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Comparison of Noise Compensation Methods for Room Acoustic Impulse Response Evaluations

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Summary

Noise that is part of the measured impulse responses is inevitable but can have a large impact on the evaluation of room acoustic parameters. Five well-known and widely used noise compensation methods are discussed and their performances are compared in this study. Two evaluation approaches are used to test the different methods: A simple parametric model to simulate the envelope of an impulse response including measurement noise, and in a second approach, special designed long-term measurements. These were conducted to be able to evaluate the errors as a function of the noise level. The results that are obtained using the model and the measurement approach are consistent with each other. When these methods are used to suppress noise effects, their performances differ significantly. This is also true for the three methods that are compliant with ISO 3382. Four methods cause systematic errors depending on the peak-signal to noise ratio. The reverberation time is more sensitive to noise than energy parameters such as clarity or definition. A comparison of the different excitation signals that are used for the measurement approach shows that there is no difference with regard to sine sweeps and maximum length sequences, if no impulsive noise or nonlinearities occur.

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1. Introduction

The acoustical properties of concert halls and opera houses are characterized by room acoustic parameters such as for instance reverberation time and clarity [1]. The precision of the measured parameters is important to compare different auditoria and to verify target values. In addition, the reverberation time is an essential quantity to determine other acoustical parameters, such as the absorption coefficient [2], the sound reduction index [3] or the scattering coefficient [4]. The precision of these (evaluated) parameters depends directly on the precision of the reverberation time determination. The reverberation time is derived from the impulse response in a post-processing step, standardized as the so-called Schroeder integration [5]. During every measurement of room impulse responses noise will occur and it can have a large impact on the evaluation of the parameter. The main component is the acoustical ambient noise in the room. Other components of electrical noise are introduced by the measurement chain (i.e. amplifier or analog-to-digital converter). In 2004 Katz carried out a round robin for room acoustic analysis software [6]. All participants (19 institutions with 25 different software packages) received the same measured room impulse response and reported the evaluated parameters. The differences between the software packages remained within the range or exceeded the subjective difference limen of the corresponding parameter. Katz pointed out that the variations occur mostly due to the noise in the impulse response. Another noise-related problem was the lack of indication for noise effects that are too high. Results for reverberation time can be obtained with most software packages, even if the signal to noise ratio is insufficient.

The effects of noise or techniques to suppress the noise were subject of a lot of research [7, 8, 9, 10]. There are various ways to minimize the influence of noise. ISO 3382 allows three different ways to handle the noise. Hak *et al.* [11] performed the first systematic analysis of a noise compensation technique depending on the noise level. Random white noise with different levels was added to a real measured impulse responses to simulate different signal to noise ratios. There are also alternative ways to calculate the reverberation time [12, 13, 14] that use nonlinear regression methods. However, these methods are rarely used and therefore not investigated in this study. In this study five established and commonly used noise compensation methods are selected and their performances are compared.

2. Noise Compensation Methods

All five investigated noise compensation methods use the impulse response as input, apply the compensation and supply the noise compensated energy decay curve (EDC, also known as backward integrated impulse response or Schroeder curve). The reverberation time is determined by

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performing a linear regression of the EDC in a defined dynamic range (for example from -5 dB to -35 dB for T_{30}). The clarity index, C_{80}, can also be calculated directly from the energy decay curve $E(t)$ at 80 ms:

$$C_{80} = 10 \log_{10} \left(\frac{E(0\,\mathrm{ms})}{E(80\,\mathrm{ms})} - 1 \right).$$

Other room acoustic parameters (such as the definition D_{50} or the sound strength G) can be calculated similarly from the noise compensated EDC, so that the calculation of all parameters benefits from the same noise compensation.

In the following subsections, the investigated methods will be described and illustrated using an artificial (model) impulse response. The impulse response represented a reverberation time of 2 s and a signal-peak to noise ratio of 40 dB. The model impulse response represents any measurement in the statistical region of the room transfer function where neither modal effects of the room nor effects of too narrow filter bands are present. The corresponding EDC is also shown to obtain the error that occurs due to the noise. Some methods require additional parameters of the impulse response (such as noise level, intersection time and late reverberation time). These are estimated using the iterative algorithm proposed by Lundeby *et al.* [9]. This algorithm has been proven to provide reliable results for real measured room impulse responses. The complete evaluation is fully automated and realized in Matlab. All described noise compensation methods are part of the ITA-Toolbox, an open source toolbox for Matlab [15]. Method A is the default method described in ISO 3382 [1]. Methods B and C are optional extensions also defined in the standard. Methods D and E use a noise suppression technique that is so far not described in ISO 3382.

