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Influence of Filter Characteristics on the Bias of Reverberation Time Estimation

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ABSTRACT Bandpass filters with a rectangular amplitude-frequency response (AFR) are relatively easy to implement in the frequency domain by zeroing out spectral components outside the desired passband. An advantage of this approach is the ease of constructing high-order non-recursive filters with a linear phase response. However, as demonstrated in this paper, this benefit comes at the cost of significant bias in reverberation time (RT) estimates, particularly in cases of narrow bandwidth and short reverberation time. It can be assumed that this drawback may be largely eliminated by using filters with a non-rectangular AFR, which is simple to implement in practice. However, the validity of this assumption remained untested until recently. In this paper, the influence of the filter's AFR shape and bandwidth on the bias in RT estimation is analyzed. It is shown that the bias of RT estimates based on T20 and T30 ranges from 60% to 100% when one-third-octave filters with a rectangular AFR are used in the frequency range of 25–200 Hz. When a Tukey window is used as the filter AFR, the bias can be reduced to 4%. Similar results were obtained for Early Decay Time and T10 estimates of RT.

KEYWORDS reverberation time estimation; bias analysis; room impulse response; amplitude-frequency response; filter bandwidth.

I. INTRODUCTION

Reverberation time is one of the most important parameters characterizing the acoustic properties of a room [1, 2]. Reverberation time estimates are used to analyze speech intelligibility and the overall acoustic perception in indoor environments [3–9].

To measure the reverberation time, either the interrupted noise method or the room impulse response (RIR) $h(t)$ analysis method is used. In both cases, the envelopes of the signals recorded at the output of the measurement microphone are nearly identical. For clarity, we will assume in this paper that the reverberation time is determined using the impulse response analysis method.

The envelope $D(t)$ of the signal $h(t)$ is obtained using one of the two methods [10–12]. According to the first method, the signal $h(t)$ is processed by a "squaring detector – sliding integrator" system as follows:

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

where $w(t)$ is the impulse response of the filter implementing sliding (exponential or linear) averaging, and T_{det} is the effective averaging time, which should be an order of magnitude smaller than the expected reverberation time [1].

Currently, preference is given to computing the envelope using the backward integration method

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

where N is proportional to the power spectral density of noise within the measurement frequency range. The main advantage of this approach lies in the potential for significant acceleration of the measurement process, since equation (2) is derived under the condition of ensemble averaging over samples of the random process $h^2(t)$. As a result, when measuring the reverberation time at a specific point in the room, it is sufficient to perform only a single measurement session, whereas the use of equation (1) necessitates repeated measurement sessions followed by averaging of the obtained results [1, 2].

A well-known issue associated with the use of equation (2) is the presence of background interference

$$h_n(k) = h(k) + n(k),$$

which forces one to use another relation instead of (2)

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

where T_i denotes the boundary between the informative part of the room impulse response $h(t)$ and background noise. The issue of selecting the optimal value for parameter T_i is quite complex, and has been the subject of numerous studies [13–16].

Regardless of the computational approach, the envelope obtained via equation (1) or (2) is typically subjected to logarithmic transformation, resulting in the conversion of the exponential decay law of the envelope of $h(t)$ into a linear law. The moments at which the envelope crosses thresholds at – 5 dB, –15 dB, –25 dB, and –35 dB are then identified, allowing for the estimation of the corresponding reverberation time estimate values T_{10}, T_{20}, T_{30}. To calculate Early Decay Time (EDT), the thresholds of 0 dB and –10 dB are used.

To obtain information on the frequency dependence of the reverberation time, the signal $h(t)$ is filtered using octave or one-third-octave filters that comply with the IEC 61260-1:2014 standard [1-3].

In [17], the bias between Sabine-predicted and measured RT values specifically across octave bands at 500, 1,000 and 2,000 Hz was investigated. Systematic underestimation of RT at mid-bands (about −24% at 500 Hz and up to −36% at 1–2 kHz) was found. A previous correction to the sound absorption coefficient of the lining materials declared by the manufacturer was proposed, making use of an empirical correction that was achieved from in situ experimental results and through geometrical room acoustic modelling.

