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Combined quasi-anechoic and in-room equalization of loudspeaker responses

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ABSTRACT

This paper presents a combined approach to loudspeaker/room response equalization based on simple in-room measurements. In the first step, the anechoic response of the loudspeaker, which mostly determines localization and timbre perception, is equalized with a low-order non-minimum-phase equalizer. This is actually done using the gated in-room response, which means that the equalization is incorrect at low frequencies where the gate time is shorter than the anechoic impulse response. In the second step a fractional-octave resolution minimum-phase equalizer is designed for correcting the low-frequency part of the response, based on the in-room response pre-equalized with the quasi-anechoic equalizer. This second step, in addition to correcting the room response, automatically compensates the low-frequency errors made in the quasi-anechoic equalizer design when using gated responses.

1. INTRODUCTION

The digital equalization of loudspeaker/room responses has been subject of research for around three decades. Equalizing the anechoic response of the loudspeaker [1, 2] is a relatively simple problem. In [2] it was shown that perceptually good equalization can be achieved by FIR filters as short as 80-90 taps. Due to the relative shortness of anechoic responses, mixed-phase (or, time-domain) direct inversion produces reliable results. However, the importance of anechoic equalization should not be underestimated: the anechoic response of a loudspeaker has a very important effect in the perceptual assessment of sound quality, since it determines the direct sound which reaches the listener, influencing timbre perception [3]. Naturally, the requirement for such a correction is the measurement of the anechoic response.

On the other hand, the correction of the loudspeaker/room response is much more problematic because the room response is largely non-minimum-phase and position dependent. Position dependency means that the direct inversion at one point usually worsens the response at other positions [4, 5]. A simple and robust solution to the problem is designing equalizers with fractional-octave resolution which lead to more precise correction at low frequencies, where the variation with respect to space is smaller [4, 5]. Moreover, when measurements at multiple positions are available, then the common trend of the responses can be computed that is used for correction [6, 7, 8]. Problems coming from the non-minimum-phase nature of the room response are most often avoided by correcting the magnitude response only, by the use of minimum-phase equalizers. This eliminates the pre-echoes that would arise with direct inversion. When we desire to correct the phase response as well, care has to be taken that only the common (not position dependent) non-minimum-phase zeros are compensated [7]. Moreover, on the contrary to a flat anechoic loudspeaker response, a flat room response does not lead to good sound. Typically, a target with a downward slope is preferred, and it is also possible to compute the target curve by taking into account the estimated directivity index of the loudspeaker [6].

These two approaches can be combined by first equalizing the loudspeaker based on the anechoic response with a non-minimum-phase equalizer, and then correcting the room response with a minimum-phase equalizer to have the best of the two worlds [4]. The difficulty in the method of [4] lies in the separation of the contributions of the room and the loudspeaker.

2. COMBINED EQUALIZATION

The simple method proposed in this paper is based on the idea of [4], but realizes that actually there is no need to completely separate the room and the loudspeaker response. This simplifies the design and eliminates the need of anechoic measurements, which are hard to achieve at home environments. The basic idea of the method is that the quasi-anechoic loudspeaker response is corrected at mid- and high frequencies, while at low frequencies where the loudspeaker response blends with the room they are corrected jointly. In this section only the basic implementation of the method is discussed, possible enhancements are left for Sec. 4.

2.1. Quasi-anechoic loudspeaker equalization

The method starts with equalizing the anechoic loudspeaker response with a non-minimum-phase equalizer.