2.1. Method A: Full Impulse Response

The full impulse response is taken for the backward integration and the upper integration limit is the length of the recorded impulse response t_{IR},

$$E(t) = \int_t^{t_{\mathrm{IR}}} p^2(\tau)\,\mathrm{d}\tau. \tag{1}$$

Technically speaking, this is not a noise compensation method as no noise compensation technique is used. The noise contained in the impulse response causes a large overestimation of the EDC (Figure 1). The size of the error increases with the length of the impulse response t_{IR}.

2.2. Method B: Truncation at Intersection Time

A commonly used noise compensation method is to truncate the impulse response at the intersection time t_i [8, 9, 1],

$$E(t) = \int_t^{t_{\mathrm{i}}} p^2(\tau)\,\mathrm{d}\tau. \tag{2}$$

The intersection time is the time where the exponential decay of the impulse response intersects with the constant

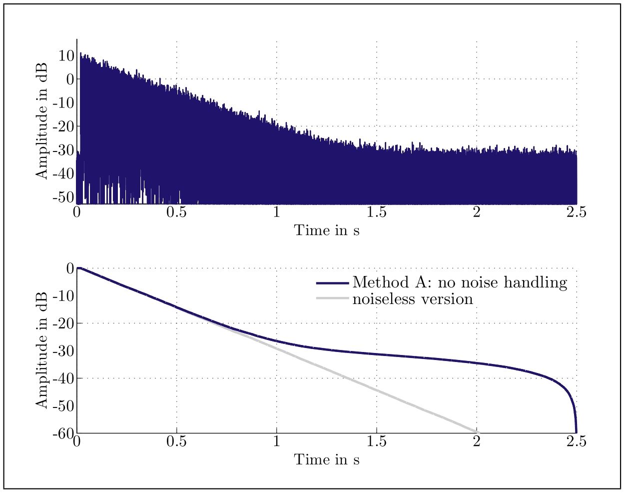

Figure 1. Example of an impulse response and corresponding energy decay curve when using the full impulse response for the backward integration (Method A). The last part of the energy decay curve is overestimated.

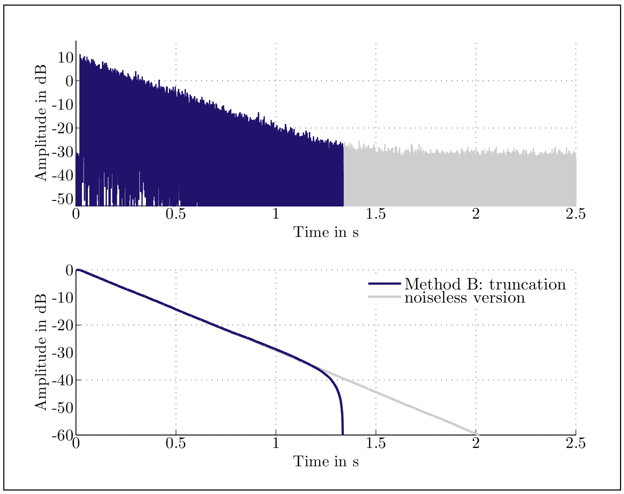

Figure 2. Example of an impulse response and corresponding EDC with truncation at the intersection time (Method B). The influence of noise is reduced significantly. The energy decay curve is underestimated.

noise floor. The noise error in EDC is reduced drastically (Figure 2), but the truncation introduces a further error. The energy decay curve is approaching minus infinity because of the missing signal energy from the truncation time to infinity. The unlimited dynamic range of the resulting EDC always allows an evaluation of the EDC, even if the signal to noise ratio is insufficient for a certain reverberation time.

