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REVERBERATION TIME ERRORS FROM FREQUENCY DOMAIN FILTERING

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Abstract—*Information about the frequency dependence of reverberation time is essential for addressing several tasks, including mitigating the impact of reverberation on speech quality and intelligibility, as well as assessing intelligibility using the indirect modulation method. To obtain this information, the room impulse response must be filtered using a bank of octave or one-third-octave filters. This paper analyzes the influence of frequency bandwidth and the shape of the filter's amplitude-frequency response on the bias of T_{20}, T_{30}, EDT, and T_{10} estimates of the T_{60} reverberation time. The analysis assumes that filtering is implemented in the spectral domain by zeroing the spectral components of the RIR outside the desired passband, and that the filter's amplitude-frequency response has the shape of a Tukey window. The results show that the use of filters with a rectangular amplitude-frequency response (Tukey window with parameter r = 0) is undesirable, as it leads to significant bias in the T_{20}, T_{30}, EDT, and T_{10} estimates. This bias can reach 60–100% for reverberation times in the range of T_{60} = 0.4–1.2 s. Using filters with a Tukey window shape and r = 1 reduces the bias to no more than 4% when filtering room impulse responses with octave filters at center frequencies $f_0 \geq 125$ Hz. For one-third-octave filters with $f_0 \geq 25$ Hz, a similar level of bias is observed for T_{20} and T_{30} estimates. For EDT and T_{10} estimates, a bias of no more than 4% is achieved within the T_{60} = 0.6–1.2 s range.*

Keywords—Reverberation time; frequency dependence; spectral domain filtering; amplitude-frequency response; Tukey window; bias of estimate.

I. INTRODUCTION

Before using voice technologies for voice-controlled unmanned aerial vehicles (UAVs), it is advisable to account for the effects of noise and reverberation by first assessing their parameters. In particular, information about the frequency dependence of reverberation time is essential for tasks such as mitigating its impact on speech quality and intelligibility, as well as evaluating intelligibility using the indirect modulation method.

When measuring T_{60} reverberation time, the noise interruption method or the method of analyzing the room impulse response (RIR) $h(t)$ is used [1], [2]. The signal recorded at the output of the measuring microphone is filtered by a bandpass filter bank [3], which allows one to subsequently obtain the dependence of the reverberation time on the frequency. The envelopes $D_k(t)$ of the signals at the output of the filters $h_k(t)$, k is the frequency channel number, are practically the same for both measurement methods. Therefore, for clarity, we will refer to the method of analyzing the RIR. Also, for simplicity, the index k in $h_k(t)$ and $D_k(t)$ will be omitted in the future.

The envelope $D(t)$ of the signal $h(t)$ is obtained by one of 2 methods. According to the first method [1], [2], the signal $h(t)$ is processed using a "squarer – sliding integrator" system

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

$w(t)$ is the impulse response of the filter implementing sliding (exponential or linear) averaging; $T_{\det}$ is the effective averaging time, which should be significantly shorter than the expected reverberation time.

Today, the envelope calculation by the inverse integration method is preferred [4]

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

N is proportional to the spectral density of the noise power in the measurement frequency range. The reason for this advantage is the possibility of significantly speeding up measurements, since (2) is obtained under the condition of averaging over the ensemble of samples of the random process $h^2(t)$. This makes it possible to limit the measurement to a single session when determining the reverberation time at a specific point in the room, while the use of expression (1) leads to the need to repeat the measurement sessions with subsequent averaging of the obtained results [1], [2].

Next, the obtained envelope $D(t)$ is logarithmized, as a result of which the exponential law of decay of the envelope $D(t)$ is replaced by a linear law. The determined moments of intersection of the envelope $D(t)$ of the thresholds at the levels of minus 5 dB, minus 15 dB, minus 25 dB and minus 35 dB allow one to calculate the corresponding estimates of the reverberation time T_{10}, T_{20}, and T_{30}. To calculate the Early Decay Time (EDT), the thresholds 0 dB and minus 10 dB are used.

The practical use of (2) is complicated by the presence of background noise $n(t)$

$$h_n(k) = h(k) + n(k), \tag{3}$$

Therefore, instead of (2), another relation is usually used

$$\hat{D}_s(t) \approx N\int_t^{T_i} h_n^2(\tau)d\tau, \tag{4}$$

T_i is the so-called "truncation point", which separates the informative part of the RIR $n(t)$ from the background noise.

