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1aAAa6. On the effects of pre-processing of impulse responses in the evaluation of acoustic parameters on room acoustics

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The evaluation of room acoustics characteristics has been thoroughly described in several papers since 60-ies. Moreover, the ISO 3382 standard describes several acoustic parameters and their measurements. However, there is only a few information about the methods of pre-processing the impulse responses (background noise compensation) that are required before calculating those parameters. In this paper the main processing methods (based on Chu, Lundeby and Hirata) are analyzed. Moreover they are compared with the results obtained without any processing (original Schroeder backward integration). In a further step, these methods are applied in some acoustic measurements of some opera houses in Italy. Finally some acoustic parameters are compared with the JND that is actually accepted in the evaluation of the mono-aural, binaural and spatial acoustic parameters.

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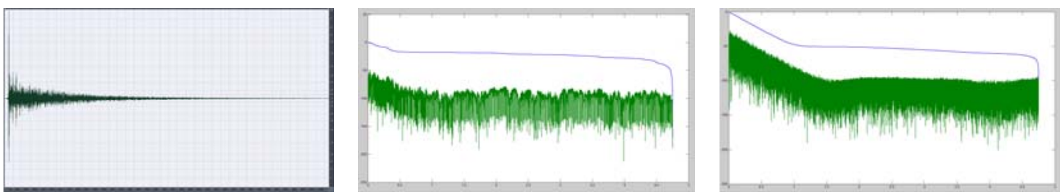
INTRODUCTION

ISO 3382 [1] represents the standard that has to be followed to calculate acoustical parameters. Impulse response is the full descriptive quantity for a LTI system, therefore information about all the acoustic parameters (i.e. reverberation time, clarity, etc.) is contained in it. Using the backward integration of the squared impulse response it is possible to obtain a smooth quantity (Schröder integral) [2] that is suited to calculate all of them (1): reverberation time is proportional to the slope of best fit line on the Schröder integral expressed in dB scale.

$$E(t) = \int_t^\infty h^2(t)dt = \int_0^\infty h^2(t)dt - \int_0^t h^2(t)dt \tag{1}$$

The results obtained using (1) could be sensitive to background noise, for example the unconstrained end of the integration could let the noise be included. These problems have been addressed by many techniques [3-6]. Figure 1 shows IR and two decay curves of Teatro 1763 Villa Aldrovandi Mazzacorati in Bologna, Italy.

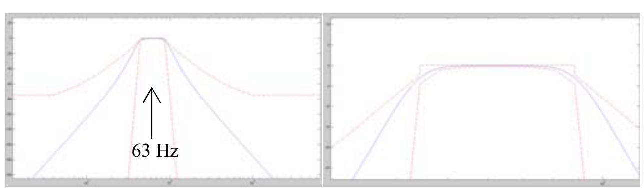


(a) (b) (c)

FIGURE 1. (a) Impulse response. (b) (c) Schröder integral curve and squared IR in dB scale (filtered at 63 Hz and 4 kHz)

NOISE COMPENSATION METHODS

Chu [3], Lundeby [4] and Hirata [5] methods have been designed to compensate for background noise. The same way to find the starting point of Schröder integration and the same octave filters bank has been employed during the analysis with [3-5]. The starting point is chosen as the instant in which the signal is higher than the noise and it is more than 20 dB below its maximum, this procedure is applied after the signal has been filtered [1]. Filters bank is made up of octave band-pass filters with a designed slope of 36 dB/oct (class 0 IEC 61260 compliant) [7], Figure 2. Each filter has a zero phase response. The methods mentioned above are described below and some results are shown. In the following sections the term “Schröder method” is used as a synonymous for “without-noise compensation method”.



