Real-Time Room Equalization based on Complex Smoothing: Robustness Results

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ABSTRACT

The aim of this study is to investigate the robustness of room acoustics real-time equalization using inverse filters derived from the Complex Smoothing of the Transfer Function using perceptual criteria. The robustness of the method is assessed by real-time tests which compare the performance of Complex Smoothing-based equalization (for different filter lengths) with the traditional, ideal inverse filtering, over a range of room locations, which differ to the ones where response measurements were taken. Objective measurements and audio examples will show that the Complex Smoothing-based equalization performance is largely immune to position changes and does not introduce processing artifacts, problems affecting the traditional ideal inversion.

0. INTRODUCTION

Many aspects of room acoustics equalization (dereverberation) have been recently brought to attention [1-3]. Due to this renewed interest, it is now widely accepted that the highly irregular and variable nature of Room Transfer Functions causes direct application of filters derived from their theoretical inverses to fail during practical tests. It is also becoming increasingly obvious that response pre-processing and modification prior to design of such filters offers a theoretically suboptimal but nevertheless a perceptually more compliant and hence practically viable solution to such problems. An obvious approach for such pre-processing is by "conditioning" the measured RTFs, so that sharp dips due to response zeros are partially ignored during the design of the inverse filters, so that the compensating poles of such inverse functions are not generating post- or pre- echo artifacts after equalization [4-8]. These authors have been concerned with the equalization methods that would compensate for both spectral and time domain artifacts of room reverberation [9] and for some years now have introduced and tested response pre-conditioning based on Complex Smoothing [10]. In an earlier paper [11], objective and subjective results have been presented showing that such a method would improve all measurable acoustic criteria (e.g. Early Decay Time, Centre Time, Clarity, Definition), as well as Spectral Deviation, for spaces ranging from office to large auditorium of varying size (from 52 m^2 up to 11000 m^2) and properties. In this earlier paper, the authors - as is the case of the great majority of comparable works - have employed a simulated (off-line) methodology. That is, during those previous studies, the spaces under test had their impulse responses measured and then any filtering and analysis was carried in a "non real-time" sense, i.e. within the computer.

This work presents tests conducted in-situ: test signals were pre-filtered in real-time by the equalization filters and the resulting signals were measured at specific positions. Hence, this work presents results which examine and highlight a number of additional important aspects of room equalization:

(i) the mismatch of the theoretically ideal equalization between simulated (off-line) or real-time tests.

(ii) the good matching of Complex Smoothing-based equalization performance during simulated (off-line) or real-time tests.

(iii) the position-independence and robustness of Complex Smoothing-based equalization performance, even when filter for a specific position were employed at different positions within the room. In contrast, the theoretically ideal equalization achieves a poor performance in this respect (as is well known).

(iv) the increasing mismatch of he theoretically ideal equalization between simulated (off-line) or real-time tests, for increasing inverse filter lengths. In contrast, Complex Smoothing-based equalization performance remains largely independent of filter length and achieves satisfactory results for sensible number of FIR filter coefficients.

Some of these important aspects are fully described and analyzed in a lengthier document [12]. Here, the theoretical principles of the above-mentioned schemes are briefly described (Section 1), and detailed experiments and results are presented in Sections 2 and 3 respectively. Conclusions are discussed in Section 4.

1. THEORY

1.1. Room response equalization methods

1.1.1. Ideal inversion

An "inverse filter" $h_i(n)$ with length L_i (samples) has to be introduced in the processing path, such that a new response $d_i(n)$ will be produced, according to the following expression:

$$d_i(n) = h(n) \otimes h_i(n) \cong \delta(n), n = 0,1,\ldots, L_d \quad (1)$$

where: $h(n)$ is a previously-measured room impulse response of length L (samples), $\delta(n)$ is the ideal impulse response, $L_d = L + L_i - 1$ (samples) is the length of the inverted response and the symbol $\otimes$ denotes the discrete-time linear convolution. The inverse filter response $h_i(n)$ was calculated here to have the best (least-square) approximation for the ideal response $\delta(n)$ [13].