There are various proposals regarding the algorithm for determining the parameter T_i. In article [5], 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 of the estimate $h(t)$. However, the specified interval is too large and therefore is not suitable for practical use. In article [6] this drawback is partially eliminated and it is proposed to take into account, in addition to T_{60} and dynamic range of the estimate $h(t)$, the value of the permissible relative bias of the estimate (4) when choosing T_i.

In article [7], it is recommended to avoid finding the value of T_i at all, subtracting the mean square of the background noise from $h_n^2(t)$

$$D_s^*(t) = N\int_t^{\infty}\left[h^2(\tau) + 2h(\tau)n(\tau) + \left(n^2(\tau) - \overline{n}^2\right)\right]d\tau, \tag{5}$$

$\overline{n}^2$ is the mean square of the background noise calculated from the tail of the RIR record containing the background noise.

Finally, one of the main characteristics of the reverberation time calculation algorithm is the accuracy of the reverberation time estimate, which is characterized by the bias and standard deviation of the estimate. With bandpass filtering $h_n(k)$, the problem of ensuring the desired accuracy of reverberation time measurements is complicated by the increase in variance and bias of the estimate.

In the ISO 3382-1:2009 standard [1], [2], the results of studies of the influence of filtering on the variance of the reverberation time estimate [8], [9] are presented. These results are given in a form convenient for practical use. As for the influence of filtering on the bias of the reverberation time estimate, there is no sufficient information about the nature and extent of such influence.

The purpose of this paper is to eliminate this drawback by analyzing the influence of the shape of the AFR and the bandwidth of the filter on the bias of the reverberation time estimate.

II. Problem Statement

The conditions for obtaining reliable estimates of reverberation time when $h(t)$ is filtered using octave or one-third-octave filters that comply with the IEC 61260-1:2014 standard [3], are defined as [1]

$$\Delta f \cdot T_{60} > 16, \tag{6}$$

Δf is the filter bandwidth in Hz.

According to the standards [10], [11] recommendations, the reverberation time should satisfy the condition $T_{60} \leq 0.6 - 0.7$ s in newly constructed classrooms for people with normal hearing. A more relaxed criterion of $T_{60} \leq 1.0$ s is acceptable in renovated classrooms. For students with hearing impairments, must be performed $T_{60} \leq 0.4$ s within the frequency range $125\,\text{Hz} \leq f \leq 5\,k\text{Hz}$.

It can be seen that condition (6) is easily satisfied for classrooms of all categories when measuring the frequency dependence $T_{60}(f)$ using seven octave-band filters with central frequencies in the range $125\,\text{Hz} \leq f_0 \leq 8\,\text{kHz}$. However, condition (6) may be difficult to satisfy in the frequency range $25 \leq f \leq 200\text{Hz}$ when $T_{60}(f)$ is measured using 1/3-octave filters using filtering in the frequency domain with zeroing of spectral components outside the frequency passband. It is reasonable to assume that this drawback can be mitigated or eliminated by selecting an appropriate amplitude-frequency response shape for the bandpass filters.

III. Set up of the Study

The research was conducted by computer modeling of the RIR with a given reverberation time T_{60}, from 0.4 s to 1.2 s with a step of 0.2 s. The simulated records of the RIR were filtered with a bank of octave or 1/3-octave filters. The reverberation time estimates (parameters EDT, T_{10}, T_{20}, and T_{30}) were measured at the output of each filter. The measurements were repeated 100 times, which allowed one to calculate the bias and standard deviation of the estimates of the parameters EDT, T_{10}, T_{20}, and T_{30} with sufficient accuracy for practical use.

The computer modeling used a model of the RIR [12]

$$h(i) = g_0 v(i) e^{-\rho i T_s},\ i = 0,1,\ldots, K = T_M F_s, \qquad (7)$$

$v(i)$ is white Gaussian noise with zero mathematical expectation and unit variance; $\rho = 6.908 / T_{60}$, $T_s = 1/f_s$ is sampling period, f_s is sampling frequency, T_M is $h(t)$ duration.

Computer modeling of process samples (3) using model (7) was performed according to the expressions

$$y(i) = v(i)\sigma(i), \qquad (8)$$

$$\sigma(i) = \left(g_0^2 \cdot e^{-2\rho i T_s} + \sigma_n^2\right)^{0.5}. \qquad (9)$$

Assuming $g_0 = 1$, one can obtain

$$y(i) = v(i)\left[\exp\left(-13.8 i / \left(F_s T_{60}\right)\right) + 10^{-0.1 \cdot SNR_{\lg}}\right]^{0.5}, \qquad (10)$$
$$SNR_{\lg} = 20\lg(g_0/\sigma_n).$$

The envelope $D(t)$ was calculated according to (5), where the upper integration limit is limited by the value T_{60}.

