Results for Room Acoustics Equalisation Based on Smoothed Responses

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ABSTRACT

Digital equalisation of room acoustics based on inverse filtering of measured response functions, introduces a number of theoretical and practical challenges. To overcome such problems, inverse filtering based on modified measured responses is proposed, derived via their complex Transfer Function smoothing, so that the processed responses are more perceptually compliant, of lower order and less position-sensitive than the original functions. Aim of this study is to evaluate via objective and subjective tests conducted for different-sized rooms and real-time reproduction, the use of such smoothed room responses for the derivation of appropriate room equalisation filters, which can improve the perceived and measured quality of audio reproduction in any reverberant environment.

0. INTRODUCTION

It is known for nearly 2 decades that the digital equalisation of room acoustics based on inverse filtering of measured impulse response functions, introduces a number of theoretical and practical challenges. These are related to the extreme high-order of the inverse filters and their sensitivity to source – receiver placement inside any enclosure. In addition, the use of direct inverse response filtering (given the mixed-phase character of room responses) can introduce pre-echo and strong narrow-band colouration effects, rendering undesirable the perceived results of the processed audio.

Many techniques have been proposed, in the past, to optimise the results of equalised audio / acoustic signals delivered in rooms [1-8]. In addition some practical systems have been also developed [9-11].

To overcome these problems, the present work, presents results for acoustic equalization achieved from inversion of appropriately modified measured room responses, based on the generalised Complex Smoothing approach, proposed by the authors [12].

1. THEORY

1.1. The concept of Complex Smoothing

Let H(k) be a Room Transfer Function and $|H(k)|^2$ be its Power Spectrum response where k is the discrete frequency index ($0 \le k \le N-1$). Traditionally, the spectral smoothing operation for discrete-time signal sequences such as the digitally-measured acoustic/electroacoustic/audio responses can be described as a circular convolution:

$$H_{ps}(k) = \sum_{i=0}^{N-1} \left| H\big((k-i) \bmod N\big) \right|^2 \cdot W_{sm}(m,i) \quad (1)$$

where $W_{sm}(m,k)$ is a (zero-phase) spectral smoothing window function, having the general form of a low-pass filter where m (samples) is a sample index corresponding to the cut-off frequency f_c (Hz).

The complex smoothing operation has been defined [12] as a generalized version of eq. (1). The spectral smoothing function $W_{sm}(m,k)$ operates on the complex Room Transfer Function H(k) accordingly to eq. (1):

$$H_{cs}(k) = \sum_{i=0}^{N-1} H((k-i) \bmod N) \cdot W_{sm}(m,i) \quad (2)$$

The discrete variable m (samples) is defined as a function of k, allowing a variable degree of spectral averaging for each value of the discrete frequency index k, when fractional octave or other non-uniform frequency smoothing is required. Following such an approach, m(k) can be considered as a bandwidth function that expresses the dependence of the desirable spectral averaging on the frequency variation. Figure 1 shows some well-established alternative frequency/resolution profiles.

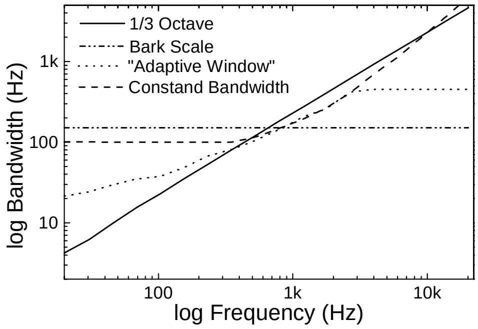


Figure 1: Relationship of smoothing bandwidths as function of frequency for various known profiles.

The complex nature of the above operation, allows mapping of the Complex Smoothed Room Transfer Function into the corresponding smoothed room impulse response [12].