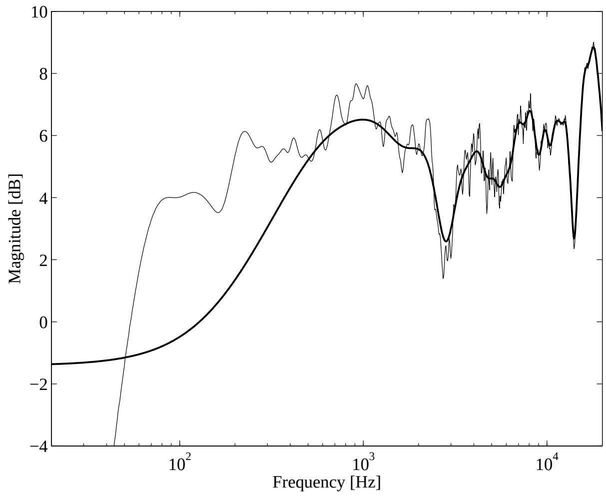

Fig. 1: On-axis frequency response of a loudspeaker measured in an anechoic chamber: full response (thin line) and the frequency response computed after gating with a 2 ms long half Hann window (thick line).

Since the anechoic response is not available, a gated version (up to the first reflection) of the in-room response is used, from which a short (50-100 tap) FIR equalizer is designed in the time domain.

Gated loudspeaker measurements only follow the anechoic loudspeaker response at mid- and high-frequencies, since a significant amount of energy is lost at low frequencies where the impulse response is longer than the gate time [9, 10]. This is illustrated in Fig. 1 for a two-way floorstading loudspeaker, showing that the response gated with a 2 ms half Hann window (thick line) describes the anechoic behavior relatively well down to 500 Hz, but below it is completely inaccurate. One option to address this shortcoming is to perform an additional nearfield measurement at low frequencies and crossfade between the nearfield and gated responses [9]. It is also possible to shorten the measured response with a suitable filter before gating [10, 11], improving low-frequency accuracy. However, in our application this is not necessary, because any low-frequency error we make by gating will be corrected together with the room response in the in-room equalization step of Sec. 2.2.

The first 150 samples of the in-room impulse response $y(n)$ measured at a 2 m distance from the loudspeaker are shown in Fig. 2 (a), thin line (the speaker is the same as for Fig. 1). The floor reflection is clearly visible at around $n = 80$, and this is eliminated by gating the response with a 88 sample (2 ms) long half Hann window. The gated response $y_{\mathrm{g}}(n)$ is shown by thick line in Fig. 2 (a). The target response for the quasi-anechoic equalizer design is a delayed unit pulse $y_{\mathrm{lst}}(n) = \delta(n-D)$, shown by dotted line[1]. The modeling delay D is necessary for proper equalization due to the non-minimum-phase nature of the loudspeaker response [1].

Now the task is to design an equalizer with an impulse response $h_{\mathrm{lseq}}(n)$ such that the error between the target $y_{\mathrm{lst}}(n)$ and equalized gated response $y_{\mathrm{g,eqd}}(n) = h_{\mathrm{lseq}}(n) * y_{\mathrm{g}}(n)$ is minimal. Evaluating the error in the mean squared sense

$$e = \sum_{n=1}^{N} (y_{\mathrm{g,eqd}}(n) - y_{\mathrm{lst}}(n))^2 \qquad (1)$$

leads to the well known normal equations [1] and the solution is given in a closed form

$$\mathbf{h}_{\mathrm{lseq}} = (\mathbf{M}^T\mathbf{M})^{-1}\mathbf{M}^T\mathbf{y}_{\mathrm{lst}}, \qquad (2)$$

where $\mathbf{y}_{\mathrm{lst}}$ is a column vector composed of the samples of the target response, the columns of the modeling signal matrix $\mathbf{M}$ contain the gated loudspeaker response and its delayed versions, and finally $\mathbf{h}_{\mathrm{lseq}}$ is a vector containing the impulse response of the equalizer. In this case a 70 sample long equalizer is estimated, and the result is shown in Fig. 2 (b). The equalizer is directly implemented as an FIR filter, whose magnitude response is shown in Fig. 4 (d).

As expected, the FIR equalizer results in an almost perfect pulse and thus a constant magnitude response when applied to the gated response from which it was designed (not shown). Figure 2 (c) displays the ungated in-room response $y(n)$ filtered by the FIR equalizer, showing that the time-domain response is practically a unit pulse up to the first reflection.