2.3. Method C: Truncation and Correction

Lundeby *et al.* proposed a correction term to prevent the truncation error [9]. The missing signal energy from truncation time to infinity (triangle in Figure 3) C_{comp} is estimated and added to the truncated integral,

$$E(t) = \int_t^{t_{\mathrm{i}}} p^2(\tau)\,\mathrm{d}\tau + C_{\mathrm{comp}}. \tag{3}$$

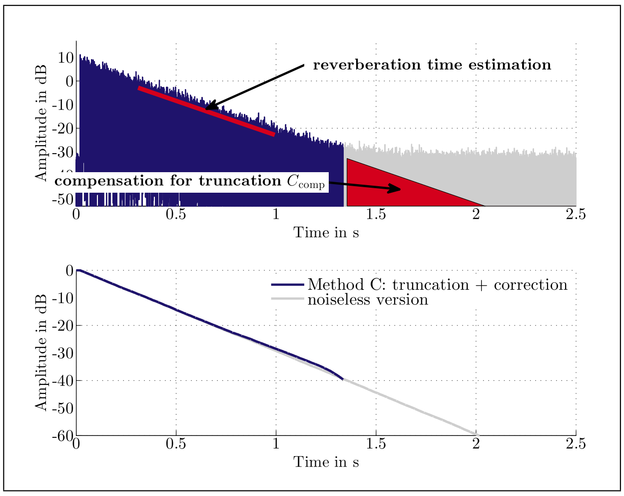

Figure 3. Example of an impulse response and corresponding EDC for truncation and correction term for the truncation (Method C). The influence of noise is reduced, just a small overestimation can be observed. The truncation error is eliminated and the EDC is limited to a reasonable range.

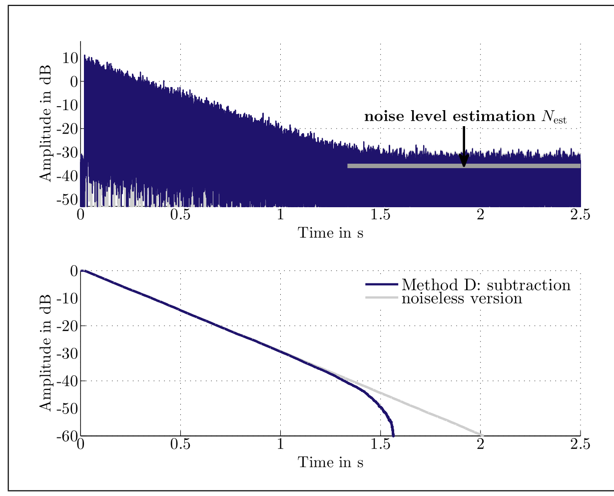

Figure 4. Example of an impulse response and corresponding EDC with subtraction of estimated noise level (Method D). The influence of noise in EDC is minimized, but the dynamic range of EDC is unlimited.

ISO 3382 recommends to calculate C_{comp} by assuming an exponential decay. The decay rate should be the same as given by the squared impulse response between t_1 and the intersection time. t_1 is the time corresponding to a level 10 dB above the level at the intersection time. This part of the impulse response, however, is already influenced by noise. At the intersection time signal and noise energy are equal by definition. Hence, Lundeby *et al.* suggested to leave a safety margin of 5–10 dB above the level corresponding to the intersection time to reduce the influence of the noise. In this study a safety margin of 10 dB is used, if not further specified.

The resulting EDC shows no truncation error any more (see Figure 3). The dynamic range of the EDC is limited according to the signal to noise ratio. The error caused by the noise is reduced significantly, even though a slight overestimation caused by the noise before the truncation time can be observed.

2.4. Method D: Subtraction of Noise

Chu proposed the "subtraction of noise"-method [10]. The noise level N_{est} is estimated and subtracted from the impulse response before backward integration,

$$E(t) = \int_t^{t_{\text{IR}}} \left(p^2(\tau) - N_{\text{est}}^2\right) \, \mathrm{d}\tau. \tag{4}$$

In the original measured impulse response the noise term is squared and therefore always positive. Due to the integration the error sums up and results in an overestimation of EDC. By subtracting the estimated noise level, the distribution of the new noise component can be considered as zero mean. The noise error is reduced by temporal averaging during the integration. This technique works well for the first part of the impulse response where the signal energy is dominant. For the later part, after the intersection time, this technique fails because the EDC is approaching minus infinity and is not monotonically decreasing any more (see Figure 4). Similar to Method B, the problem of an unlimited dynamic range of EDC occurs as well.