(a) (b)

FIGURE 2. (a) Response at 63 Hz, (b) Zoom-in of Figure 2(a) response. The dotted lines define the IEC 61260 class 0 mask

Chu

The RMS value of the last 10% of the signal is computed for each band. It is supposed that this interval contains background noise. The backward integration is applied to the squared filtered impulse response minus the squared RMS value of noise in that band. Due to the term $-\mathrm{RMS}^2$ the decay curve is not necessarily monotonically decreasing. Figure 3 compares two decay curves obtained applying (1) and (2) to a specimen IR. The straighter profile of the Chu curve should provide a more secure RT extraction.

$$E(t)=\int_t^{\infty}\left(h^2(t)-RMS^2\right)dt=\int_0^{\infty}\left(h^2(t)-RMS^2\right)dt-\int_0^{t}\left(h^2(t)-RMS^2\right)dt \tag{2}$$

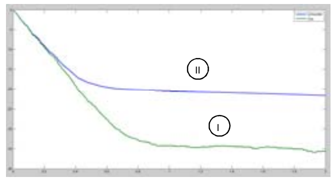


FIGURE 3. Comparison of decay curves at 1000 Hz, with (I) and without (II) Chu noise correction

Lundeby

Lundeby method is based on an iterative algorithm that chooses the right temporal interval to which the linear regression is calculated. This will leave out unwanted contributions. Algorithm description is reported below [4].

1. Average squared IR in local time intervals (10-50 ms)
2. Estimate background noise level using tail (last 10% of the signal)
3. Estimate slope of decay from 0 dB to noise level (the "left" point is 0 dB. Search the "right" point 5-10 dB above noise level)
4. Find the crosspoint defined by regression line and noise level
5. Find a new local time intervals based on the actual slope (use 3-10 intervals per 10 dB of decay)
6. Average squared IR in new local time intervals
7. Estimate background noise (starting from a time corresponding to a decay of 5-10 dB based on actual slope after the actual crosspoint but 10% of length of IR should be the minimum length)
8. Estimate late decay slope (a dynamic range of 10-20 dB should be evaluated, starting 5-10 dB above the noise level)
9. Find a new crosspoint
10. Repeat 7, 8 and 9 until convergence of crosspoint is achieved

Hirata

Hirata method needs two IRs, $h_1(t)$ and $h_2(t)$, of the same measurement point. The Schröder integral is calculated on the product of the two impulse responses. In this way the correlated components are kept (common decays) while the uncorrelated components (noise) are rejected. Hirata method is applied to two versions of the previous specimen IR, $h_1(t)$ and $h_2(t)$. They differ only in the background noise signal (uncorrelated pink noise).

1000 Hz	EDT [s]	T20 [s]	T30 [s]	T40 [s]
Schröder	1.41	24.42	22.05	20.43
Chu	1.30	1.27	3.97	15.23
Lundeby	1.30	1.27	1.26	1.19
Hirata	1.29	1.22	1.19	1.12
ref. RT	1.2	1.2	1.2	1.2

TABLE 1. Schröder, Chu, Lundeby and Hirata result comparison

Table 1 summarizes the results obtained analyzing an artificially created IR with RT at 1 kHz equals to 1.2 seconds and a S/N of 20 dB.

METHODS TESTS

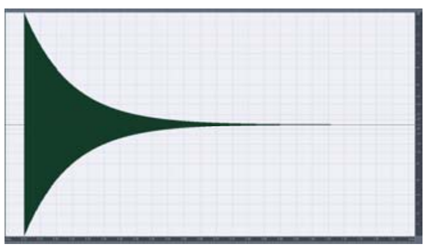
Some IRs with known acoustical features are required to evaluate the noise compensation methods of Lundeby and Chu. A MATLAB tool has been developed to create synthetic impulse responses. It could generate IRs with known RMS power and constant decay slope for each octave band (62.5-8000 Hz). Every synthetic IR could be a sum of eight sinusoids (62.5, 125, 250, 500, 1000, 2000, 4000, 8000 Hz) or a sum of eight octave filtered white noise. Equation (3) shows how the sum of sinusoid signal is designed. The tool allows adding pink or white noise as background. These two scenarios simulate the noise contained in an IR measured using a direct (gunshot) or indirect (exponential sweep [8]) technique.

$$\begin{cases} \tau_i = \dfrac{-\ln(10^{-3})}{T_{60_i}} \\ y_i = A_i \cdot e^{-\tau_i \cdot t} \cdot \cos(2\pi f_i t) \\ y = \sum_{y=1}^{8} y_i \end{cases} \tag{3}$$

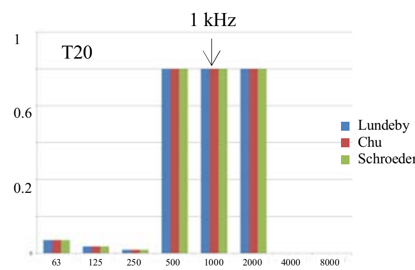
Synthetic IRs

Pure Sinusoidal Signals

The very first test is to evaluate the Lundeby, Chu and Schröder methods with a decaying single tone at 1000 Hz, with a RT of 1 second, IR#0, Figure 4a.