2.2. In-room equalization

[1]Note that if the loudspeaker has a significant high-frequency roll-off, a low-pass type of target response should be used to prevent the unnecessary boosting of high-frequencies. The same reasoning would require a target whose low-frequency behavior takes into account the capabilities of the loudspeaker. However, this is not critical, because that will be taken into account in the in-room equalizer design in the next step.

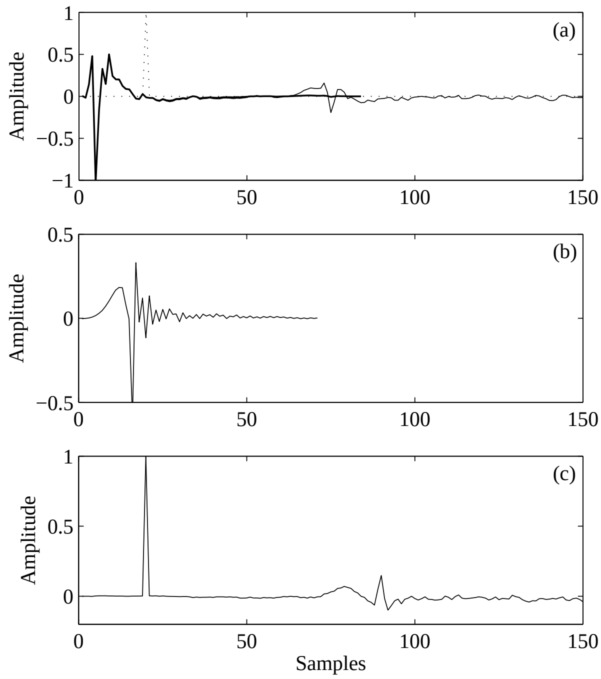

Fig. 2: Quasi-anechoic loudspeaker equazlization: (a) the in-room impulse response (thin line) together with the gated impulse response (thick line) and the target (dotted line), (b) the impulse response of the FIR equalizer, and (c) the equalized in-room impulse response. Sampling frequency is $f_s = 44.1$ kHz.

In the second step, we apply the quasi-anechoic FIR equalizer to the complete (non-gated) loudspeaker/room response $y_{\mathrm{eqd}}(n) = h_{\mathrm{lseq}}(n) * y(n)$. The in-room magnitude response of the loudspeaker $Y(f) = \mathcal{F}\{y(n)\}$ is displayed in Fig. 3 (a), while Fig. 3 (b) showns the response after the quasi-anechoic equalizer is applied $Y\mathrm{eqd}(f) = \mathcal{F}\{y_{\mathrm{eqd}}(n)\}$.

Next, we design a traditional, fractional-octave resolution minimum-phase equalizer to correct the magnitude of this pre-equalized response $Y\mathrm{eqd}(f)$ at low frequencies. The smoothed frequency response after 1/12th-octave power smoothing is displayed in Fig. 3 (c), which is crossfaded to flat response at a crossover frequency $f_{\mathrm{c}} = 500$ Hz, shown in Fig. 3 (d). This crossfade is used to limit the frequency range of in-room equalization.

The choice of using f_{c} is limited by the fact that the role of the in-room equalizer is not only to correct the room