Known and novel model-based approaches for the estimation of the subband RT were investigated in the technical report [18]. This document was focused explicitly on sub-band RT bias, evaluating model-based approaches to estimate RT per octave or finer bands. Average absolute errors per sub-band (about 0.04 s for higher sub-bands and about 0.10 s overall) and band-dependent estimation errors were determined. A recently presented approach to estimate the subband RT by extrapolating the RTs for the higher subband from the estimates of the lower subbands [19] has been investigated and compared with the related approaches presented in [20]. It turned out that the model described in [20] achieves the lowest average error per subband for all the tested room impulse responses.

Studies on the influence of filtering on the variance of the reverberation time estimate are presented in [10, 11]. These results are included in the ISO 3382-1:2009 standard [1] in a form convenient for practical application. However, regarding the influence of filtering on the bias of the reverberation time estimate, the standard [1] provides no information concerning the nature or extent of such bias.

The objective of this paper is to remove this restriction by analyzing the influence of the AFR shape and bandwidth on the bias in RT estimation. In this study, bandpass filtering with the AFR implemented in the frequency domain by suppressing spectral components outside the desired passband is considered. An advantage of this approach is the ease of constructing high-order non-recursive filters with a linear phase response. However, this benefit comes at the cost of significant bias when filters with a rectangular AFR are used, particularly in cases of narrow bandwidths and short reverberation times. It can be assumed that this drawback may be largely eliminated by using filters with a non-rectangular AFR, as such a response is simple to implement in practice. However, the validity of this assumption remained untested until recently.

II. PROBLEM STATEMENT

The standard [1] states that at low reverberation time values, the decay curve may be distorted by the filter and the detector. Therefore, the lower bounds of result reliability are determined according to specific rules

$$\Delta f \cdot T_{60} > 16, \tag{3}$$

$$T_{60} > 2T_{det}, \tag{4}$$

where Δf is the filter bandwidth in Hz, T_{60} is the measured reverberation time in seconds, and T_{det} is the time constant of the averaging detector in seconds.

According to the recommendations outlined in standards [4] and [5], the reverberation time should satisfy the condition $T_{60} \leq 0.6 - 0.7$ s in newly constructed unoccupied classrooms for students with normal hearing. In renovated unoccupied classrooms, a more relaxed criterion of $T_{60} \leq 1.0$ s is acceptable. For students with hearing impairments, the acoustic requirements are more stringent, specifically $T_{60} \leq 0.4$ s within the frequency range 125 Hz $\leq f \leq$ 4 kHz.

It can be seen that condition (3) 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 Hz $\leq f_0 \leq$ 8 kHz. Moreover, the resulting $T_{60}(f)$ curve can later be used for assessing speech intelligibility by means of the indirect modulation method [21].

However, condition (3) may be difficult to satisfy in the frequency range $25\,Hz \leq f \leq 200$ Hz when $T_{60}(f)$ is measured using one-third-octave filters. The reason is the bias in T_{60} estimation, which can be significant, reaching tens or even hundreds of percent. It is reasonable to assume that this drawback can be mitigated or eliminated by choosing an appropriate amplitude-frequency response (AFR) shape for the bandpass filters.

Since this issue remains insufficiently explored, the goal of this work was to address the mentioned limitation.

III. EXPERIMENTAL SETUP

The study was carried out by means of computer simulation of the filtering procedure of the room impulse response $h(t)$ within the kth frequency band

$$h_k(\tau) = \int_{-\infty}^{\infty} h(v) h_{fk}(\tau - v) dv, \tag{5}$$

where $h_{fk}(t)$ is the impulse response of the bandpass filter with central frequency f_{0k}.

It can be shown that when using models

$$h(t) = h_0(t) sin2\pi f_{0k} t, \;\; h_{fk}(t) = h_{fk0}(t) sin2\pi f_{0k} t, \tag{6}$$

where $h_0(t) = g_0 e^{-at/T_{60}}$, $a = 3/lge \approx 6.9$, $h_{fk0}(t)$ is the envelope of $h_{fk}(t)$, the (2) for the kth frequency band can be represented as

$$D_{sk}(t) \approx \frac{N g_0^2}{8} \int_t^{\infty} E_k(\tau) d\tau, \tag{7}$$

$$E_k(\tau) = F^{-1}\{F[e^{-a\tau/T_{60}}] \cdot F[h_{fk0}(\tau)]\}, \tag{8}$$

where F and F^{-1} are the operators of the forward and inverse Fourier transform, respectively.