When modeling, $SNR_{\lg} = 45$ dB was assumed, which provided the possibility of evaluating all parameters EDT, T_{10}, T_{20}, and T_{30}.

The filtering of stochastic models of the reverberation process was performed in the frequency domain, which allowed the implementation of non-recursive filters of the same order with AFRs of different shapes and symmetrical impulse response (IR). It was assumed that the AFR of the filter has the form of a Tukey window

$$H_{f0}(f) = 0.5\left\{1 + \cos\left[\frac{\pi}{r \cdot \Delta f}\left(|f| - \frac{\Delta f}{2}(1-r)\right)\right]\right\}, \qquad (11)$$

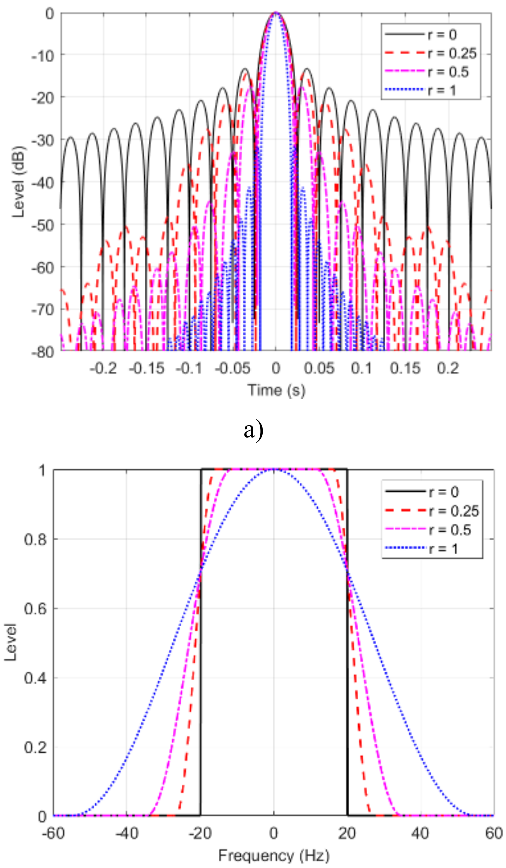
$$\frac{\Delta f}{2}(1-r) < |f| < \frac{\Delta f}{2}(1+r)$$

and

$$H_{f0}(f) = 1,\quad |f| \le \frac{\Delta f}{2}(1-r).$$

where the window shape is adjusted by changing the value of the parameter $0 \le r \le 1$.

Figure 1 shows the graphs of the IR $h_{f0}(t)$ and the AFR $H_{f0}(f)$ of filter with a bandwidth of $\Delta f = 40$ Hz at a level of –3 dB from the maximum, for $0 \le r \le 1$.



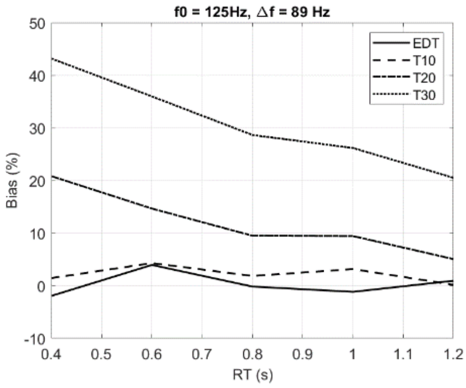
a) b)

Fig. 1. $h_{f0}(t)$ $h_{f0}(t)$ (a) and $H_{f0}(f)$ (b) for $0 \le r \le 1$

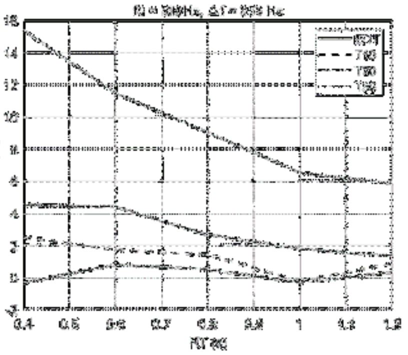
IV. Results of the Study

A. *Case of Octave Filters*

Figures 2–5 show the graphs of relative bias and standard deviation of the reverberation time estimates EDT, T_{10}, T_{20}, and T_{30} for different frequency bands of octave filters with different AFR shapes. The center frequencies 125–8000 Hz of the octave filters are usually used when measuring speech intelligibility using the indirect modulation method [13].