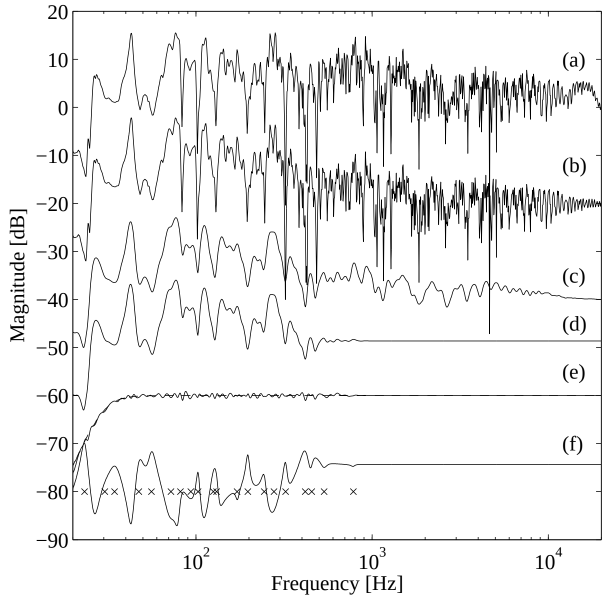

Fig. 3: In-room loudspeaker equalization: (a) the in-room response of the loudspeaker, (b) equalized by the quasi-anechoic FIR equalizer, (c) the equalized response sixth-octave smoothed, (d) crossfaded to constant gain above $f_{\mathrm{c}} = 500Hz$, and finally (e) equalized by the in-room equalizer, together with the target shown by dashed line. The equalizer is implemented as a 40th-order fixed-pole parallel filter, whose frequency response is shown in (f), together with its pole frequencies (crosses).

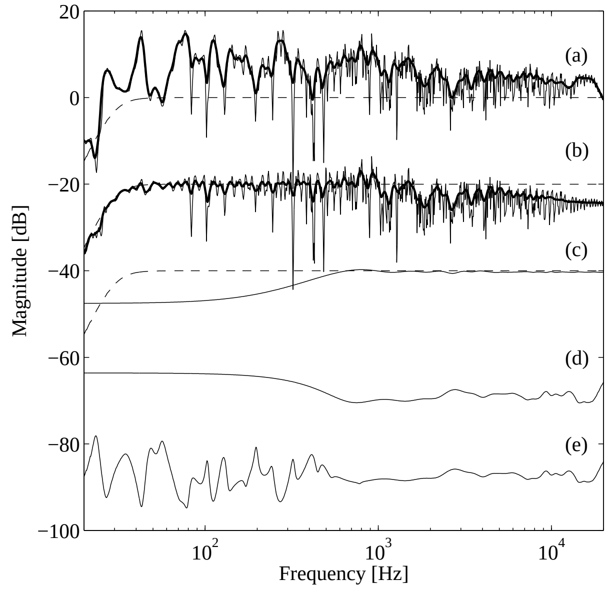

Fig. 4: The combined loudspeaker equalization: the in-room magnitude response (a) unequalized and (b) equalized by the quasi-anechoic and in-room equalizers jointly. Thick lines show the 1/12th-octave smoothed transfer functions. The magnitude response computed from the early part of the equalized impulse response is shown in (c). The dashed lines display the target response. The frequency responses of the quasi-anechoic FIR equalizer and the complete equalization chain are displayed in (d) and (e), respectively.

response, but to counteract the errors we made when we were using the gated response instead of the true anechoic response in Sec. 2.1. Coming from the 2 ms gate time and also from Fig. 1, the lowest possible f_{c} is 500 Hz. By looking at Fig. 3 (a), 500 Hz itself seems to be a good option, since below this frequency the strong modal behavior of the room is apparent, while above the large modal overlap leads to a less perceivable room influence.[2]

In this example a 40th order fixed-pole parallel filter [12] was designed based on the minimum-phase version of the smoothed-crossfaded response of Fig. 3 (d). The poles of the parallel filter were identified using warping-based pole positioning with $\lambda = 0.98$ [13]. The equalizer response is shown in Fig. 3 (f), while the pole positions of the filter are indicated by crosses. The target is a forth-order high-pass filter with a cutoff frequency of 30 Hz, displayed by dashed line in Fig. 3 (e). The equalized response is shown in Fig. 3 (e), solid line, following the target accurately.

2.3. The resulting equalization

The result of the combined equalization is displayed in Fig. 4. The unequalized in-room response is shown in

[2] This 2 ms gate time and thus the 500 Hz lower frequency limit comes from the fact that an in-room measurement with a 2 m distance was used for computing the gated response, with the benefit that the whole equalizer process is based on a single measurement. This makes the comparison with traditional, smoothing-based methods using a single response more fair in Sec. 3. In practice it is advised that the user makes an additional in-room measurement at 1 m distance, since this practically doubles the time until the floor reflection and leads to around 4 ms gate time. Thus, the lowest possible f_c will be 250 Hz, giving greater flexibility in which frequency range we would like to correct the room behavior.