In this paper, the analysis of the influence of the filter's AFR shape on the bias of reverberation time estimation was carried out using the Tukey window $H_{f0}(f)$ [22]

$$F\left[h_{f0}(\tau)\right] = H_{f0}(f), \quad (9)$$

$$H_{f0}(f) = 0.5 \cdot \left\{1 + cos\left[\frac{\pi}{r \cdot \Delta f}\left(|f| - \frac{\Delta f}{2}(1-r)\right)\right]\right\}, \quad (10)$$

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

and

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

Thus, the window shape is adjusted by varying the value of parameter r.

Figure 1 shows the plots of $h_{f0}(t)$ and $H_{f0}(f)$ for filters with a passband of Δf = 40 Hz for $0 \le r \le 1$.

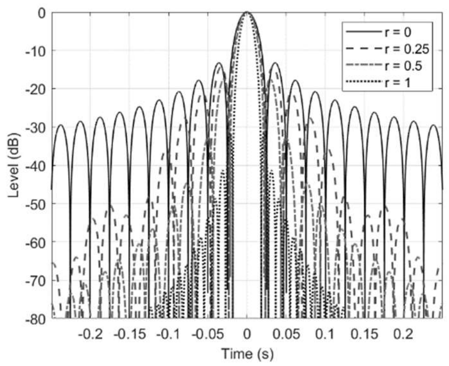
It can be seen that in the special case r =0

$$H_{f0}(f) = \begin{cases} 1, & |f| < \Delta f/2, \\ 0, & other\ f, \end{cases}$$

therefore, the side lobes of $h_{f0}(t) = \frac{sin\pi\Delta ft}{\pi\Delta ft}$ decay slowly, at a rate of $\frac{1}{t}$. It can be expected that the bias in the reverberation time estimate in this case will be the highest compared to the cases where the side lobes of $h_{f0}(t)$ decay more rapidly.

In estimating the bias of the EDT, T_{10}, T_{20} and T_{30} values, the reverberation time range $0.4\ s \le T_{60} \le 1.2\ s$ was considered, as it is typical for educational spaces [4, 5].

The value of Δf was varied from 5 Hz to 700 Hz, which proved sufficient to demonstrate the influence of the AFR shape and Δf on the bias in the estimates of EDT, T_{10}, T_{20}, and T_{30}.



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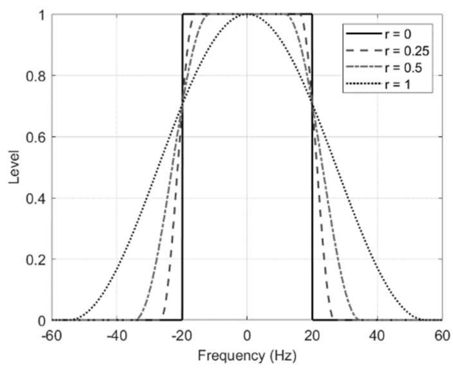


(b)

Figure 1. $h_{f0}(t)$ (a) and $H_{f0}(f)$ (b) for Δf = 40 Hz, $0 \le r \le 1$

IV. RESEARCH RESULTS

Typically, the greatest attention is given to the estimation of the T_{20} and T_{30} parameters, which characterize the room as a physical space. Significantly less attention is paid to the EDT and T_{10} parameters, as they are more closely related to the perception of room acoustics by the human auditory system [1]. This paper presents bias estimates for all four reverberation time parameters.

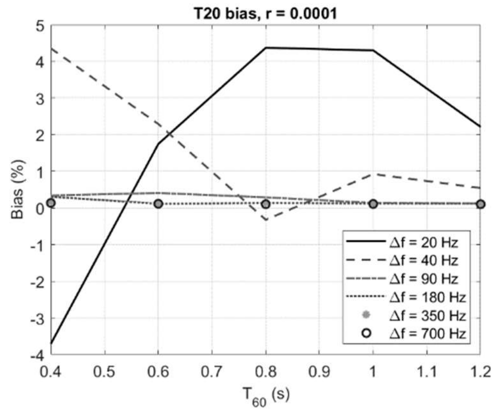
A. BIAS ESTIMATES FOR T_{20} AND T_{30}

The results of a study on the influence of filtering on the bias in reverberation time estimates are presented starting with the T_{20} and T_{30} estimates as they are the most commonly used. Relative bias estimates for T_{20} and T_{30}, depending on r, T_{60}, and Δf, are shown in Figs 2–4.