(a)



(b)

FIGURE 4. (a) Impulse response of IR#0, (b) T20 of IR#0 (Lundeby, Chu, Schröder) 63-8000 Hz

All the methods provide the correct result even if the reference tone is perceived by neighboring bands. It is clear that the values outside 1000 Hz make no sense but this experiment shows how energy spreading (filter selectivity) could cheat the results, Figure 4b. Next IR#1 and IR#2 have been analyzed. Both the signals are a sum of cosines [1]. IR#1 and IR#2 differ only in S/N, in IR#1 it is 96 dB while in IR#2 it is 40 dB. Table 3 shows IR#1 features.

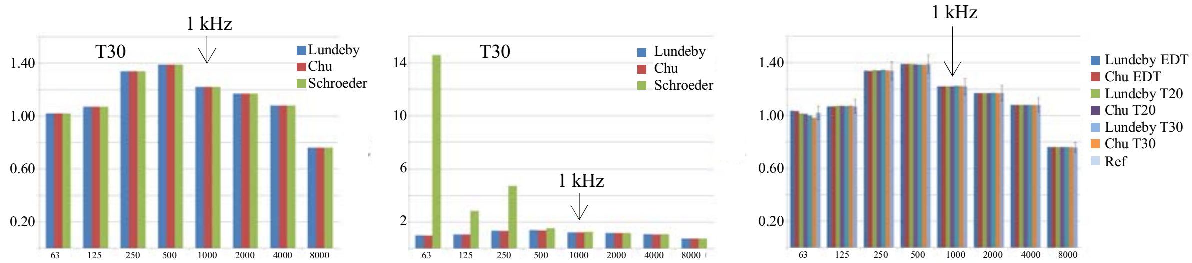
IR#1	**63**	**125**	**250**	**500**	**1000**	**2000**	**4000**	**8000**	**Hz**
GAIN	-8.51	-5.02	-5.53	-2.78	0	4.79	8.51	6.38	**dB**
T60	1.02	1.07	1.34	1.39	1.22	1.17	1.08	0.76	**s**
Noise	Pink noise at -96 dB from the RMS value of the clean signal								

TABLE 1. IR#1 features

In IR#1 example all methods provide the same and correct result both for EDT, T20 and T30. Figure 5a reports T30 (Lundeby, Chu and Schröder).

In IR#2, Schröder method fails, whereas both Lundeby and Chu provide comparable and quite right results. Figure 5b reports T30. The EDT, T20 and T30, calculated with Lundeby and Chu methods, fit in a ±5% tolerance from the correct value, Figure 5c.

[1] Cosines have been chosen to achieve a synchronous attack.



(a) (b) (c)

FIGURE 5. (a) IR#1 T30 results, (b) IR#2 T30 results, (c) IR#2 EDT-T20-T30 results (Lundeby and Chu)