Fig. 4 (a), while the response after applying the full equalization chain is displayed in Fig. 4 (b). The thick lines indicate the 1/12th-octave smoothed versions of the corresponding transfer functions, and the target is displayed by dashed line.

It can be seen in Fig. 4 that the in-room equalizer results in a flat magnitude response below $f_\mathrm{c} = 500$ Hz and also counteracts the low-frequency errors induced by using the gated response for the quasi-anechoic equalizer design. Above f_c the magnitude response is not flat but has a downward slope, since in that range no room equalization is applied. The performance above f_c is evaluated by gating the equalized impulse response until the first reflection and taking its Fourier transform. The gated equalized response is shown in Fig. 4 (c), which is perfectly flat.

Finally, Fig. 4 (d) shows the magnitude response of the quasi-anechoic FIR equalizer, while Fig. 4 (e) displays the magnitude response of the complete equalization chain.

The total computational complexity of the equalization is low: the current example requires a 70th order FIR filter and a 40th order IIR filter (20 second-order sections), which is easily implemented in DSPs and in more powerful microcontrollers. In addition, the parameter estimation for the equalizer is simple and straightforward, since it only requires low-order least-squares filter design both for the FIR and IIR parts.

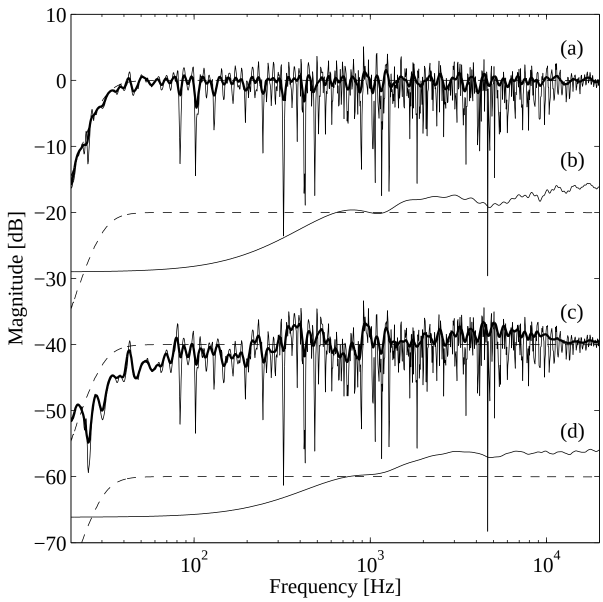

Fig. 5: Comparison with traditional 1/12th-octave equalization. Equalization based on the 1/12th-octave power smoothed in-room response: (a) magnitude response and (b) the magnitude response of the early part before the first reflection. Equalization based on the original (non-minimum-phase) 1/12th-octave complex smoothed in-room response: (c) magnitude response and (d) the magnitude response of the early part before the first reflection. Thick lines show the 1/12th-octave smoothed transfer functions and dashed lines display the target response.

3. COMPARISON

The standard way of designing a room equalizer is based on the fractional-octave smoothed in-room response. This section compares the combined quasi-anechoic in-room equalizer to this more traditional way of equalizer design.

First, a 1/12th-octave resolution minimum-phase equalizer is designed using the same in-room response as in Fig. 4 (a). This is done by computing the 1/12th-octave power-smoothed in-room response, making it minimum-phase, and designing a 60th order fixed-pole parallel filter width poles obtained from multiband warping [14]. The magnitude response after equalization is displayed in Fig. 5 (a), together with its 1/12th-octave smoothed version with thick line. As expected from a decent minimum-phase room equalizer, the equalized magnitude response follows the target properly. Figure 5 (b) shows the Fourier transform of the early part of the impulse response (up to the first reflection), indicating a high-frequency increase due to the equalizer. This is well in line with the practical experience that when designing an equalizer based on the in-room magnitude response, a flat target leads to an overly bright sound, and a target with a downward slope should be used.