It can be seen (Fig. 2) that when filters with a rectangular-shaped AFR ($r \approx 0$) are used, the bias of the T_{30} estimate reaches 50% at Δf = 20 Hz and RT = 0.4 s. At the same time, the bias of the T_{20} parameter does not exceed 5%, which is acceptable for practical measurements.

Increasing r to 0.5 makes it possible to noticeably modify the shape of the AFR (Fig. 1) without changing the passband. As a result, under the same conditions, the bias of the T_{30} parameter is reduced to 2%, while that of the T_{20} parameter decreases to 0.5% (Fig. 3).

A further increase of r to 1 leads to a reduction in the bias of the T_{30} and T_{20} parameters to 0.14% and 0.2%, respectively (Fig. 4).



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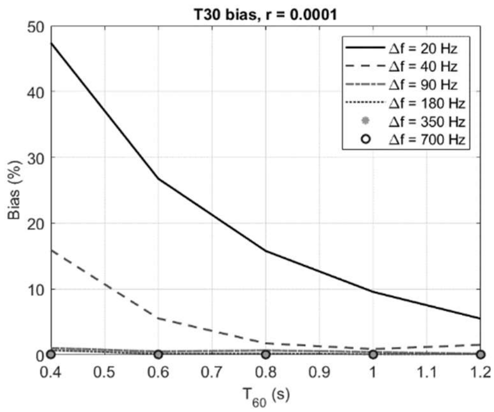
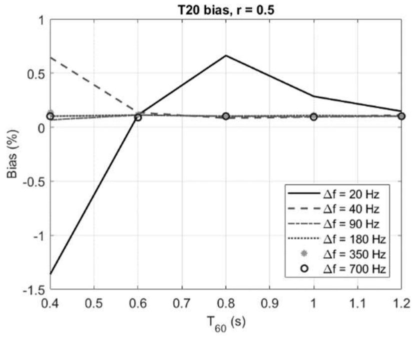


(b)

Figure 2. Bias estimates for T_{20} (a) and T_{30} (b), $r \approx 0$



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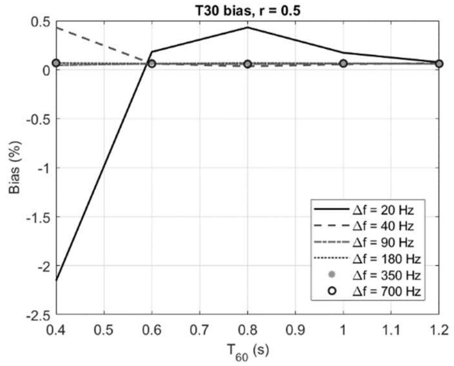
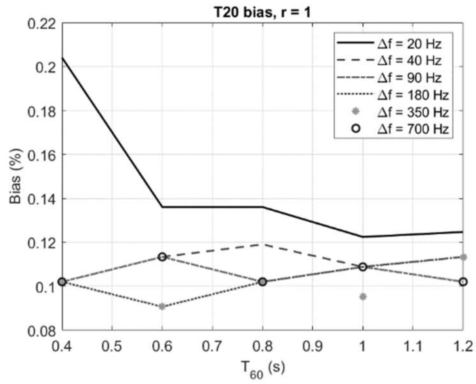


(b)

Figure 3. Bias estimates for T_{20} (a) and T_{30} (b), $r = 0.5$



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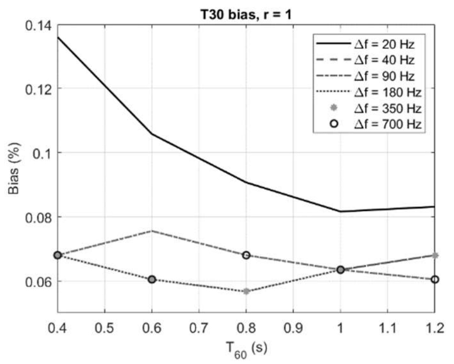


(b)

Figure 4. Bias estimates for T_{20} (a) and T_{30} (b), $r \approx 1$

Thus, it can be observed that the use of filters with a rectangular-shaped AFR ($r \approx 0$) is the least desirable compared to the cases of $r = 0.5$ and $r \approx 1$. Moreover, the bias increases significantly as Δf decreases for a fixed AFR shape.

Tables 1–3 present the relative bias estimates for the T_{20} and T_{30} parameters as a function of r, T_{60}, and Δf.