2.5. Method E: Truncation, Correction and Subtraction

The fifth method is a combination of all techniques mentioned above: The estimated noise level is subtracted before backward integration, the impulse response is truncated at the intersection time and the correction for the truncation is applied,

$$E(t) = \int_t^{t_{\mathrm{i}}} \left(p^2(\tau) - N_{\text{est}}^2\right) \mathrm{d}\tau + C_{\text{comp}}. \tag{5}$$

The influence of noise is minimized by subtracting the noise level. The truncation of the impulse response suppresses errors in the later part of the room impulse response resulting from noise subtraction and the correction for the truncation ensure a reasonable dynamic range (Figure 5).

3. Evaluation Approaches

The analysis of the different noise compensation techniques is carried out using two evaluation approaches that are described in the following two sections.

3.1. Model Approach

The model approach is a parametric description of an ideal room impulse response with added noise. Only the envelope of the impulse response is modeled without any temporal fine structure. The signal decay is assumed to be exponential and described by the reverberation time RT. The noise is stationary and its level is defined using the peak-signal-to-noise-ratio (PNR). The maximum of the signal

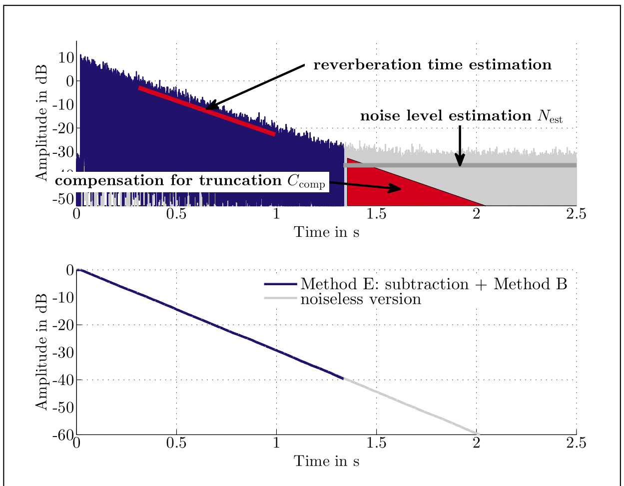

Figure 5. Example of an impulse response and corresponding EDC with truncation at intersection time, correction for truncation and subtraction of noise level (Method E). The influence of noise is minimized and the dynamic range of EDC is limited.

part is compared with the mean level of the noise. The third parameter is the length of the measured impulse response (t_{IR}), since it has an impact on the results obtained using one of the methods. The model room impulse response can be described as

$$p(t) = 10^{-6t/RT} + 10^{-\mathrm{PNR}/10}. \tag{6}$$

The linear regression is performed numerically using a sufficiently high sampling rate to exclude influences that occur due to sampling. The difference between the envelope and the mean value of the background noise is assumed to be approximately 11 dB, supposing a Gaussian distribution.

3.2. Measurement Approach

Most measurement approaches described in the relevant literature only investigate a few examples of different noise levels. This study tries to present a systematic approach, as in the study published by Hak [11], rather than some exemplary results that only show that there is a difference. In contrast to Hak's study where additive white noise is used, the different signal to noise ratios are obtained by performing real measurements and using real background noise. These long-term measurements were conducted in the general assembly hall of RWTH Aachen University (rectangular shape, 600 seats, volume 5500 m^3). A two-way omnidirectional custom-made dodecahedron loudspeaker was used. The receiver side consisted of one $\frac{1}{2}$-inch condenser microphone (B&K Type 4190) and 15 Sennheiser KE4 microphones. The complete input and output measurement chain was calibrated to make it possible to measure absolute levels. For the excitation, an exponential sine sweep with a frequency range from 20 Hz to 14 kHz and a length of 6 seconds was used. In the post processing the impulse responses were truncated at 4 seconds to guarantee a robust noise detection that is not influenced by the nonlinear components at the end. The measurements were performed using the Measurement Application of the ITA-Toolbox [15].