1.1.2. Room response spectral Complex Smoothing and inversion

The principles of such equalization have been established in earlier papers [10, 11]. The Complex Smoothed room discrete-frequency response $H_{cs}(\omega_k)$ is transformed into a corresponding reduced-order smoothed room impulse response $h_{cs}(n)$. Then, an inverse filter $h_{csi}(n)$ is evaluated, which inverts the Complex Smoothed response, i.e.:

$$d_{cs\,i}(n) = h_{cs}(n) \otimes h_{csi}(n) \cong \delta(n) \quad (2)$$

In theory such an inverse filter response $h_{csi}(n)$ will achieve a compromised result when employed on the original measured room response $h(n)$, according to the following expression:

$$d_i(n) = h(n) \otimes h_{csi}(n) \quad (3)$$

1.2. Room response equalization tests

Based on the two alternative design procedures for the inverse filter $h_i(n)$ (i.e. by the ideal or the complex smoothing-based inversion algorithms defined in the previous subsections), the 2 different test procedures shown below for assessing inversion performance can be adopted:

1.2.1. Off-line room response equalization

The test chain for such inversion is shown in Figure 1, where it can be observed that all processing is carried-out within a simulated computer environment, employing a previously-measured room response. The majority of results presented in the room equalization literature were derived by such off-line tests.

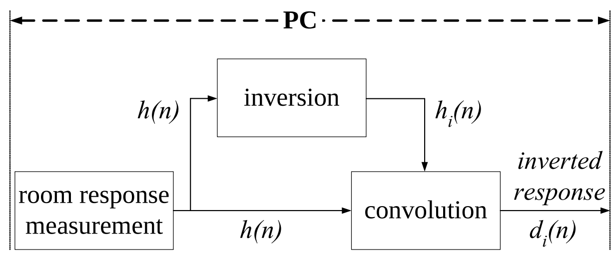

Figure 1 Off-line room impulse response inversion

1.2.2. Real-time room response equalization

In this case, as it is shown in Figure 2, the excitation signal $s_{te}(n)$ used as source for the room response measurement, is pre-filtered in real-time by the (previously-measured room response $h(n)$) inverse filter $h_i(n)$ which is designed by the ideal or the complex smoothing-based inversion algorithms.

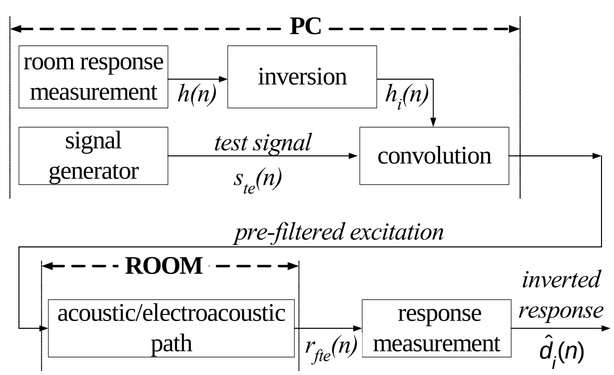

Figure 2 Real-time room impulse response inversion

After real-time reproduction in the room, the recorded excitation signal $r_{fte}(n)$ is described by the following expression:

$$r_{fte}(n) = s_{te}(n) \otimes h_i(n) \otimes h(n) \qquad (4)$$

The above signal is appropriately processed (usually via deconvolution of the excitation signal $s_{te}(n)$) yielding the measured real-time inverted discrete-time room response $\hat{d}_i(n), \quad n = 0,1,\ldots,L_d$.

1.3. Assessment of room equalization performance

For the objective assessment of the room equalization performance in the time domain, the following criteria were employed:

(i) the "Equalization Error Energy" J (dB), defined for both off-line and real-time equalization as:

$$J = 10\log_{10}\left\{\frac{1}{L_d}\cdot\sum_{n=0}^{L_d-1}\left[e(n)\right]^2\right\} \qquad (5)$$

where $e(n)$ is the "Equalization Error" i.e. the time domain difference between the desired ideal response $\delta(n)$ and the off-line ($d_i(n)$) or real-time ($\hat{d}_i(n)$) inverted response.

(ii) the "Clarity for 50 msec" criterion (C50) as a special case of the "Early / Late ratio" criterion i.e. the ratio (in dB) between the energy of the early-arriving reflections (including the direct sound up to 50 msec) and the energy of the late reflections.

(iii) the "Equalization Spectral Deviation" i.e. the standard deviation V (dB) of the DFT modulus of any measured or equalized response, used as an objective measure of the spectral flatness of the original or the inverted responses.

Most other known acoustic criteria have been also tested wielding similar results to those 3 measures, but due to space restrictions, are not included in this discussion.

2. EXPERIMENTAL PROCEDURE

Equalization tests were carried-out for both simulated (off-line) and real-time experiments, as described in Section 1, within two different spaces:

(i) a classroom (referred to as Room 1) with dimensions L:10.20m X W:7,05m: X H:2.65m, with Reverberation Time, RT = 1.1 sec (average).