Table 1. Relative (%) bias of T_{20} and T_{30} for $r \approx 0$

	Δf	T_{60}=0.4	T_{60}=0.6	T_{60}=0.8	T_{60}=1.0	T_{60}=1.2
T_{20}	5	98,333	103,265	65,068	35,156	15,125
	10	65,068	15,057	-3,963	12,884	1,780
	20	-3,707	1,746	4,371	4,299	2,211
	40	4,354	2,290	-0,323	0,925	0,544
	90	0,340	0,408	0,289	0,136	0,125
	180	0,306	0,113	0,136	0,122	0,125
T_{30}	5	44,558	56,342	60,351	62,812	44,739
	10	60,363	45,397	47,200	22,676	26,447
	20	47,415	26,772	15,794	9,578	5,510
	40	15,941	5,548	1,757	0,898	1,534
	90	1,066	0,514	0,703	0,444	0,181
	180	0,726	0,181	0,204	0,109	0,128

Table 2. Relative (%) bias of T_{20} and T_{30} for $r = 0.5$

	Δf	T_{60}=0.4	T_{60}=0.6	T_{60}=0.8	T_{60}=1.0	T_{60}=1.2
T_{20}	5	-4,354	-30,431	-21,922	-6,136	-3,254
	10	-20,782	-3,039	-1,344	-0,735	0,125
	20	-1,361	0,113	0,663	0,286	0,147
	40	0,646	0,136	0,085	0,095	0,113
	90	0,068	0,113	0,102	0,095	0,102
	180	0,102	0,113	0,102	0,109	0,102
T_{30}	5	26,395	-39,607	-1,825	0,345	-2,373
	10	-1,429	-2,570	-2,132	-0,127	0,181
	20	-2,154	0,181	0,431	0,172	0,076
	40	0,431	0,060	0,034	0,054	0,068
	90	0,045	0,060	0,057	0,063	0,060
	180	0,068	0,060	0,068	0,063	0,060

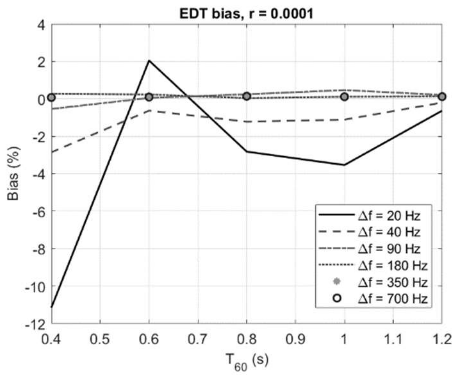
Table 3. Relative (%) bias of T_{20} and T_{30} for $r \approx 1$

	Δf	T_{60}=0.4	T_{60}=0.6	T_{60}=0.8	T_{60}=1.0	T_{60}=1.2
T_{20}	5	3,844	1,043	0,357	0,190	0,340
	10	0,068	0,159	0,255	0,245	0,181
	20	0,204	0,136	0,136	0,122	0,125
	40	0,102	0,113	0,119	0,109	0,113
	90	0,102	0,113	0,102	0,109	0,102
	180	0,102	0,091	0,102	0,109	0,113
T_{30}	5	4,150	1,315	0,442	0,299	0,348
	10	0,091	0,151	0,227	0,190	0,151
	20	0,136	0,106	0,091	0,082	0,083
	40	0,068	0,076	0,068	0,063	0,068
	90	0,068	0,076	0,068	0,063	0,060
	180	0,068	0,060	0,057	0,063	0,068

Unlike Figs 2–4, these tables consider the range $5\,Hz \leq \Delta f \leq 180\,Hz$, where the highest relative bias values in the reverberation time estimates are observed.

B. BIAS ESTIMATES FOR EDT AND T_{10}

The bias in the estimates of EDT and T_{10} is presented in Figures 5–7.



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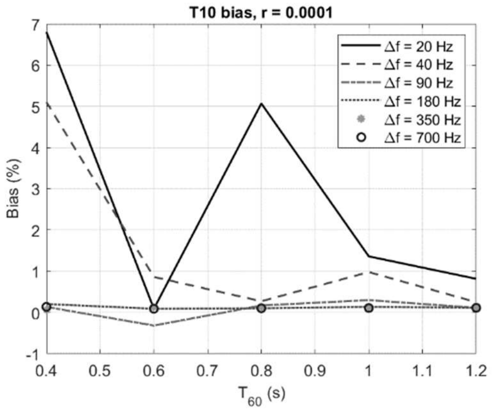
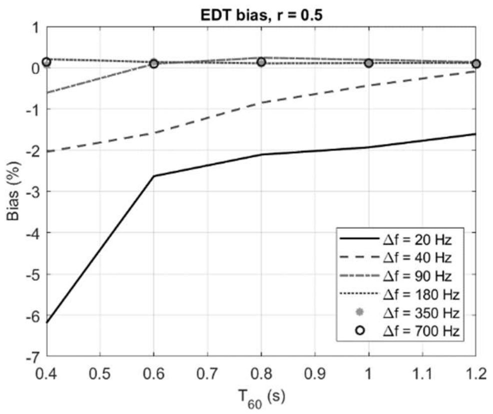