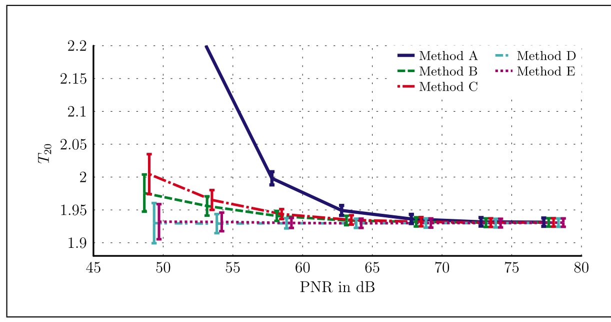

Figure 6. Evaluated reverberation time T_{20} dependent on the peak-signal to noise ratio for all five analyzed methods. The error bars indicate the standard deviation.

Between the single measurements the amplification of the excitation signal was changed to obtain different peak-signal-to-noise-ratios. A set of seven different amplifications was repeated 72 times to allow a statistical statement of the resulting parameters. The PNR was calculated separately for each measured impulse response to account for changing noise levels during the measurement session. The measurements were clustered into groups according to their PNR values. For each group the mean values and the standard deviation of the evaluated room acoustic parameter were calculated. The maximum difference in PNR in one group was 1 dB. Using this small range a high resolution of the PNR dependency was obtained and smearing effects were avoided. 504 broadband impulse responses were obtained using this measurement procedure. They have the same signal content, while the noise part changes for every measurement under realistic conditions (temporal distribution and frequency spectrum). The 16 microphones were spread over the listener area and used simultaneously.

The most important prerequisite for the success of this approach is the acoustical stability of the room. If the room changes while the measurement is carried out, it is not possible to distinguish these changes from the effects of noise. That is why there were no persons present during the measurements. After the last person had left the room the measurements taken in the first 90 minutes were discarded to make sure that the room conditions had settled. Temperature and humidity were monitored during the measurement and can be considered constant within a maximum difference of 0.6°C and 6% relative humidity.

4. Comparison of Noise Compensation Methods

The mean value and the standard deviation of the evaluated reverberation time T_{20} for the measurement approach are plotted in Figure 6 as a function of the peak-signal-to-noise ratio. Most compensation methods show a clear dependency of the evaluated reverberation time on the PNR.

For very high PNRs, however, T_{20} converges towards a constant value. This implies that the noise does not affect the evaluation for sufficiently high PNRs. These results are in accordance with the conclusions of Hak *et al.* [11]. Furthermore, all noise compensation methods show the same results for high PNR values. This also demonstrates that the compensation method has no influence on the evaluation for sufficient high PNRs. The fact that the results are not affected by the output level for high PNRs also shows that nonlinearities of the loudspeaker (that increase with output level) have no influence on the parameter. The resulting parameter for the highest PNR are considered as the true value and are used as a reference (separately for each noise compensation method) to show the relative error of the room acoustic parameters in the following analysis. Other room acoustic parameters show the same dependencies and are therefore also referred to their best estimate.

4.1. Reverberation time

First of all the reverberation time T_{40} is investigated. Although T_{40} is not very often used in practice, it is very suitable to compare the results of the two evaluation approaches. This parameter covers a wide range of PNR values from insufficient to very good signal to noise ratios and thus allows observing all occurring effects. ISO 3382 recommends a decay range of 55 dB for the parameter T_{40} [1] wich is equal to a PNR of 55 dB in diffuse sound fields.

Figure 7 shows the relative error of T_{40} for all five methods. Using no noise compensation (Method A) results in a huge overestimation of the reverberation time. This is in accordance with the effect seen in the EDC (see section 2). The truncation of the impulse response at the intersection time (Method B) reduces the noise effect considerably for high and medium PNRs. For low PNRs the reverberation time is strongly underestimated. This effect is caused by the underestimation in the energy decay curve. Applying the correction for the truncation (Method C) prevents this effect. Method C automatically yields no results for an insufficient dynamic range of the EDC. The slightly bigger errors for medium PNRs compared to Method B are caused by the absence of one of the two opposing effects (overestimation caused by noise and now missing underestimation caused by truncation). The subtraction of noise technique (Method D) gives perfect results for the model approach. However, this is caused by the simplicity of the model and the results cannot be interpreted. The measurement approach shows that there is nearly no systematic error for medium and low PNRs. For low PNRs the unlimited range of EDC results in an underestimation, similar to Method C. Method E (truncation, correction and subtraction of noise) yields the best results for the measurement approach. There is nearly no systematic error for mid and high PNRs and for insufficient PNRs the algorithm automatically provides no results. The results of the model approach can again only partly be interpreted due to the simplicity of the model.