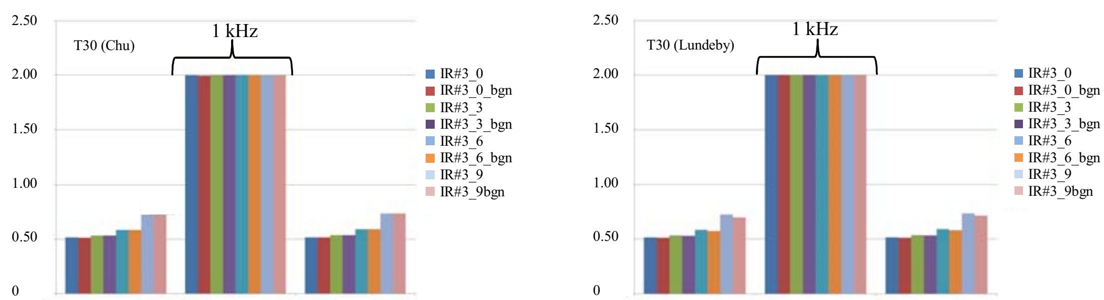
The following test (IR#3 set) has been thought to investigate the effect of filters selectivity during analysis: energy coming from adjacent band could generate an artificial dual slope in the decay curve, as shown in Figure 6. In order to test this phenomenon a set of IRs has been designed and analyzed with Lundeby and Chu. Impulse responses have the same RTs while the gain of one component (1000 Hz) changes with a step of 3 dB, from 0 to 9. The higher the 1000 Hz energy is the more it will be perceived by adjacent bands. The filtering is performed by means of 36 dB/oct octave band-pass bank. Each signal is provided in two versions: with and without background noise (-40 dB pink noise). Eight signals have been therefore analyzed.

FIGURE 6. Artificial dual slope due to poor filter selectivity

IR#3	63	125	250	500	1000	2000	4000	8000	Hz
GAIN	-192	-192	-192	0	0,3,6,9	0	-192	-192	**dB**
T60	1	1	1	0.5	2	0.5	1	1	**s**

TABLE 4. IR#3 features

The maximum error is 2% for EDT, 8% for T20 and 48% for T30. These data refer to the case of +9dB energy difference. In the case of +3dB the maximum error is 2% for T20 and 8% for T30. EDT value is estimated perfectly by both methods. Looking at Figure 7 (T30) it could be noticed how some background noise could even improve the confidence of the T30 at adjacent octaves as it could help to mask the artificial dual slope.



(a) (b)

FIGURE 7. IR#3 set. (a) T30 Chu method, (b) T30 Lundeby method

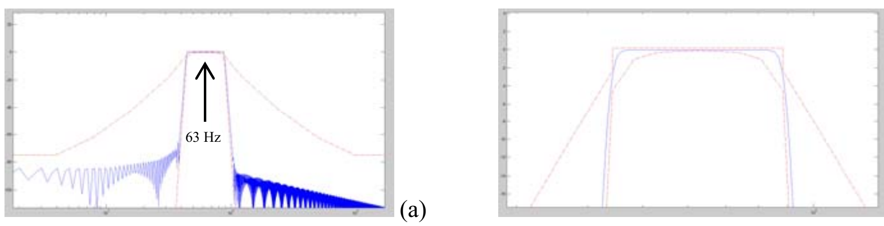
Filtered White Noise Signals

In this case the synthetic IR is designed as the sum of filtered white noises. A filter bank has been employed to create each component. Each octave filter has been designed with a 144 dB/oct slope, Figure 8. Such a high selectivity is necessary in order to limit the influence of common frequencies between bands. The specimen impulse response features are shown below. S/N ratio and energy difference between bands has been chosen accordingly to previous experiences (40 dB and 3 dB).

IR#4	**63**	**125**	**250**	**500**	**1000**	**2000**	**4000**	**8000**	**Hz**
GAIN	-192	-192	-192	0	3	0	-192	-192	**dB**
T60	1	1	1	0.5	2	0.5	1	1	**s**
Noise	Pink noise at -40 dB from the RMS value of the clean signal								

TABLE 5. IR#4 features

The purpose is to find if the decay related parameters of the 500 Hz band are compromised by the +3 dB energy of the 1000 Hz band. Before it is summed with the others, EDT, T20 and T30 of the 500 Hz band signal are calculated. These provide the reference values. IR#4 acoustic parameters are then extracted using filter banks with different selectivity (36 and 144 dB/oct slope), both belonging to class 0 IEC 61260.