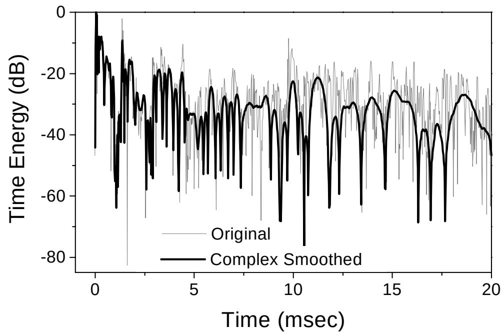
Typical results illustrating how the Complex Smoothing operation affects the Room Transfer Function amplitude spectrum (in somewhat traditional fashion) and also the time response, are presented in Figure 2.

Observing Fig. 2(a), it is evident that in the time-domain, the application of Complex Smoothing retains the initial high-frequency content of early components (i.e. transient features of direct signal and first reflection) and then, progressively introduces low-pass properties to the later time (room reflection) components. This effect is generally desirable, since it is appropriate from a perceptual point of view to represent impulse responses of audio / acoustic systems with the exact definition of their initial transient portion, whereas later time components can be "less-focused" i.e. can be progressively low-passed [13].

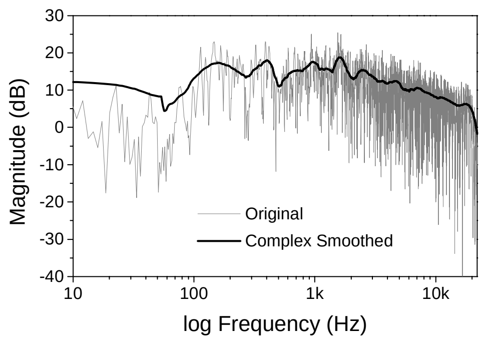
1.2. Equalization filter design

Let h(n) describe a discrete-time room impulse response in a specific source/receiver position. Following the direct approach for ideal digital equalization [1-2], an "inverse filter" $h^{-1}(n)$ and Transfer Function $H^{-1}(k)$, has to be introduced in the sound transmission path, such that a new equalised room response $h_{eq}(n)$ will be produced, accordingly to the following expression:

$$
\begin{aligned}
h_{eq}(n) &= h(n) * h^{-1}(n) = \delta(n) \\
H_{eq}(k) &= (H(k) \cdot H^{-1}(k) = 1
\end{aligned}
\tag{4}
$$



(a)



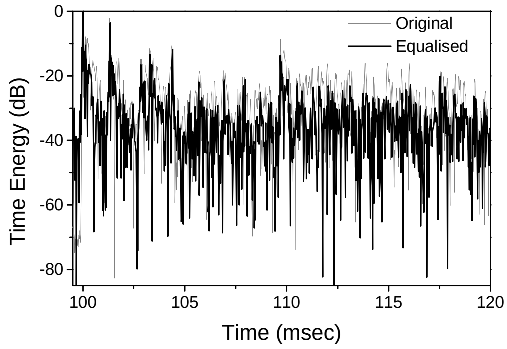
(b)

Figure 2: Complex Smoothed vs original room response: (a) Time domain (Energy), (b) Frequency domain

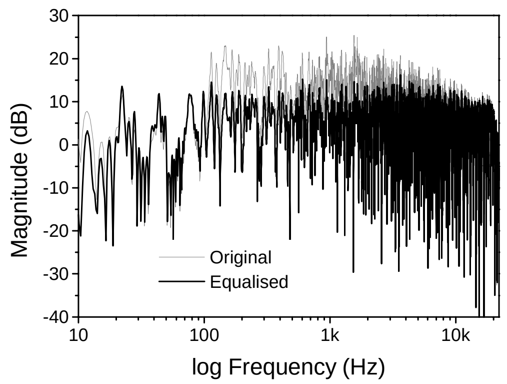
where δ(n) is the ideal impulse response and the symbol $*$ denotes the disctete-time convolution. The inverse filter response $h^{-1}(n)$ is calculated as the one that yields the best (least-square) approximation to the ideal response δ(n), when it is convolved with the measured room response.