(b)

Figure 5. Bias estimates for EDT (a) and T_{10} (b), $r \approx 0$



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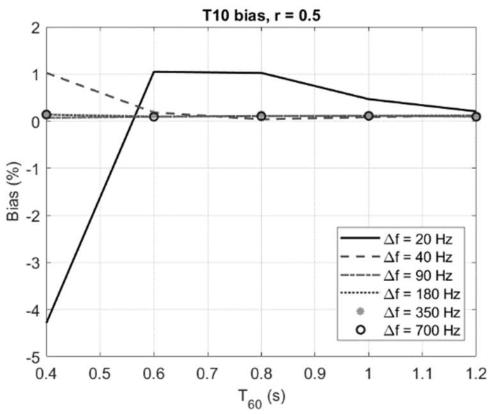
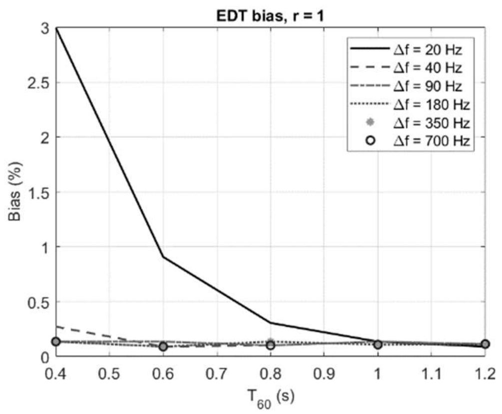


(b)

Figure 6. Bias estimates for EDT (a) and T_{10} (b), r=0.5



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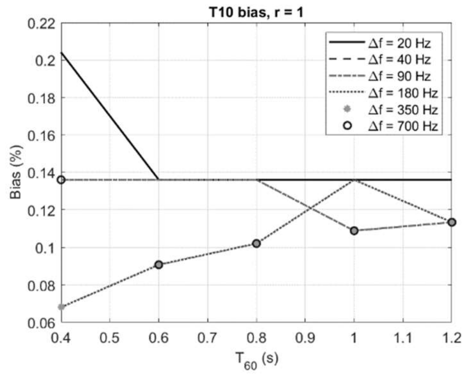

b

Figure 7. Bias estimates for EDT (a) and T_{10} (b), $r \approx 1$

By comparing these results with those obtained for the T_{20} and T_{30} estimates, it can be observed that the nature of the bias dependence on the filter's AFR shape and bandwidth is similar. Specifically, the use of filters with a rectangular-shaped AFR is the least desirable compared to the alternative cases. For a fixed AFR shape, the bias increases noticeably as Δf decreases. Tables 4–6 present the dependencies of the relative bias estimates for EDT and T_{10} on r, T_{60}, and Δf.

Table 4. Relative (%) bias of EDT and T_{10} for $r \approx 0$

	Δf	T_{60}=0.4	T_{60}=0.6	T_{60}=0.8	T_{60}=1.0	T_{60}=1.2
EDT	5	104,626	48,798	20,714	4,844	-4,467
	10	20,408	-4,671	-11,190	-4,218	2,018
	20	-11,156	2,041	-2,823	-3,537	-0,635
	40	-2,857	-0,635	-1,224	-1,116	-0,204
	90	-0,544	0,045	0,238	0,463	0,204
	180	0,272	0,227	0,034	0,109	0,136
T_{10}	5	42,109	-0,998	-20,102	-27,810	-27,347
	10	-20,272	-27,029	6,463	6,503	0,159
	20	6,803	0,091	5,068	1,361	0,816
	40	5,102	0,862	0,272	0,980	0,249
	90	0,136	-0,317	0,170	0,299	0,113
	180	0,204	0,091	0,102	0,136	0,113