Next, a 1/12th-octave resolution equalizer is designed based on the original (non-minimum-phase) in-room response. This is achieved by applying 1/12th-octave complex smoothing [15] to the measured response, then designing a 40th-order fixed-pole parallel filter with a 70th order parallel FIR part. A modeling delay is added to the high-pass target as in 2.1. The equalized magnitude response is shown in Fig. 5 (c). Now we see more problems: the low-frequencies are not correctly equalized because some low-frequency energy is lost due to the windowing effect of complex smoothing, and the early part still shows a high-frequency increase in Fig. 5 (d).

The first listening impressions about the proposed combined equalizer are very positive: it improves sound quality significantly by compensating the low-frequency problems in the room response while still leads to a natural, uncolored timbre. On the other hand, while both the minimum-phase and non-minimum-phase in-room equalizers correct the low-frequency room behavior, they result in a harsh sound, as expected from Figure 5.

Note that probably it is possible to choose a special target response for the in-room-only equalizers such that the resulting timbre is similar to that of the combined equalizer. Nevertheless, one of the benefits of the combined equalizer is that it seems to lead to an uncolored timbre without the need for target adjustments.

4. FURTHER ENHANCEMENTS

Section 2 considered the basic implementation of the combined equalization algorithm. Some of the possible further improvements are outlined below.

4.1. Automatic computation of the crossover frequency

In the current equalizer the only free parameter that needs to be adjusted by the user is the frequency f_c up to which the in-room equalization is applied. For certain applications it could be beneficial to find an automated way for choosing f_c based on the measured response. This requires gaining further understanding about the perceptual effect of changing f_c.

4.2. Smoothing and IIR filter design for the quasi-anechoic part

Since the 2 ms gate time used in the example already results in a smoothing of the transfer function with a 500 Hz wide window, it was assumed that no further smoothing is necessary. However, it would make sense to evaluate whether the fractional-octave complex-smoothing of the gated response increases the spatial robustness, especially when longer gate times are used. In addition, if even lower computational complexity is desired, IIR (or warped IIR) filter design can be utilized instead of the FIR filter.

4.3. Taking into account multiple measurements

While informal listening has not revealed any prominent artifacts outside the listening position, a formal investigation is needed to asses the spatial robustness of the proposed method. If needed, spatial dependency can be decreased by using multiple measurements in the equalizer design. For the quasi-anechoic equalizer part a couple of measurements of the loudspeaker at 1 m distance and in an angle corresponding to the listening window can be taken, windowed to the first reflection, then averaged. For the averaging, separate magnitude and phase averaging should be used as suggested in [16].

For the in-room equalizer part, multiple measurements can be taken into account similarly as usually done for single-input multiple-output room equalizers. For example, the common trend of the responses can be estimated by averaging the responses taken throughout the listening area [8], and additional measurements at random points around the room can be used to limit the gain during equalization [6].

5. CONCLUSION

This paper has presented a combined equalization approach based on quasi-anechoic and in-room measurements.

First a short FIR equalizer was designed based on the quasi-anechoic response of the loudspeaker. The quasi-anechoic response was obtained by gating the in-room response. Next, the complete in-room response was pre-equalized by the quasi-anechoic equalizer, and this response was used to design a fractional-octave minimum-phase room equalizer at low frequencies. In addition to correcting room effects, this second step also compensates for the errors made in the quasi-anechoic equalizer design due to gating.

As a result of the proposed equalization, at mid and high frequencies, where the direct sound determines timbre and localization, the equalized system will behave like if the equalizer has been designed from anechoic measurements, while at low frequencies where the loudspeaker response perceptually blends with the room response, their common minimum-phase equalization solves their problems jointly.

The implementation of the proposed method is straightforward and computationally light both for equalizer design and actual filtering.

Since this work is still in progress, a significant amount of tasks is left for future research. This includes formal evaluation of the spatial robustness of the basic algorithm of Sec. 2, the implementation and testing of the enhancements proposed in Sec. 4, and the thorough comparison with other equalization techniques via listening tests.

6. ACKNOWLEDGEMENT

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