In order to overcome the known problems of direct equalization, the scheme proposed in the present work is based on the pre-processing of room responses via Complex Smoothing. After the application of this operation on a measured room response h(n), a reduced-order complex smoothed response $h_{cs}(n)$ and a Transfer Function $H_{cs}(k)$ are produced. Then, an inverse equalization filter $[h_{cs}]^{-1}(n)$ is evaluated, which equalises the complex smoothed response, i.e.:

$$
\begin{aligned}
[h_{cs}]_{eq}(n) &= h_{cs}(n) * [h_{cs}]^{-1}(n) = \delta(n) \\
[H_{cs}]_{eq}(k) &= (H_{cs}(k) \cdot [H_{cs}]^{-1}(k) = 1
\end{aligned}
\tag{5}
$$



(a)



(b)

Figure 3: Room Response before and after the proposed equalisation method: (a) Time domain (Energy) (b) Frequency domain

This inverse filter response $[h_{cs}]^{-1}(n)$ will be the "equaliser", employed on the original measured room response h(n) accordingly to the following expression:

$$
\begin{aligned}
h_{eq}(n) &= h(n) * [h_{cs}]^{-1}(n) \\
H_{eq}(n) &= H(n) \cdot [H_{cs}]^{-1}(n)
\end{aligned}
\tag{6}
$$

An example of a room response function before and after the proposed equalisation procedure is shown in Figure 3. As it can be observed, in the time domain the equalised response has more power shaped in the direct and early reflection path sounds and less power allocated in some of the reverberant components. In the frequency domain, the equalisation procedure corrects gross spectral effects due to early room reflections, without attempting to compensate for many of the original narrow-bandwidth spectral dips.

2. METHODOLOGY

2.1. Room response measurement and processing.

The proposed method was applied to measured responses of closed acoustic spaces with different properties (e.g. size, reverberation and function), ranging from a small office to a large concert hall. The dimensions of the six selected rooms are given in Table 1, classified with respect to their volume as small, medium and large-sized spaces.

In Table 1, Room 1 and Room 2 are laboratory spaces and Room 2 (with dimensions are L: 7,15m X W: 4,60m X H: 2,90m) was employed for the subjective listening tests discussed later in this paper (see Figure 4).

Size	**Spaces**	**Volume (m^3)**
small < 100 m^3	Room 1 Room 2	52 87
medium >100 m^3 <1000 m^3	Room 3	386
large >1000 m^3	Room 4 Room 5 Room 6	1230 2000 11000

Table 1: Properties of measured rooms.

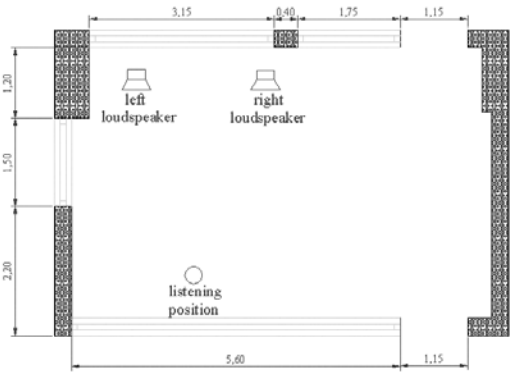


Figure 4: Plan view of the laboratory used for subjective tests.

For all room response measurements, an omnidirectional microphone was employed and the excitation signal was a maximum-length sequence (MLS). The acoustic properties, detailed diagrams and auralizations for most the tested spaces can be found in the web address:
http://www.wcl.ee.upatras.gr/audiogroup
/PatrasVRML
/Patra's%20halls_en.htm

After the measurement procedure, the derived room responses $h_i(n)$, where i=Room1,…,Room6, were processed by the complex smoothing operation, producing the complex smoothed responses $(h_{cs})_i(n)$ which were used to construct the appropriate inverse filter responses $[(h_{cs})_i]^{-1}(n)$. The smoothed responses exhibit different acoustic properties when compared to the original measurements, as was described in an earlier paper [14]. Nevertheless, main objective of this response pre-processing in the present study, was for the design of perceptually-optimised inverse equalization filters.