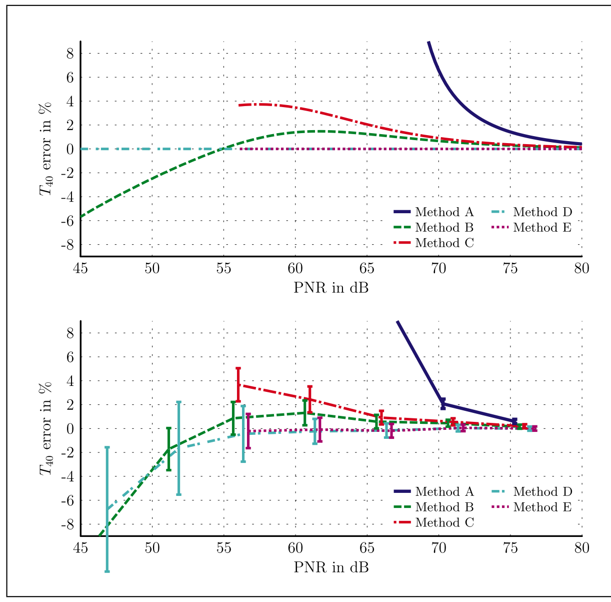

Figure 7. Relative error of the reverberation time T_{40} for model (top) and measurement approach (bottom). The measurement approach shows the 250 Hz octave band.

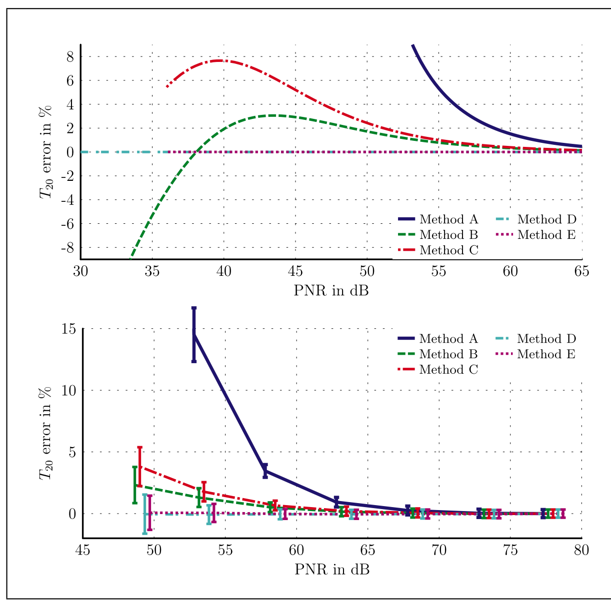

Figure 8. Relative error of the reverberation time T_{20} for model (top) and measurement approach (bottom). The measurement results show only high PNRs due to the chosen output amplitudes during the measurements. (For the sake of visibility of effects the ranges of x-axis is not equal.)

The results that are obtained using the model and the measurement approach are consistent with each other and thus the measurements confirm the validity of the model. An exception is Method D where the simple constant noise approach of the model results in perfect output parameters.

Afterwards the reverberation time T_{20} is examined since it is of greater relevance in practice. In Figure 8 it can be seen that the measurement approach only provides data for the high to medium PNR range. The model results have to be investigated to analyze the lower and therefore more interesting PNR range. Compared with T_{40} the results are shifted on the PNR axis which is a result of the different dynamic range of T_{20}. The smaller evaluation range and the resulting bigger percentage of the noise influenced part is the reason for the greater relative errors.

A relative error of 5% is used to determine the required minimal PNRs for each method, since 5% is also the commonly accepted just noticeable difference for the reverberation time. The systematic error for Method A (no noise compensation) exceeds the 5% limit at 55 dB. This is 20 dB higher than the PNR of 35 dB recommended in ISO 3382. Since the error is dependent on the total length of the impulse response, these limits are only valid for this example where the impulse response length $t_{IR} = 4$ s and the reverberation time RT = 2 s. For the truncation technique (Method B) the limit is in accordance with the ISO recommendation. When using truncation and correction (Method C) the 5% limit is at about 45 dB. However, the systematic error for this method will never exceed 8%, whereas for the previous methods the error can increase significantly. No systematic error is predicted for Method E (truncation, correction and subtraction of noise) by the model approach. The measurement approach confirms these results for PNRs where measurements are available. The systematic error will be clearly below 5% and the results will be discarded for insufficient PNRs, similar to the T_{40} results.