(ii) a professional laboratory room (referred to as Room 2) with dimensions and acoustics close to those recommended by the IEC 268-13 standard for loudspeaker evaluation, i.e. L:7,15m X W:4,60m X H:2,90m. The Reverberation Time was RT=0.368 sec (average).

For the response measurements and tests, a self-powered 2-way loudspeaker (ATC SCM 20-2A) and an omnidiractional microphone were used, placed at a height of 1.5 m from the floor. The variations in placement are given in Table 1.

Test condition	***Description***	***Source / receiver distance (m)***
Room 1 / Position 1	far position in classroom	6.6
Room 1 / Position 2	near position in classroom	2
Room 2 / Position 1	far position in laboratory	4
Room 2 / Position 2	near position in laboratory	2

Table 1 Source / receiver placement details.

The excitation signal in both cases was a maximum-length sequence (MLS) reproduced and averaged 16 times to reduce the effects of additive background noise. The original room responses were derived from calculations based on double-precision floating-point arithmetic.

For each of the above source/receiver placements the room response was obtained to a length of L=256K samples. The real-time inverted system responses were also obtained using the above method, as is described in Section 1.2. Pre-filtering was applied on these test (excitation) sequences by using the appropriate inverse filters, having different length L_i ranging from 1K to 32K samples. These inverse filters were designed to compensate for the measured room responses corresponding to the far positions in each room (i.e. Room 1 / Position 1 and Room 2 / Position 1, see Table 1). For the case of the closer placements of the microphone (i.e. Room 1 / Position 2 and Room 2 /

Position 2), the excitation signal was pre-filtered by the inverse filters designed for the previously-described far positions. This test would therefore indicate the robustness of the equalization method to source / receiver position variations.

The measured real-time inverted responses were trimmed to an analysis length $L_d = L + L_i$ (samples). Off-line inverted responses were also obtained by linear convolution within the computer, between the original room response (for each of the cases described above) and the corresponding inverse filters.

3. RESULTS

Each of the 3 figures presented here, displays results related to one specific criterion: Figure 3 shows the results related to the Equalization Error Energy, Figure 4 depicts results of frequency domain Equalization Spectral Deviation, and Figure 5 demonstrates results for the Clarity criterion. For each of these figures, the first four plots ((a)-(d)) provide results for all types of equalization schemes as a function of filter length with filters designed for Position 1. Plots (e) and (f) show similar results for filters of length 4K, as function of different source / receiver distance, with filters implemented for Position 1. From these results, the following conclusions can be drawn:

(i) For the case of Ideal equalization, the Off-line and Real-time tests provide significantly different performance results. Specifically, for Ideal inverse filters, in all cases the performance deteriorates for larger filters, this being more evident for the time-domain criteria than for the frequency domain criteria. Reasons for this discrepancy are discussed in detail in [12].

(ii) Unlike the Ideal equalization, Complex Smoothing equalization yields stable performance which is largely independent of filter length (optimal results are for filters under 4K), and which does not vary between for Real-time and Off-line implementation. In this sense, the large volume of off-line equalization results presented previously by the authors for a large number of different spaces [11], would be also valid during practical real-time tests.

(iii) The Ideal equalization performance deteriorates dramatically when a filter designed for one position within the room is employed for equalization at a different position (e.g. Room 1 / Position 2 and Room 2 / Position 2), for both Off-line and Real-time tests. This finding reconfirms earlier work [9, 14]. In contrast, Complex Smoothing equalization presents stable improvement for all criteria irrespectively to the receiver's position during both Off-line and Real-time tests. The robustness of the Complex Smoothing equalization method is more obvious for the Clarity criterion especially in larger rooms (Figure 5). Specifically, this criterion improves for both the position for which the filter has been designed, as well as for the different position where the same filter is used. The improvement is more evident for the larger (in volume) room, where obviously the original reverberance and degradation is higher prior to equalization and is maintained during both Real-time and Off-line testing.

4. CONCLUSIONS

This study has shown that the Ideal room equalization scheme, when it is implemented in practical real-time conditions, seems to offer marginal improvement, only for the case when inverse filters are of relative short length (observing the results for the objective criteria presented in previous sections) and only when these filters have been designed to compensate for the measured room response in a specific position. Nevertheless, such marginal improvement is further reduced in significance, given that from a subjective point of view such short Ideal inverse filters would lack in frequency resolution to provide sufficient low frequency compensation.