The implementation of the proposed equalization scheme for both the objective and subjective tests described later, is shown in Figure 5. As it can be observed, in the case of objective analysis (Fig. 5(a)), the equalisation filter responses are convolved by the original measured room responses in order to evaluate objective acoustic parameters, while in the case of subjective listening tests (Fig. 5(b)) the sound material is pre-filtered in real-time by the equalisation filter and then it is reproduced via the normal audio chain into the room.

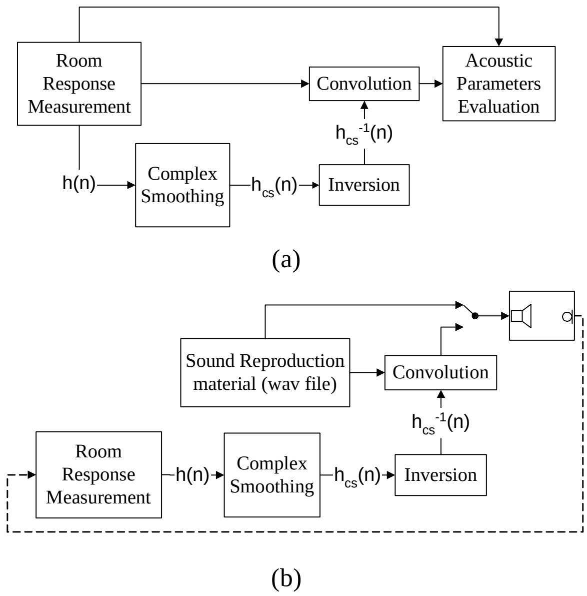


Figure 5: Implementation scheme for the proposed equalization method: (a) objective analysis (b) subjective analysis.

2.2. Objective acoustic tests

It is well-known that the reverberation properties of rooms can be derived from established acoustic criteria calculated from any measured impulse response. The evaluation and the comparison between such criteria for both the original and the equalised responses, yields an objective assessment of how the proposed method affects the room acoustic properties.

In the time domain such analysis criteria are the sound energy ratios in various forms and the reverberation time, which have been widely used to quantify and predict clarity and reverberance in room acoustics. Specifically, four time domain criteria were employed after analysing the measured and equalised responses, the “Early Decay Time” (EDT – 10dB), which can be considered as a special case of the “Reverberation Time” of the room, the “Clarity for 80 msec” (C80), which is a special case of the “Early / Late ratio” criterion, the “Definition” (D50), which indicates the percentage of the perceptually critical sound energy, and the “Centre Time” of the response (TS).

In the frequency domain, the Standard Deviation (variance) J of the modulus of the Tranfer Function was employed for both the original and equalized responses [1].

In two spaces, specifically in Room 2 and Room 6 (see Table 1), the above objective acoustical parameters were evaluated for responses derived from a second receiver position, closer to the source. These responses were equalized using the same inverse filter used to equalize the initial room response, assessing in this way potential robustness of the proposed method over different source / receiver positions within the room.

2.3. Subjective listening tests

The objective testing of the proposed method was augmented by listening tests indicating possible perceptual improvement for the processed signals.

The listening tests were carried out in the space shown in Figure 4. The loudspeaker/listening room direct evaluation, being a realistic sound reproduction situation, was chosen in preference to listening through headphones. Of importance was to correlate subjective to objective results in time and frequency domains, as well as obtaining an overall preference rating for the original or equalized signals. Hence, a “Clarity” criterion, together with a “Spectral Balance” criterion were specified, in a scale ranging from -3 (worst) to 3 (better), together with an “Overall Preference” criterion, as is shown in Table 2.

