$ of the EDT, T_{10}, T_{20}, T_{30} estimates. This approach is commonly used, since it characterizes the measurement situation "on average".

When implementing the statistical approach, the following actions were performed:

- 50 realizations of the model (3) were generated with a reverberation time T_{60} varying within 0.4–1.2 s with a step of 0.2 s;
- each realization was filtered with a seven octave filters with central frequencies $f_0 = 125,\ 250,\ 500,\ 1000,\ 2000,\ 4000,\ 8000$ Hz;
- the errors of the EDT, T_{10}, T_{20}, T_{30} estimates were calculated for different f_0 and T_{60};
- the actions in the previous points were performed for $\text{SNR}_M = 40,\ 45$ and 50 dB.



a)



b)

Fig. 4. Results of subtracting the average background noise level: before (a) and after (b) subtraction

The results of testing the third method shown in Fig. 5 indicate the appropriateness of combining the first two methods.

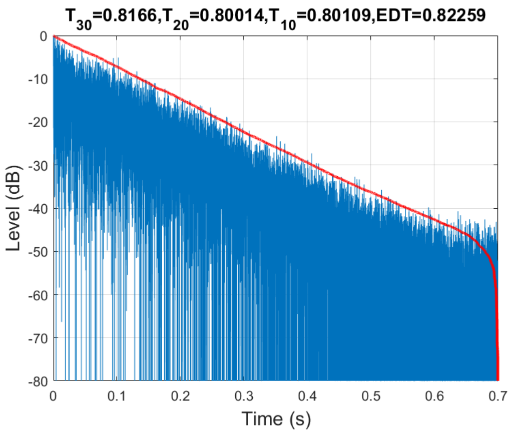


Fig. 5. Calculation results using the third method

The estimates of the relative measurement errors of EDT, T_{10}, T_{20}, T_{30} estimates obtained for $\text{SNR}_M = 45$ dB. are given in the Tables I – IV.

TABLE I. RELATIVE ERRORS FOR $f_0 = 125$ Hz

$\tilde{T}_{60}$	Error, %	T_{60} s				
		0.4	**0.6**	**0.8**	**1.0**	**1.2**
EDT	Δ	13.1	8.9	1.9	5.2	1.4
	σ	22.0	17.0	17.7	17.2	13.0
	Π	25.7	19.2	17.8	17.9	13.1
T_{10}	Δ	9.4	7.8	3.9	1.1	3.8
	σ	25.4	19.4	18.8	16.8	14.5
	Π	27.1	20.9	19.2	16.8	15.0
T_{20}	Δ	47.2	19.7	6.7	3.0	1.1
	σ	47.5	21.3	12.8	14.3	11.4
	Π	67.0	29.0	14.4	14.6	11.4
T_{30}	Δ	92.2	44.2	15.7	5.4	-1.1
	σ	69.7	39.7	26.3	18.1	14.6
	Π	115.6	59.4	30.6	18.9	14.7

TABLE II. RELATIVE ERRORS FOR $f_0 = 500$ Hz

$\tilde{T}_{60}$	Error, %	T_{60} s				
		0.4			**0.4**	
EDT	Δ	1.0	2.3	0.6	0.6	-0.1
	σ	13.7	9.2	8.1	7.3	7.7
	Π	13.7	9.4	8.1	7.3	7.7
T_{10}	Δ	1.9	2.3	0.4	0.2	-0.2
	σ	14.4	11.2	8.8	8.0	7.0
	Π	14.5	11.4	8.8	8.0	7.0
T_{20}	Δ	8.4	3.9	2.0	1.7	1.0
	σ	10.8	6.4	4.3	5.0	3.9
	Π	13.6	7.5	4.7	5.3	4.0
T_{30}	Δ	51.6	16.7	11.1	4.7	3.4
	σ	53.0	18.0	12.2	7.4	5.2
	Π	74.0	24.6	16.5	8.8	6.2