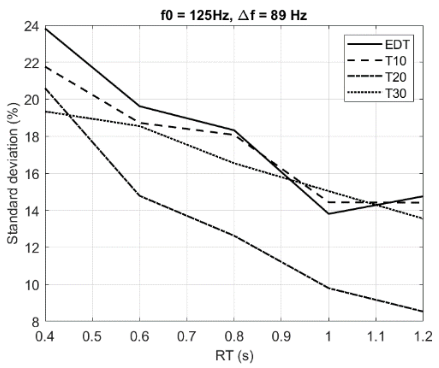


a)

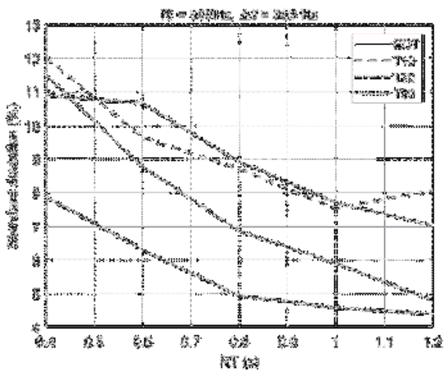


b)

Fig. 2. Bias, $r = 0$: f_0 =125 Hz (a), f_0 =500 Hz (b)

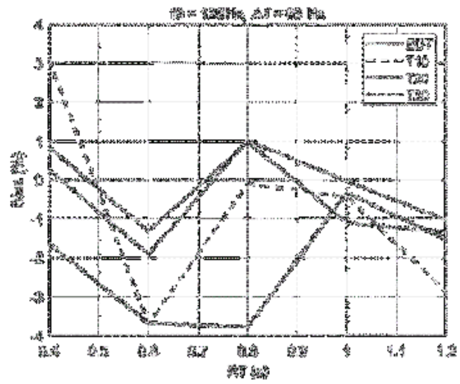


a)

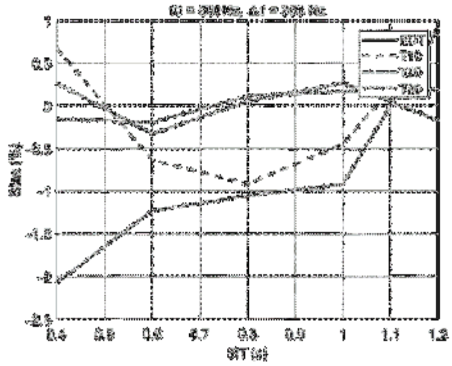


b)

Fig. 3. Standard deviation, $r = 0$, f_0 =125 Hz (a); f_0 = 500 Hz (b)

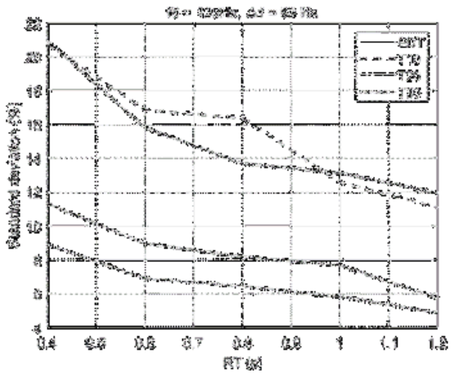


a)

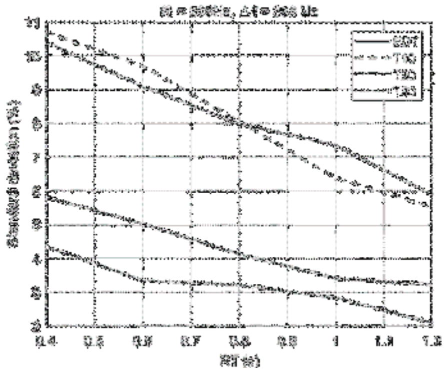


b)

Fig. 4. Bias, $r = 1$, f_0 =125 Hz (a); f_0 = 500 Hz (b)



a)



b)

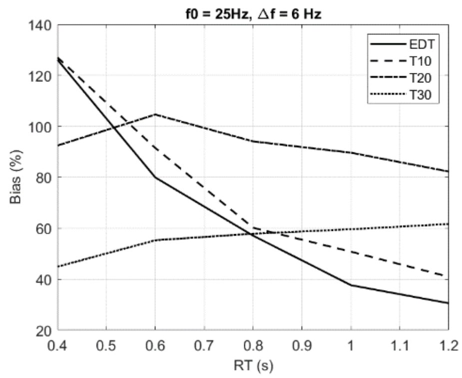
Fig. 5. Standard deviation, $r = 1$, f_0 =125 Hz (a); f_0 = 500 Hz (b)