In contrast, the Complex Smoothing-based inverse filters have been designed to follow psychoacoustic criteria (having progressively reduced frequency resolution from low to high frequencies) and hence they can compensate for the full range audio spectrum. Such filters achieve improved equalization performance in all cases (off-line and real-time equalization irrespective of filter length) not only for the position in which the have designed to compensate, but also for other receiver placements inside the room. It is also significant that the Clarity criterion improves with such filters especially for the larger rooms, something which also observed in earlier tests for a 1000-seat auditorium [11] (see also audio demos in the Audiogroup site:
http: //www.wcl.ee.upatras.gr/audiogroup/Equalization).

Hence this study has confirmed that Complex Smoothing-based equalization achieves robust performance during real-time in-situ tests, for filters of relative short size (up to 4K coefficients). It also appears that such a method may be appropriate not only for equalization of home listening spaces but also for professional applications such as Public Address audio systems.

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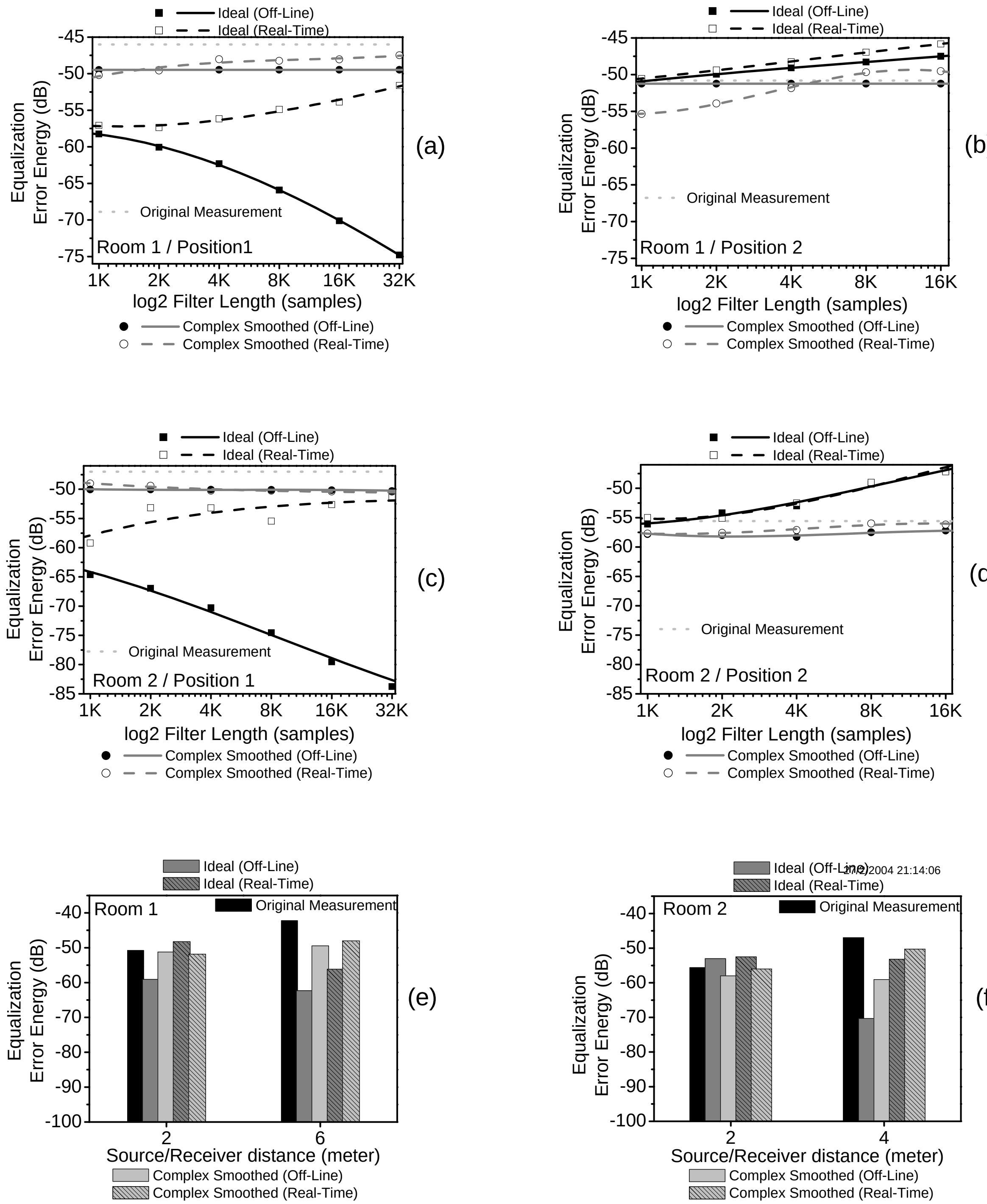

Figure 3 Time domain results for Real-time and Off-line equalization tests based on Ideal and Complex Smoothed inversion: (a)-(d) Equalization Error Energy vs inverse filter length; (e) and (f) Equalization Error Energy vs. source/receiver distance (for inverse filter length 4K samples). All filters were designed from a measurement for Position 1.