The mean errors in the reverberation time are systematic while the errors described by Hak in his study are random and have a zero mean. These differences may be caused by the artificial character of the additive noise or by the differences in the implementation of the calculation algorithms. Another reason might be that in the investigation of Hak many impulse responses from different room and reverberation times were used.

4.2. Clarity Index

The noise caused deviations appear at the end of EDC and move to the early parts for increasing noise levels. Due to the fact that the clarity index C_{80} is only based on one value of the very early EDC it is assumed that C_{80} is less sensitive to noise.

Figure 9 shows the evaluation of the results. Method A (no noise treatment) again shows the greatest sensitivity to noise. C_{80} is underestimated and the results fall below the limit of just noticeable difference of 1 dB at a peak signal to noise ratio of approximately 32 dB. The minimal required PNRs decrease by app. 10 dB to 22 dB for Method B (truncation) and to 20 dB for Method C (truncation and correction). The advantage of the automatic limitation of the results for insufficient peak signal to noise ratios for Methods C and E do not work for C_{80}. This means that for the clarity index (and also for the definition) the PNR has always to be evaluated to estimate the noise error. For inadequate PNRs the C_{80} results have to be discarded. Methods B, C, D and E perform similar and do not differ significantly. Again the two evaluation approaches are very similar.

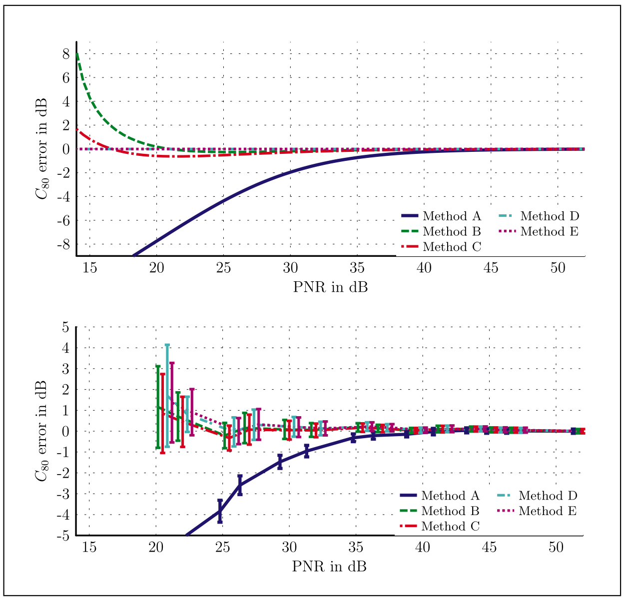

Figure 9. Error of clarity index C_{80} for model (top) and measurement approach (bottom). Using no noise compensation (Method 1) causes the largest systematic errors. All noise compensation methods (Methods B - E) reduce the noise influence and do not differ significantly.

5. Further Influences

In the following section further details of the measurement or the implementation are considered. The influences are determined based on the developed model and the measurements.

5.1. Implementation of Correction for Truncation

In section 2 two different implementations for the calculation of the truncation correction (used for Method C and E) are discussed. The late reverberation time used to determine the correction term can be calculated without (ISO correction [1]) or with a safety margin of 10 dB above noise level (Lundeby correction [9]). For the estimation of the other parameters required for the correction term (noise level and intersection time) the safety margin is always taken into account, as proposed by Lundeby. Without the safety margin the iteration algorithm does not work properly. Both evaluation approaches are used to highlight the difference in performance for both corrections.

The model approach shows that the systematic error is larger when the ISO correction is used (Figure 10): maximum error ≈ 7% for ISO correction and ≈ 5% for the Lundeby correction. The same differences between the two corrections can be observed when the measurement approach is used. The standard deviation for the ISO correction is slightly bigger than for the Lundeby correction, which is caused by the bigger influence of the noise as no safety margin is applied. The deviation between model and measurement approach is small and does not exceed the standard deviation of the measurements.