(a) (b)

FIGURE 8. 144 dB/oct slope octave filter class 0 IEC 61260 compliant (a). Zoom-in (b)

IR#4 500 Hz	**ref.**	**36 dB/oct slope**	**error**	**144 dB/oct slope**	**error**	**Δ**
EDT	0.54	0.66	22.22%	0.57	5.56%	16.66%
T20	**0.50**	**1.38**	**176%**	**0.64**	**28%**	**148%**
T30	0.50	1.78	256%	1.43	186%	70%

TABLE 6. Results of IR#4 test (Lundeby and Chu same results)

Lundeby and Chu provided the same values (2 digits precision), as shown in Table 6. Steeper filter always improves the results. T20 is the parameter that mostly has been modified by filter slope.

Real IRs

Lundeby, Chu and Hirata methods are now employed to analyze IRs of the Teatro 1763 Villa Aldrovandi Mazzacorati in Bologna, Italy and Teatro Regio in Parma, Figure 9. The parameters are calculated with both the previous filter banks. Results are compared with the JND [2] for reverberation time that is about 5% after [9]. Figure 10 shows the results of EDT, T20 and T30 compared with JND tolerance applied to the mean of the values. In all cases the results are well correlated and their differences are not acoustically noticeable. Figure 10d shows in one chart the means of EDT, T20 and T30 compared with the 5% tolerance applied to their mean. Starting from 1000 Hz all the parameters fit into the boundary, therefore the decay is quite regular. These values are obtained employing 36 dB/oct slope filters. Figure 11 shows the results obtained with 144 dB/oct slope filters. Figure 12 and Figure 13 summarize slope effects on Teatro 1763 and Teatro Regio analysis.

[2] Just Noticeable Difference: the smallest stimulus increment that is just detectable

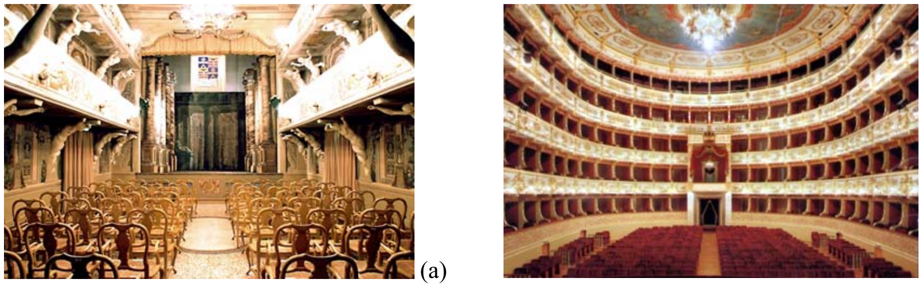


FIGURE 9. (a) Teatro 1763 Villa Aldrovandi Mazzacorati, (b) Teatro Regio Parma

36 dB/oct slope (Villa Mazzacorati)