Table 5. Relative (%) bias of EDT and T_{10} for $r = 0.5$

	Δf	T_{60}=0.4	T_{60}=0.6	T_{60}=0.8	T_{60}=1.0	T_{60}=1.2
EDT	5	75,850	32,154	11,361	0,435	-4,989
	10	10,680	-5,306	-6,088	-3,456	-2,653
	20	-6,190	-2,630	-2,109	-1,932	-1,610
	40	-2,041	-1,587	-0,850	-0,435	-0,091
	90	-0,612	0,091	0,238	0,190	0,136
	180	0,204	0,136	0,102	0,109	0,113
T_{10}	5	16,667	-12,336	-22,313	-22,857	-16,712
	10	-22,653	-16,281	-4,252	-0,599	1,043
	20	-4,286	1,043	1,020	0,463	0,204
	40	1,020	0,181	0,034	0,082	0,113
	90	0,068	0,091	0,102	0,109	0,091
	180	0,136	0,091	0,102	0,109	0,113

Table 6. Relative (%) bias of EDT and T_{10} for $r \approx 1$

	Δf	T_{60}=0.4	T_{60}=0.6	T_{60}=0.8	T_{60}=1.0	T_{60}=1.2
EDT	5	50,884	24,218	13,776	9,088	6,327
	10	13,878	6,304	3,129	1,741	0,975
	20	2,993	0,907	0,306	0,136	0,091
	40	0,272	0,091	0,102	0,136	0,113
	90	0,136	0,136	0,102	0,136	0,113
	180	0,136	0,091	0,136	0,109	0,113
T_{10}	5	9,456	0,363	0,170	0,000	0,113
	10	-0,068	-0,091	0,272	0,272	0,159
	20	0,204	0,136	0,136	0,136	0,136
	40	0,136	0,136	0,136	0,109	0,113
	90	0,136	0,136	0,136	0,109	0,113
	180	0,068	0,091	0,102	0,136	0,113

These tables cover the range $5\ Hz \le \Delta f \le 180\ Hz$, where the highest relative bias values in reverberation time estimates are observed.

V. DISCUSSION

Bandpass filters are relatively easy to implement in the frequency domain by eliminating spectral components outside the desired passband. However, it can be seen that this advantage comes at the cost of a significant bias in reverberation time estimates in the narrow passband and short reverberation time cases.

As seen in Fig. 2, for the window function (9) with $r \approx 0$ (rectangular AFR), the relative bias in the estimates of T_{30} and T_{20} can become unacceptably large when $20\ Hz \le \Delta f \le 350\ Hz$, which must be considered in practical reverberation time measurements.

In contrast, with $r \approx 1$ (cosine-shaped window), the relative bias does not exceed 0.14% for T_{30} and 0.2% for T_{20} within the range $0.4\ s \le T_{60} \le 1.2\ s$ and $20\ Hz \le \Delta f \le 350\ Hz$ (Fig. 4).

Under conditions $0.25 \le r \le 0.75$ and $0.4\ s \le T_{60} \le 1.2\ s$, the relative bias in the estimates of T_{30} and T_{20} remains moderate and does not exceed 2% (Fig. 3).

The relative bias values of T_{30} and T_{20} presented in Tables 1–3 complement the results shown in Figs 2–4 by providing analysis for the narrowest frequency bands of 5 Hz and 10 Hz, which occur when using the full set of one-third-octave filters [3]. It can be observed that, under the condition $r \approx 1$ for window function (9), the bias does not exceed 4% even for the narrowest frequency bands.

At the same time, the bias values of the T_{30} and T_{20} estimates presented in Tables 1–3 indicate the presence, under certain conditions, of two nontrivial phenomena:

- the magnitude of the positive bias can approach 100%;
- the negative bias may be substantial and can reach up to minus 40%.

To explain the nature of these phenomena, Figs 8 and 9 are used.

Fig. 8 shows the plots of the functions $h_0(t) = g_0 e^{-at/T_{60}}$ and $E_k(t)$ for the $r \approx 0$ under the conditions $T_{60} = 0.4\ s$, $\Delta f = 5\ Hz$.

It can be seen that, in this case, the considerable positive bias - reaching 98% for T_{20} - is caused by the combination of two factors. The first factor is the slow decay of the function $E_k(t)$ between thresholds -5 dB and -35 dB, which results from the rectangular shape of the filter's AFR and the very narrow bandwidth. The second factor is the steep slope of the function $h_0(t)$, which is due to the short reverberation time.