Clarity				No change			
Spectral Balance				No change			
Rating	*-3*	*-2*	*-1*	*0*	*1*	*2*	*3*
Overall preference	First signal preferable			No change	Second signal preferable		

Table 2: Rating scales for the AB listening tests

From the alternative listening test design methods [15-17], the AB method was chosen and subjects were presented with two signals: the original (A) and the filtered (B) signal, being asked to evaluate the last signal of each pair compared to the first one. In order to minimize bias, the subjects were presented with an equal number of both AB and BA sequences. All the sequences were repeated for as many times as a subject would require.

Five pieces of music representing pop, rock, jazz, classical orchestral and piano music and one male anechoic speech signal were used as test sequences. Listeners were mainly casual listeners of music and audio equipment, without special expertise in similar equalization techniques.

3. RESULTS

3.1. Objective test results

All tested and well-established objective acoustic criteria, were found to improve for all spaces, after the application of smoothed response equalization, implemented as was shown in Figure 5 Specifically, the Clarity (C80) and Definition (D50) criteria, known to describe the perceived acoustic effects on music and speech presentation, seemed to improve, irrespective of the room’s volume, as is shown in Figures 6 and 7.

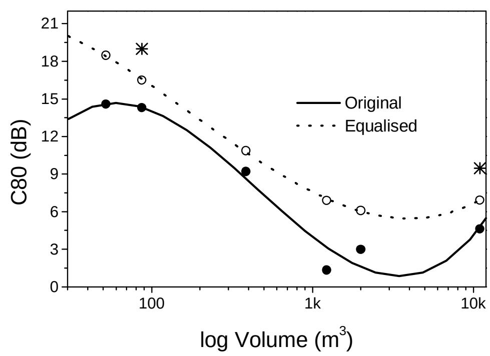


Figure 6: Clarity (C80) vs Volume of measured acoustic space responses before and after equalization.

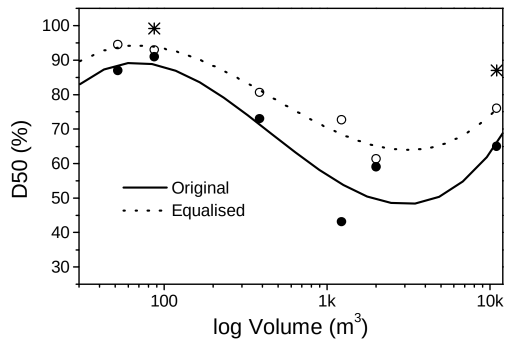


Figure 7: Definition (D50) vs Volume of measured acoustic space responses before and after equalization.

The criteria mainly describing the space’s overall reverberance, such as Early Decay Time (EDT) and Centre Time (TS), were also improved after equalization, but the improvement was found proportionally to increase for the larger and more reverberant spaces (Figure 8 and 9).

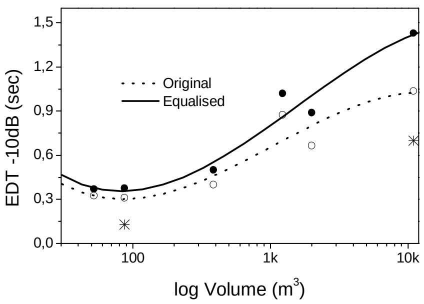


Figure 8: Early Decay Time (EDT –10dB) VS Volume of measured acoustic space responses before and after equalization.

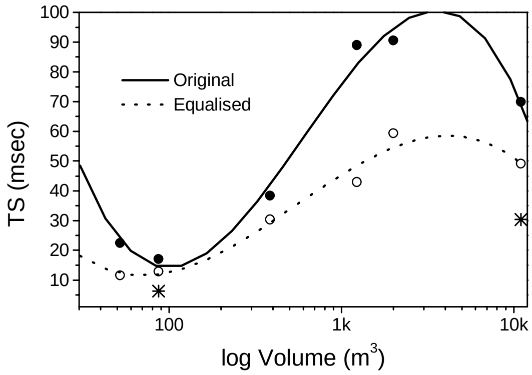


Figure 9: Centre time (TS) vs Volume of measured acoustic space responses before and after equalization.