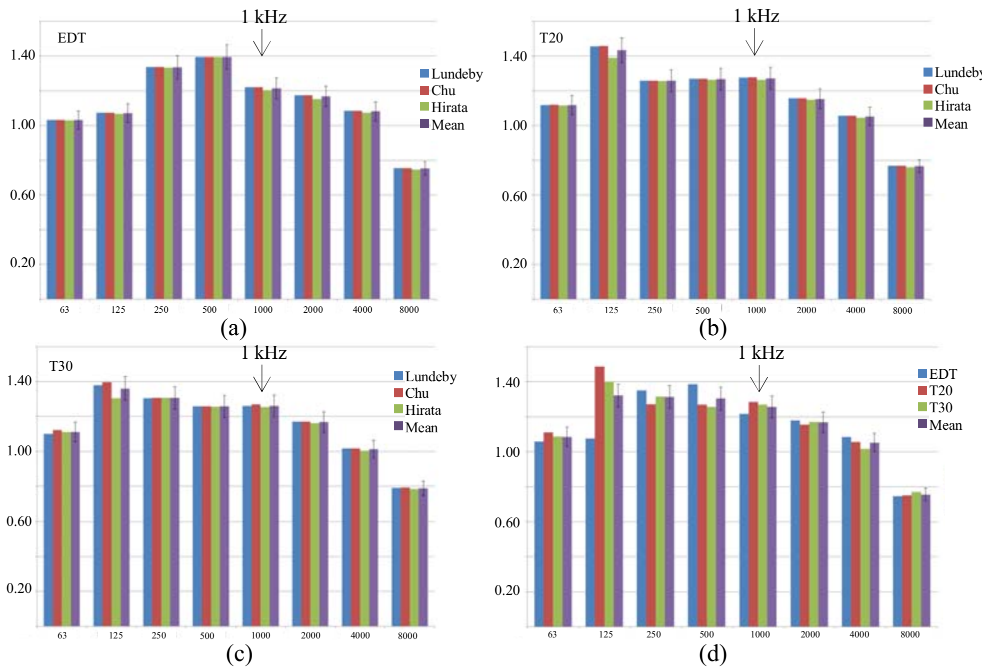
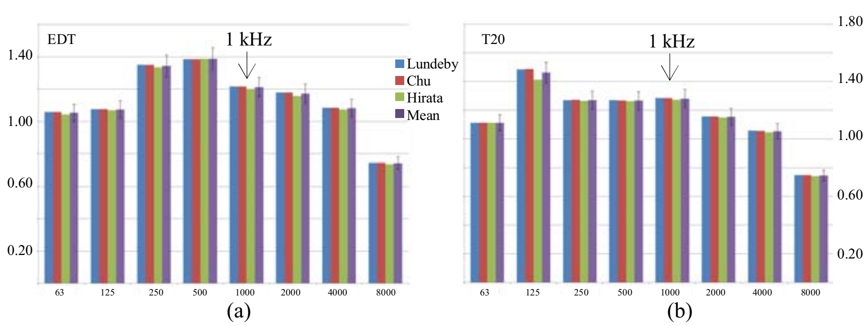
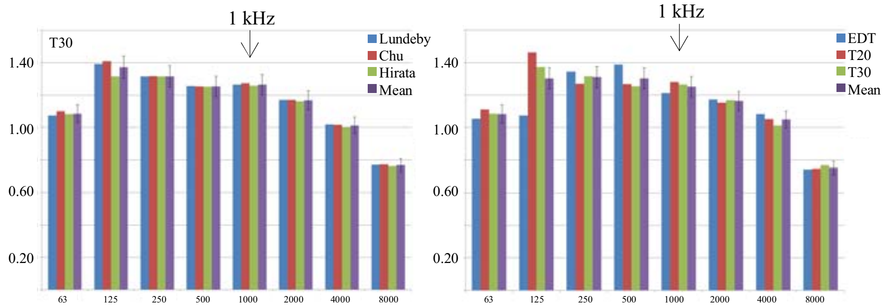


FIGURE 10. (a) EDT, (b) T20, (c) T30 employing Lundeby Chu and Hirata compared with mean value ±5%, (d) mean of EDT, T20, T30 comparison

144 dB/oct slope (Villa Mazzacorati)