It can be seen a decrease in the relative bias and standard deviation with Δf increasing, regardless of the shape of the filter AFR. At the same time, the shape of the filter AFR significantly affects the errors of the EDT, T_{10}, T_{20}, and T_{30} estimates, with the smallest bias occurring for filters with an AFR in the form of a Tukey window with r = 1 (Fig. 4). A significant bias of the T_{20} and T_{30} estimates is observed for filters with a rectangular frequency response (r = 0). It reaches 20–40% for Δf = 89 Hz (Fig. 2a), which may be unacceptable for further practical use.

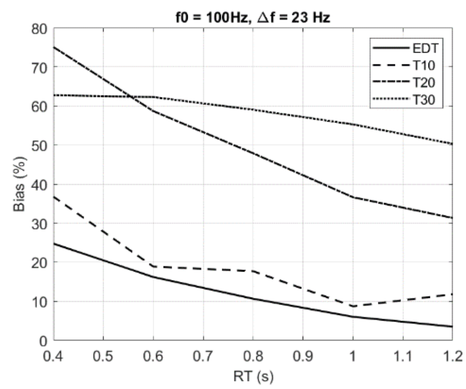
Somewhat unexpected is the fact that the use of filters with a rectangular frequency response leads to an increase, approximately 1.5–2 times, compared to the cases of $r \neq 0$, of the T_{20} and T_{30} estimates standard deviation. This phenomenon is observed for $89 \leq \Delta f \leq 353\,\text{Hz}$ (Fig. 3).

B. Case of One-Third-Octave Filters

Figures 6–9 show graphs of relative bias and standard deviation of reverberation time estimates EDT, T_{10}, T_{20}, and T_{30} for different frequency bands of 1/3-octave filters with different frequency response shapes.

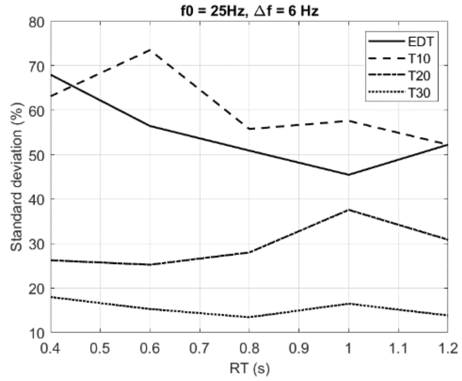


a)

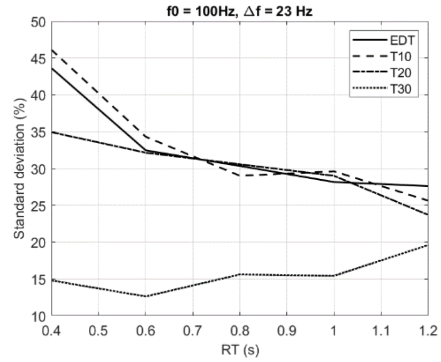


b)

Fig. 6. Bias, $r = 0, f_0 = 25$ Hz (a); $f_0 = 100$ Hz (b)

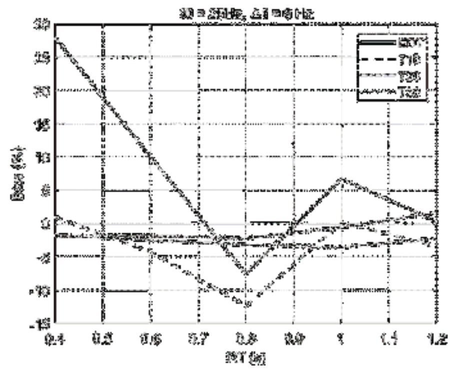


a)

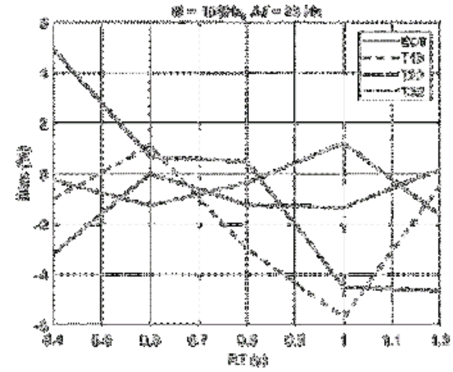


b)

Fig. 7. Standard deviation, $r = 0, f_0 = 25$ Hz (a); $f_0 = 100$ Hz (b)



a)



b)