TABLE III. RELATIVE ERRORS FOR $f_0 = 8$ kHz

$\tilde{T}_{60}$	Error, %	T_{60} s				
		0.4			**0.4**	
EDT	Δ	0.4	-0.5	-0.6	-0.4	-0.1
	σ	2.9	2.2	2.2	2.1	2.3
	Π	2.9	2.3	2.3	2.1	2.3
T_{10}	Δ	-0.1	-0.1	0.1	0.3	0.7
	σ	3.3	2.8	2.6	2.1	1.8
	Π	3.3	2.8	2.6	2.1	2.0
T_{20}	Δ	0.7	0.5	0.6	0.8	0.9
	σ	1.9	1.7	1.2	1.1	1.1
	Π	2.0	1.8	1.3	1.4	1.4
T_{30}	Δ	5.4	3.9	3.6	3.4	3.2
	σ	7.0	3.8	2.7	1.8	1.9
	Π	8.9	5.4	4.5	3.9	3.7

Note that initially it was planned to use 100 realizations of model (3) to estimate these errors. However, in the end it was necessary use 50 realizations due to the appearance of anomalous measurement results for filtered signals at frequencies of 125, 250 and 500 Hz.

TABLE IV. RELATIVE ERRORS FOR WIDE BAND

$\tilde{T}_{60}$	Error. %	T_{60},s				
		0.4	**0.6**	**0.8**	**1.0**	**1.2**
EDT	Δ	0.0	-0.5	0.0	0.0	-0.1
	σ	1.7	1.8	1.7	1.3	1.3
	Π	1.7	1.9	1.7	1.3	1.3
T_{10}	Δ	0.3	0.1	0.1	0.2	0.5
	σ	2.3	2.0	1.6	1.5	1.2
	Π	2.3	2.0	1.6	1.5	1.3
T_{20}	Δ	1.0	0.7	0.8	0.6	0.8
	σ	1.3	1.1	0.8	0.8	0.6
	Π	1.7	1.3	1.1	1.0	1.0
T_{30}	Δ	7.2	3.7	4.2	3.5	3.3
	σ	10.9	3.8	2.6	2.0	1.5
	Π	13.0	5.3	4.9	4.1	3.6

For example, for the case $T_{60} = 0.4$ s and $f_0 = 125$ Hz, the total errors of the EDT, T_{10}, T_{20}, T_{30} estimates are 26%, 27%, 67% and 116%, respectively. With an increase in the frequency band, the accuracy increases. For $f_0 = 500$ Hz, the total errors decrease to 14%, 15%, 14% and 74%, and for $f_0 = 8$ kHz we have 3%, 3%, 2% and 9%. For the full frequency band, the errors are 2%, 2%, 2%, and 13%.

With an increase in T_{60}, the accuracy of the measurements improves. Let us limit ourselves to considering the values T_{60} = 0.8 s and T_{60}= 1.2 s, within which most of the T_{60} values of university classrooms are located.

For T_{60} = 0.8 s and f_0 = 125 Hz, the total errors of the estimates of EDT, T_{10}, T_{20}, T_{30} are 18%, 19%, 14% and 31%, respectively. For f_0 = 500 Hz, these values decrease to 8%, 9%, 5% and 17%, respectively, and for f_0 = 8 kHz we have 2%, 3%, 1% and 4%. For the full frequency band, the errors are 2%, 2%, 1% and 5%, respectively.

For T_{60} = 1.2 s and f_0 = 125 Hz, the total errors are 13%, 15%, 11% and 15%, respectively. For f_0 = 500 Hz the errors decrease to 8%, 7%, 4% and 6%, and for f_0 = 8 kHz we have 2%, 2%, 1% and 4%. For the full frequency band the errors are 1%, 1%, 1%, and 4%, respectively.

Let us present a number of graphs that clearly confirm the presence of a significant difference in the accuracy of reverberation time measurements in different frequency ranges. Fig. 6 shows graphs for the cases f_0 = 125 Hz, 500 Hz and 8 kHz. Each graph shows the dependences of the total error Π for the EDT, T_{10}, T_{20}, T_{30} estimates.

As can be seen, in the case of f_0 = 125 Hz (Fig. 6a) the measurement accuracy is unsatisfactory, and only for cases of $f_0 \geq 500$ Hz the total error does not exceed 10% (Fig. 6b, c).