Fig. 9 shows the plots of the functions $h_0(t)$ and $E_k(t)$ for the $r = 0.5$ under the conditions $T_{60} = 0.6\ s$ and $\Delta f = 5\ Hz$.

Unlike the previous case shown in Fig. 8, here the decay rate of the function $E_k(t)$ between thresholds -5 dB and -35 dB is noticeably lower than that of the function $h_0(t)$. As a result of this difference, a clearly negative bias of up to –30% and –40% is observed in the estimation of the T_{20} and T_{30} parameters, respectively.

It may seem that the considered cases are somewhat academic, as the simultaneous occurrence of the above-mentioned factors is unlikely in practical measurements. Nevertheless, it is useful to be aware of the causes of potentially anomalous reverberation time estimates and of relatively simple ways to avoid them. Indeed, it can be seen that when $r \approx 1$, the absolute value of the relative bias in the estimates of T_{30} and T_{20} does not exceed 4.2%, even for $\Delta f = 5\ Hz$ and $T_{60} = 0.4\ s$.

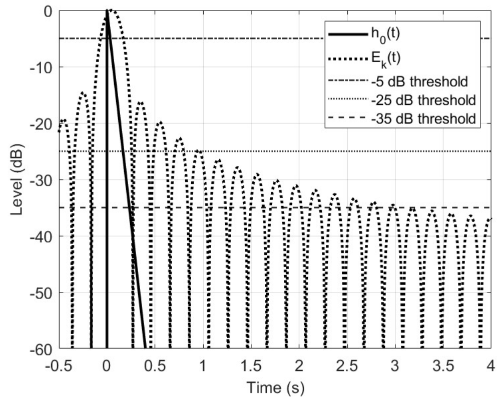

Figure 8. $h_0(t)$ and $E_k(t)$ for $r \approx 0$ under the conditions $T_{60} = 0.4\ s$ and $\Delta f = 5\ Hz$

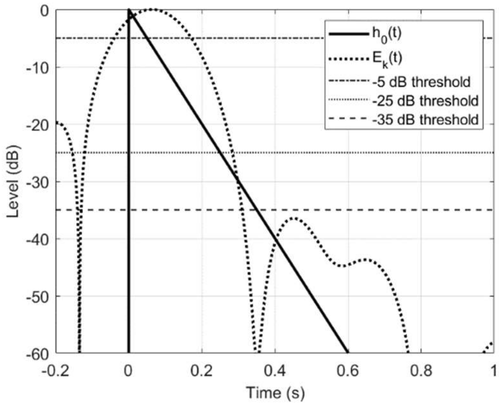

Figure 9. $h_0(t)$ and $E_k(t)$ for $r = 0.5$ under the conditions $T_{60} = 0.6\ s$ and $\Delta f = 5\ Hz$

The bias in the estimates of EDT and T_{10} is also minimal under the condition $r \approx 1$ for window function (9) (Figs 5–7). But under the conditions $T_{60} = 0.4\ s$ and $\Delta f = 5\ Hz$, the absolute value of the relative bias is about 52% for EDT and 9.5% for T_{10} (Tables 4–6).

These results indicate that, in practical measurements of T_{30}, T_{20}, EDT, and T_{10}, it is important to consider the potential impact of the AFR shape of bandpass filters, as well as the influence of passband width and the expected reverberation time on estimation bias.

This can be accounted for either at the calibration stage of the measurement system of reverberation time or through correction of existing reverberation time estimation algorithms [23, 24].

In future work, it would be advisable to evaluate the effectiveness of the proposed recommendations under conditions involving background noise [25].

VI. CONCLUSIONS

The influence of filtering on the variance of reverberation time estimates is well established and is explicitly considered in ISO 3382-1:2009. However, the standard does not characterize either the nature or the extent of bias in reverberation time estimates caused by filtering.

In this paper, an analysis was carried out to investigate the influence of the filter's amplitude-frequency response shape on the bias of reverberation time estimates, under the assumption that the AFR has the shape of a Tukey window. It is shown that such influence can lead to unacceptably large bias in the estimates of T_{20} and T_{30} for certain combinations of AFR shape, filter bandwidth, and the reverberation time being measured. Similar results are obtained for the EDT and T_{10} estimates.

The obtained results indicate the advisability of incorporating bias estimation into the calibration procedure of reverberation time measurement systems.

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