Fig. 8. Bias, $r = 1$, $f_0 = 25$ Hz (a), $f_0 = 100$ Hz (b)

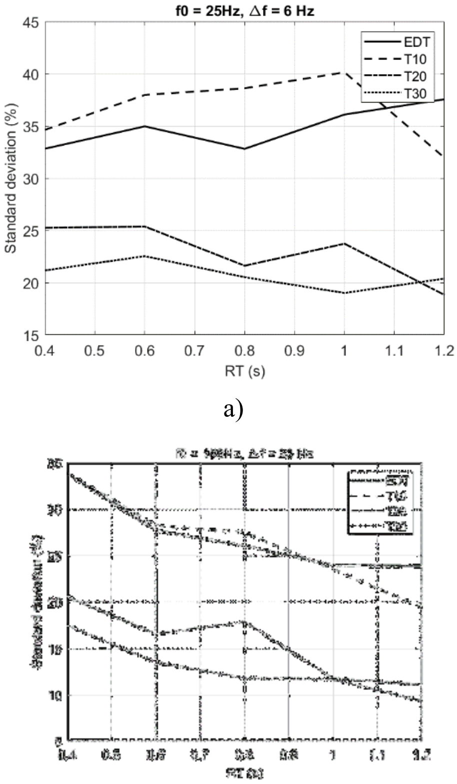


Fig. 9. Standard deviation, $r = 1$, $f_0 = 25$ Hz (a); $f_0 = 100$ Hz (b)

Unlike the case of octave filters, a much smaller frequency range is considered here. This is done because the largest bias is expected for narrow frequency bands when using filters with a rectangular frequency response. These expectations are confirmed by the behavior of the graphs in Fig. 6, where the relative bias of the T_{20} and T_{30} estimates is close to 40–100%. The smallest bias of the EDT, T_{10}, T_{20}, and T_{30} estimates also occurs for filters with a frequency response in the form of a Tukey window with $r = 1$ (Fig. 8).

As for the standard deviation of the T_{20} and T_{30} estimates, the use of 1/3-octave filters with a rectangular frequency response does not lead to such a difference between the cases $r = 0$ and $r \neq 0$ (Figs 7 and 9), as was the case for octave filters.

V. Discussion

As already noted, a feature of the performed research is the implementation of bandpass filtering of the RIR in the frequency domain. The attractiveness of such filtering lies in the possibility of easily implementing a non-recursive high-order bandpass filters by zeroing the spectral components of the RIR outside the frequency bands. Since such zeroing is equivalent to using a filters with a rectangular frequency responses, one can expect an undesirably large bias in the reverberation time estimates due to the low decay rate of the side lobes of the corresponding filter IR. An obvious way to reduce such bias is to correct the shape of the filter AFR.

At the same time, the magnitude of the bias in the reverberation time estimates should also be affected by the bandwidth of the frequency band Δf and the value of the reverberation time T_{60} (the steepness of the RIR). It should be expected that the bias will increase with a decrease in the values of Δf and T_{60}. However, quantitative estimates of such a bias, as well as the ratio of bias and standard deviation of reverberation time estimates, remained unknown until recently. Analysis of the research results presented in Figs 2–9 allows one to eliminate these shortcomings.

A. Case of Octave Filters

In the case of octave filters (Figs 2–5), it can be seen that the use of filters with a rectangular frequency response ($r = 0$ for the Tukey window) may be undesirable when estimating the reverberation time, since at $f_0 = 125$ Hz, $\Delta f = 89$ Hz the bias of the estimates T_{20} and T_{30} reaches 20–40% and significantly exceeds the standard deviation (Figs 2a and 3a). With an increase in Δf to 353 Hz, such an excess disappears (Figs 2b and 3b). It can be shown that in a wide frequency band ($f_0 = 8$ kHz, $\Delta f = 5657$ Hz), the above-mentioned excess is absent.

The use of filters with non-rectangular frequency response ($r = 1$, Fig. 4) allows one to reduce the bias of the estimates T_{20} and T_{30} to 1–2% at $f_0 = 125$ Hz, $\Delta f = 89$ Hz. For larger values of f_0 and Δf, the bias decreases even more significantly.

The standard deviation of the estimates T_{20} and T_{30} practically does not depend on the shape of the frequency response of the filters, although in the case of $r = 0$, $f_0 = 125$ Hz, $\Delta f = 89$ Hz the standard deviation of the estimates T_{20} and T_{30} reaches 10–20% (Fig. 3a), which is almost twice as much compared to the case $r = 1$ (Figs 5a and 7a).