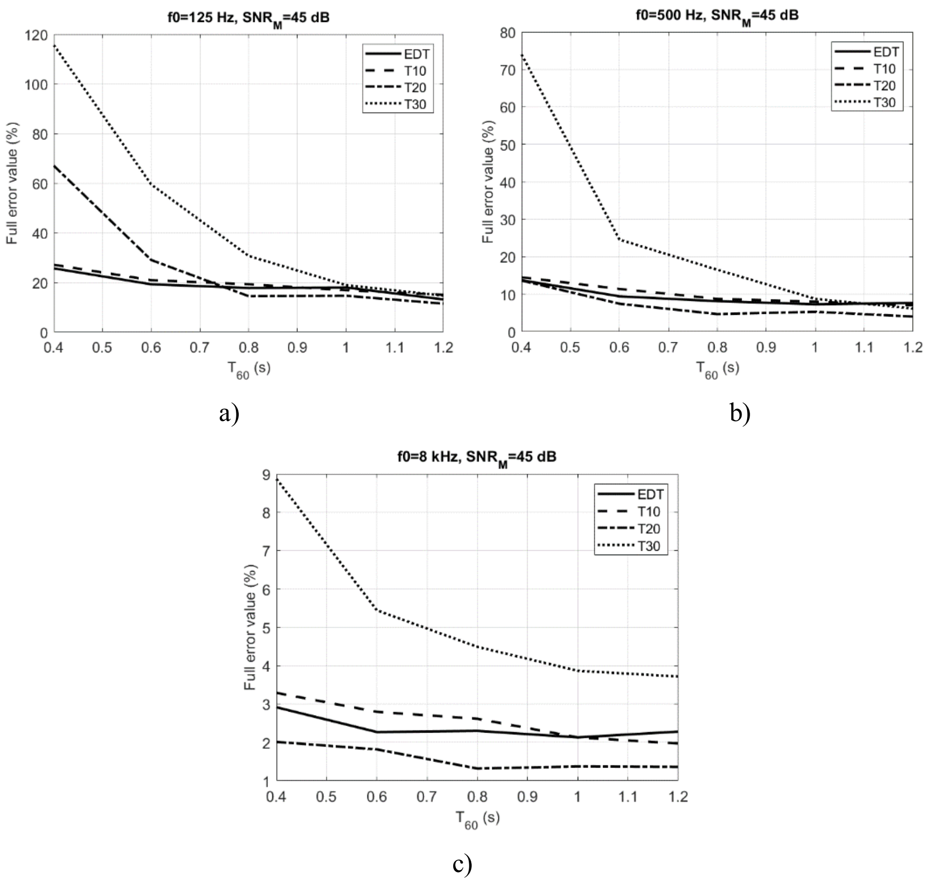


Fig. 6. Total errors of the EDT, T_{10}, T_{20}, T_{30} estimates for f_0 = 125 Hz (a), 500 Hz (b), 8 kHz (c)

The reverberation time estimates obtained for the full frequency band (80 Hz – 11 kHz) are acceptable in terms of accuracy and can be used for acoustic certification of premises in the reverberation time range of 0.5–1.2 s (Fig. 7). Note that at $\mathrm{SNR}_M = 40$ dB the T_{30} estimates turned out to be unacceptable due to an error of 30–40% (Fig. 7a).

The specified algorithm can also be used in the estimation of the speech transmission index (STI) by the indirect modulation method, where the reverberation time is measured in octave frequency bands [19], [20]. It should be borne in mind that the total measurement error at f_0 = 125 Hz for the range of values T_{60} = 0.7–1.2 s reaches 20%, decreasing to 8–10% at f_0 = 500 Hz and to 1.5–2% at f_0 = 8 kHz.