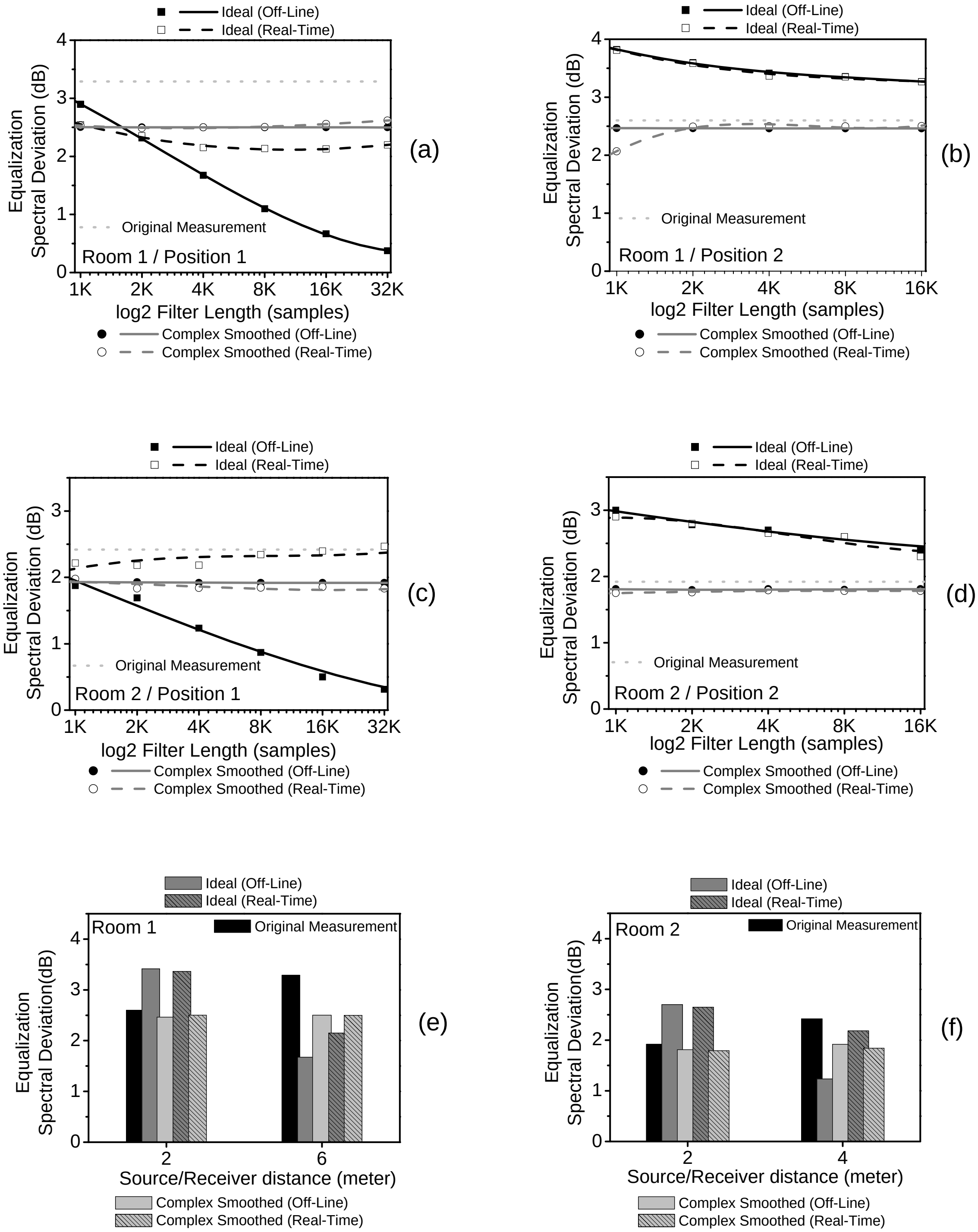

Figure 4 Frequency domain results for Real-time and Off-line equalization tests based on ideal and Complex Smoothed inversion: (a) - (d) Equalization Spectral Deviation vs inverse filter length, (e) and (f) Equalization Spectral Deviation vs. source/receiver distance (for inverse filter length 4K samples). All filters were designed from a measurement for Position 1.