The Room Transfer Function amplitude Spectral Deviation was also found to be reduced after equalization, for all spaces, by an amount generally proportional to the degree of the room's original spectral irregularity, which was not always increasing with its volume (Figure 10).

Significantly, for a number of cases, smoothed response equalization was carried out using inverse filters designed from measurements at different positions within the same space (though for the largest space, a 1000-seat concert hall, this difference

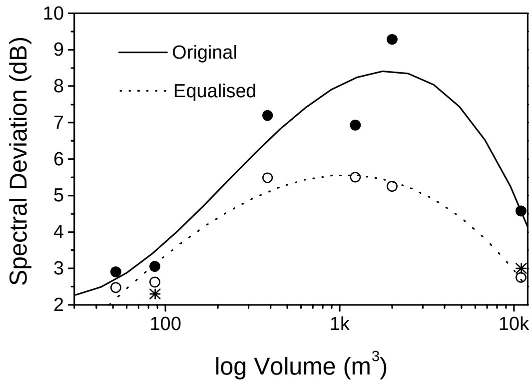


Figure 10: Spectral Deviation vs Volume of measured acoustic space responses before and after equalization.

in positions could be up to 10 m). Even for these tests, it was found that equalization improve the acoustic criteria, as can be observed in Figures 6-10, where such results are indicated with star (*) pointers. A theoretical and experimental analysis of this aspect of the performance of the proposed method, will be a topic of a future publication.

3.2. Subjective test results

Analysis of the listener questionnaires, expressed in percentage scale, has produced the results shown in Table 3.

	Clarity	**Spectral Balance**	**Overall preference**
Percentage improvement or Overall preference	15 %	20%	58%

Table 3: Total percentage improvement or preference for the equalized signals, during the subjective listening tests.

These results indicate a small, but nevertheless significant improvement and preference for the equalized signal over the original. These results must be also considered as promising, on the light of the little previous experience and expectancy of any listener with such reduced acoustic distortions within a space for which he / she has already formed a preconceived subconscious and subjective acoustic image.

4. DISCUSSION AND CONCLUSIONS

The realization of digital acoustic equalization methods, requires the use of a special class of signal processing, which is optimised largely following perceptual considerations. Room Transfer Functions vary so dramatically from point to point within the same enclosure [19], so that it is unrealistic to consider that ideal equalization filters could be designed for all such potential source / receiver positions. In addition, direct response inversion of the mixed-phase room response via the introduction of a delay function [18], can suffer from perceptually disturbing time / frequency domain artifacts related to pre-echoes and ringing poles compensating for the original spectral dips. These reasons alone, not even considering potential implementation benefits, call for the initial pre-processing of the measured room responses, to generate versions of reduced spatial, spectral and time complexity, but still capable of reducing phase, transient and spectral distortions introduced by room acoustics. Design of inverse filters for such functions, would not achieve a theoretically-perfect room acoustics deconvolution, but would realize a good compromise between reduction of perceived reverberation effects and the introduction of inaudible processing artifacts. Such a method, based on Complex Smoothing, was employed and experimentally verified here.

The results, based on the effect of the equalization on well-established objective acoustic criteria, have shown that improvement was measured in all cases, irrespective of the size and the acoustics of the original space, which for these test ranged from small offices to a 1000-seat auditorium. Preliminary results, also shown here, indicate that such improvements can be achieved not to the detriment, but indeed to the benefit of reproduction in other positions within the same enclosure. Hence, the proposed method achieves a space and position-independent reduction of measured distortions imposed by reverberation and room acoustics on signals. Apart from these performance benefits, the technique can be efficiently implemented via any sample or block-based FIR convolution architecture, not requiring specialist design for multiband / multirate filtering or other intermediate processing stages.

It is also promising that such objective improvements could be also confirmed in a real-time listening situation with a group of largely non-expert listeners.

5. ACKNOWLEDGMENT

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