As for the estimates of the EDT and T_{10} parameters, their bias in the case of octave filters does not exceed 4% for the range of values $f_0 = 125$–8000 Hz even when using filters with a rectangular frequency response (Fig. 2). The use of filters with a frequency response in the form of a Tukey window, $r = 1$, has practically no effect on the magnitude of the bias of the EDT and T_{10} parameters (Fig. 4).

B. Case of One-Third-Octave Filters

In the case of 1/3-octave filters (Figs 6–9), the use of filters with a rectangular frequency response is even more undesirable when estimating the reverberation time, since the bias of the estimates T_{20} and T_{30} is 2–6 times higher than the standard deviation and reaches 60–100% (Figs 6a and 7a). Obviously, such an excess is more significant than in the case of octave filters, which can be explained by a much narrower frequency band (6 Hz for a 1/3-octave filter with f_0 = 25 Hz versus 89 Hz for an octave filter with f_0 = 125 Hz). With an increase in f_0 and Δf, this situation gradually improves (Figs. 6b and 7b).

In the case of a Tukey window, *r* = 1, the bias of the T_{20} and T_{30} estimates does not exceed 4% even in the most problematic case f_0 = 25 Hz, Δf = 6 Hz (Fig. 8a).

The situation with the accuracy of the EDT and T_{10} estimates in the case of 1/3-octave filters is different (Figs 6 and 8). For example, the use of filters with a rectangular frequency response leads to a bias of 120% at f_0 = 25 Hz, Δf = 6 Hz for T_{60} = 0.4 s (Fig. 6a). Even for filters with the frequency response in the form of a Tukey window, *r* = 1, the bias of the EDT and T_{10} estimates for the specified values of f_0, Δf and T_{60} remains significant and reaches 30% (Fig. 8a). However, for $T_{60} \geq 0.6$ s, when using filters with the frequency response in the form of a Tukey window, *r* = 1, the bias of the EDT and T_{10} estimates does not exceed 10% and is noticeably smaller than the standard deviation of 35% (Fig. 9).

When measuring speech transmission index (STI), the modulation transfer function (MTF) can be calculated for any combination of SNR_k and T_{60k} [13]

$$m_{k,i} = \left[1 + \left(2\pi F_i \frac{T_{60k}}{13.8}\right)^2\right]^{-\frac{1}{2}} \left[1 + 10^{-\frac{SNR_k}{10}}\right]^{-1},$$

F_i is modulation frequency. Therefore, in the future, it is advisable to investigate the impact of the accuracy of T_{60} estimates on the accuracy of STI estimates.

VI. Conclusion

The obtained results indicate a significant influence of the shape and the bandwidth of the filter frequency response on the bias of the T_{20}, T_{30}, EDT, and T_{10} estimates. The use of filters with an AFR in the form of a Tukey window, *r* = 1, allows to achieve a bias of no more than 4% when filtering the RIR with octave filters with f_0 ≥ 125 Hz for the considered range of values of T_{60} = 0.4–1.2 s. When filtering the RIR with 1/3-octave filters with f_0 ≥ 25 Hz, it is possible to ensure a bias of the estimates of T_{20} and T_{30} of no more than 4% for the range of values of T_{60} = 0.4–1.2 s. A bias of the estimates of EDT and T_{10} of no more than 4% can be ensured for a smaller range of values of T_{60} = 0.6–1.2 s.

In the future, it is appropriate to assess the degree of positive influence of the recommendations obtained in this article on the accuracy of the algorithms for estimating the reverberation time, considered in [14], [15], as well as the accuracy of STI estimates.

References

[1] ISO 3382-1:2009 Acoustics – Measurement of room acoustic parameters – Part 1: Performance spaces, Geneva: International Organization for Standardization, 2009.

[2] ISO 3382-2:2008 Acoustics – Measurement of room acoustic parameters – Part 2: Reverberation time in ordinary rooms, Geneva: International Organization for Standardization, 2008.

[3] IEC 61260-1-2014 Electroacoustics – Octave-band and fractional-octaveband filters, European Committee for Electrotechnical Standardization, 2014.

[4] M. Schroeder, “New method of measuring reverberation time,” *J. Acoust. Soc. Amer.* 37, 1965, pp. 409–412. https://doi.org/10.1121/1.1909343

[5] R. Kurer and U. Kurze, "Integrationsverfahren zur Nachhallauswertung," ("Integration Method for the Evaluation of Reverberation Measurement"), *Acustica*, vol. 19 (1967/68), pp. 313–322.