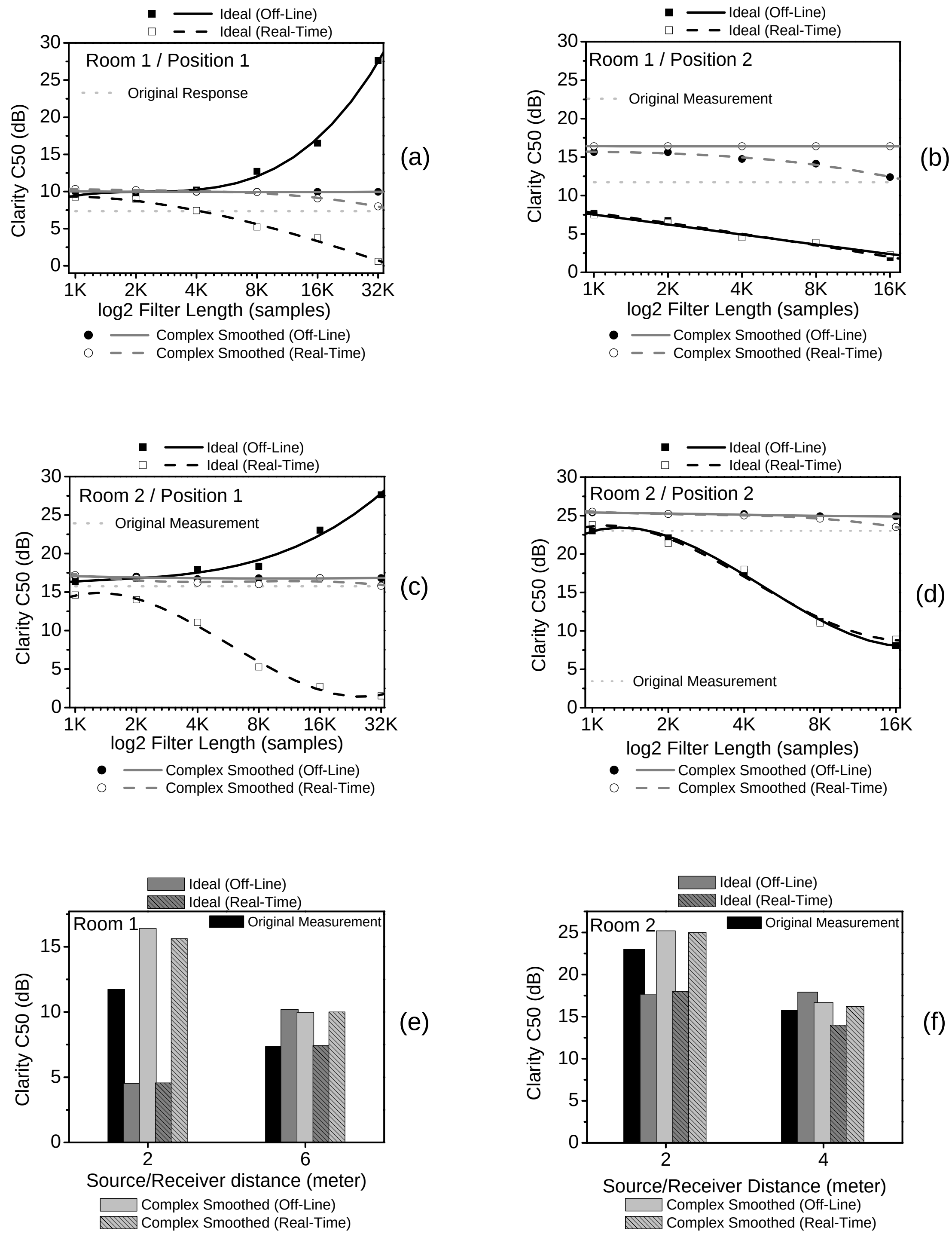

Figure 5 Clarity (C50) for Real-Time and Off-Line equalization tests based on Ideal and Complex Smoothed inversion: (a)-(d) Clarity vs inverse filter length, (e) and (f) Clarity vs source/receiver distance (for inverse filter length 4K samples). All filters were designed from a measurement for Position 1.