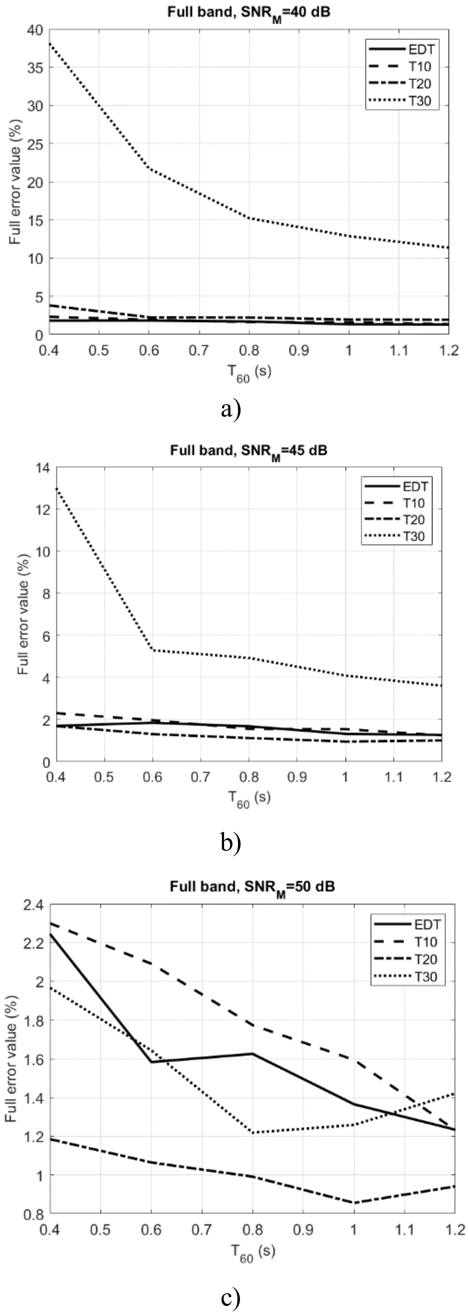


Fig. 7. Total errors of the EDT, T_{10}, T_{20}, T_{30} estimates in the full frequency band for SNR_M = 40 dB (a), 45 dB (b) and 50 dB (b)

Comparing the estimates of the bias and standard deviation (see Tables I – IV), it can be seen that when measuring EDT and T_{10} estimates, the bias is noticeably smaller than the standard deviation. However, for T_{20}, T_{30} estimates, the bias increases and becomes close to the standard deviation.

VI. Conclusion

In this paper, statistical testing of the proposed algorithm for estimating reverberation time was performed. The dependences of the errors of the EDT, T_{10}, T_{20}, T_{30} estimates on the reverberation time T_{60} were obtained for seven octave frequency bands, as well as for the full frequency band 80 Hz – 11 kHz. Given the acceptable accuracy of T_{60} estimating, the proposed algorithm can be used for acoustic examination of university auditoriums, in particular, for estimating the speech transmission index by the indirect modulation method. When measuring in a wide frequency band 80 Hz – 11 kHz, the total measurement error does not exceed 10% provided that the SNR is 45 dB. When measuring in octave frequency bands for the range of values $T_{60} = 0.7 - 1.2$ s, the total measurement error at $f_0 = 125$ Hz reaches 20%, decreasing to 8–10% at f_0 = 500 Hz and to 1.5–2% at f_0 = 8 kHz.

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А. М. Продеус, А. С. Найда. Точність алгоритму оцінки часу реверберації

Задання оцінки часу реверберації є актуальним для акустичної експертизи приміщень як цивільного призначення (дитячі садки, навчальні заклади, концертні зали тощо), так і військового призначення (центри керування). Вимірювання часу реверберації зазвичай виконують методом оберненого інтегрування оцінки імпульсної характеристики приміщення. Однак при цьому через існування фонової завади виникає проблема вибору моменту часу (час відсікання), з якого потрібно починати інтегрування. В даній статті досліджено точність алгоритму обчислення часу реверберації, де час відсікання визначається як момент наближення до нуля похідної сигналу на виході системи «квадратор – ковзний інтегратор». Отримано оцінки зміщення, стандартного відхилення та повної похибки для значень відношення сигнал-шум та часу реверберації, типових для навчальних аудиторій. При відношенні сигнал-шум 45 дБ, для вимірювань в широкий смузі частот 80 Гц – 11кГц, повна відносна похибка оцінювання часу реверберації не перевищує 6% для значень часу реверберації 0,6–1,2 с. При вимірюваннях часу реверберації в октавних смугах частот, похибка сягає 20% для смуги частот в

околиці 125 Гц, знижується до 10% для смуги частот в околиці 500 Гц, й не перевищує 3% для смуги частот в околиці 8 кГц.
Ключові слова: час реверберації; час відсікання; імпульсна характеристика приміщення; квадратор – ковзний інтегратор; метод зворотного інтегрування.

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