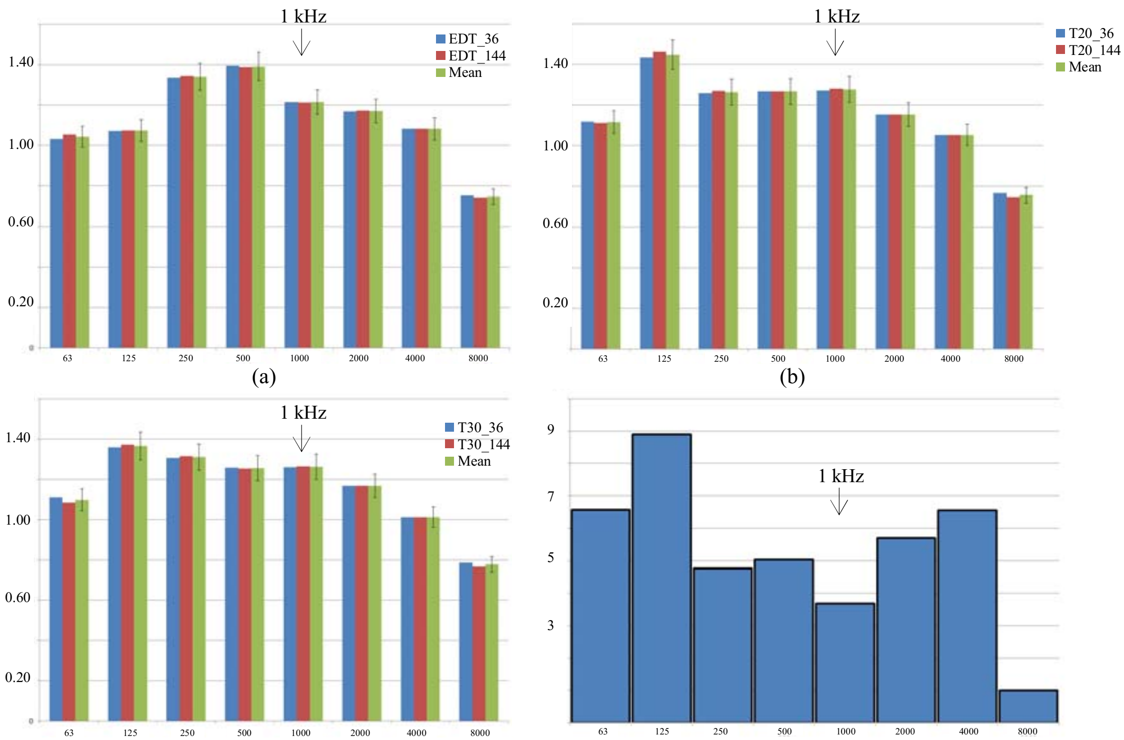




(c) (d)

FIGURE 11. (a) EDT, (b) T20, (c) T30 employing Lundeby, Chu and Hirata compared with mean value ±5%, (d) mean of EDT, T20, T30 comparison

Slopes comparison (Villa Mazzacorati)



(c) (d)

FIGURE 12. (a) EDT, (b) T20, (c) T30 slopes comparison (d) octave spectrum

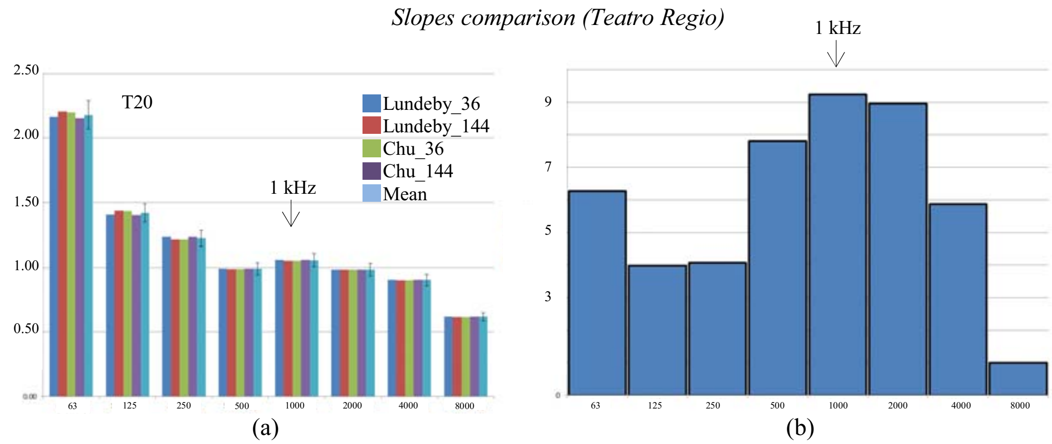


FIGURE 13. (a) Methods and slopes comparison with mean ±5%, (b) octave spectrum

CONCLUSIONS

It has been noticed how different octave filters belonging to the same IEC 61260 class could generate different results when employed to calculate acoustical parameters. T20 has revealed more sensible to filter selectivity compared to EDT or T30. The problem has been highlighted employing specimen IRs with known acoustical features. Some real impulse responses have been analyzed too. In these cases selectivity seems not to affect significantly the final result probably because in real IRs adjacent bands have decays and energies that vary quite smoothly. However a common sense solution could suggest employing smooth filters to extract “short” (EDT, T10) parameters in order not to alter significantly the time response and steeper filters for “long” parameters in order to reject energy of adjacent band on the late part of the IR. Broadband time decay could give a hint on the slope to choose. It should be mandatory to use a poor selective filter when the time decay is “short”: the Uncertainly Principle limits the slope of the filter if it is necessary that its time response is short.

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