[6] K. Bordlund, "On the use of the integrated impulse response method for laboratory reverberation measurements," *Journal of Sound and Vibration*, 56(3), 1978, pp. 341–362. https://doi.org/10.1016/S0022-460X(78)80152-7

[7] W. Chu, "Comparison of Reverberation Measurements Using Schroeder's Impulse Method and Decay-Curve Averaging Method," *J. Acoust. Soc. Am.*, vol. 63, 1978, pp. 1444–1450. https://doi.org/10.1121/1.381889

[8] J. Davy, “The variance of decay rates in reverberation rooms,” *Acustica*, vol.43, no.1, pp. 51–56, 1979.

[9] J. Davy, "The variance of impulse decays," *Acustica*, vol. 44, no.1, pp. 51–56. 1980. https://doi.org/10.1109/ACOUSTICS.2018.8502277.

[10] ANSI/ASA S12.60-2010/Part 1 American National Standard Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools, Part 1: Permanent Schools.

[11] Acoustic design of schools: performance standards. *Building bulletin 93*. UK Department for Education, 2015. Availabe in: https://assets.publishing.service.gov.uk/media/5a8170d3e5274a2e8ab54012/BB93_February_2015.pdf

[12] H. Löllmann, "Estimation of the Reverberation Time in Noisy Environments," *In Proceedings of the International Workshop on Acoustic Echo and Noise Control (IWAENC)*. Seattle, US, 2008. Available in: https://www.iwaenc.org/proceedings/2008/contents/papers/9033.pdf

[13] H. Steeneken and T. Houtgast, "Basics of the STI measuring method," *in Past, present and future of the Speech Transmission Index*. Soesterberg: TNO Human Factors, 2002.

[14] A. Prodeus and A. Naida, "A Two-Stage Algorithm for Determining the Truncation Time and Reverberation Time," *2024 IEEE 7th International Conference Actual Problems of Unmanned Aerial Vehicles Developments (APUAVD)*, October 22-24, 2024, Kyiv, Ukraine. https://doi.org/10.1109/APUAVD64488.2024.10765903

[15] A. Prodeus and A. Naida, "Reverberation Time Estimation Algorithm Accuracy," *Electronics and Control Systems*, 2025. no. 1(83), pp. 9–17. https://doi.org/10.18372/1990-5548.83.19362

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А. М. Продеус, А. С. Найда, М. В. Дідковська. Похибки оцінок часу реверберації, спричинені фільтрацією в частотній області

Інформацію про залежність часу реверберації від частоти потрібно мати при розв'язанні низки завдань, серед яких, зокрема, нейтралізація впливу реверберації на якість та розбірливість мовлення, оцінка розбірливості мовлення непрямим варіантом модуляційного методу. Для отримання такої інформації запис імпульсної характеристики (ІХ) приміщення треба піддати фільтрації, використовуючи гребінку октавних або 1/3-октавних фільтрів. В даній статті проаналізовано вплив ширини смуги частот та форми амплітудно-частотної характеристики фільтру на зміщення оцінок T_{20}, T_{30}, EDT та T_{10} часу реверберації T_{60}. Аналіз виконано за припущень, що фільтрація реалізується в спектральній області шляхом обнуління спектральних складових сигналу поза межами смуги частот, а амплітудно-частотна характеристика фільтра має форму вікна Tukey (вікно Tukey window також є відомим як вікно косинусної форми). Результати аналізу свідчать, що використання фільтрів із АЧХ прямокутної форми (параметр $r = 0$ вікна Tukey) є небажаним, оскільки призводить до зміщення оцінок T_{20}, T_{30}, EDT та T_{10}, що може сягати 60–100% для часу реверберації T_{60} = 0.4–1.2 с. Використання фільтрів із АЧХ у формі вікна Tukey із параметром $r = 1$ дозволяє забезпечити зміщення не більше 4% при фільтрації ІХ октавними фільтрами із центральною частотою $f_0 \geq 125$ Гц. При фільтрації ІХ 1/3-октавними фільтрами із $f_0 \geq 25$ Гц такого ж зміщення вдається досягти для оцінок T_{20}, T_{30}. Для оцінок EDT та T_{10} зміщення не більше 4% досягається в діапазоні T_{60} = 0.6–1.2 с.

Ключові слова: час реверберації; частотна залежність; фільтрація в спектральній області; амплітудно-частотна характеристика; вікно Т'юкі; зміщення оцінки.

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