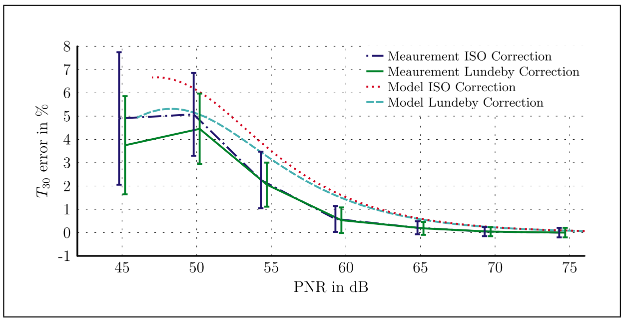

Figure 10. Relative error of reverberation time T_{30} for two different implementations of the truncation correction term.

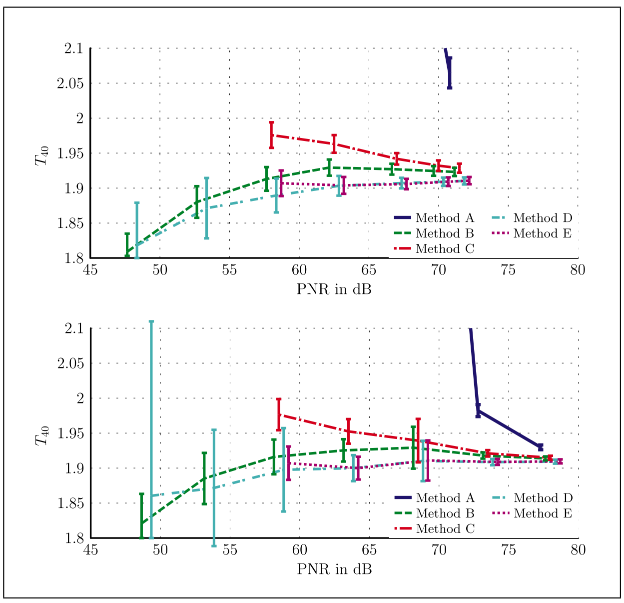

Figure 11. Comparison of evaluated reverberation time T_{40} for MLS (top) and sweep (bottom) excitation. For both results the microphone position is equal and the 1 kHz octave band is shown.

5.2. Excitation Signal

In this section the effect of the utilized excitation signal on the noise sensibility is analyzed. Therefore the two most commonly used excitation signals are discussed: sine sweeps and maximum length sequences (MLS). Both signals show different behavior for impulsive noise during the measurement and nonlinear components in the measurement chain.

The Hadamard transformation is used for MLS measurements to obtain the impulse response from the recorded MLS signal [16]. The impulsive noise that occurs during the measurement is transformed into a stationary noise and thus distributed over the entire impulse response. Due to this time spreading the noise amplitude is considerably reduced. The random temporal structure and the low amplitudes allow considering impulsive noise as additional stationary noise term in the impulse response as well. All noise detection and compensation techniques are thus taken into account in a similar way. The effect on the evaluated room acoustic parameters is small compared with the amplitude of the impulsive noise. In contrast, sweep measurements are more sensitive to impulsive noise. The deconvolution transforms impulsive noise into an inverse sweep in the impulse response. Especially after the frequency band filtering this inverse sweep distorts the noise detection. The room acoustic parameter is heavily distorted in the frequency band where the inverse sweep and the signal part of the impulse response overlap. A separate step to check whether impulsive noise occurred is required.

The excitation signals also display a different behavior for nonlinear components. The loudspeaker is usually the part of the measurement chain that introduces major nonlinear components. For exponential sweeps the major part of the harmonics of a nonlinear system appear at definite positions at the end of the impulse response (in case of cyclic deconvolution). This allows an easy estimation of the nonlinearity of the measured system and makes it possible to reduce these components by truncation of the impulse response. However, Ćirić *et al.* have shown that the nonlinearities also introduce an error to the fundamental part of the impulse response [17]. For MLS measurements the distortions cause spikes spread over the entire impulse response. It is not possible to remove the nonlinear components. The spikes first appear in the noise part of the impulse response, because of the lower amplitudes. In case of increasing nonlinearities the signal part is also affected.

Figure 11 shows the absolute errors for the reverberation time for MLS and sweep measurements. The frequency band and the position of the microphone are equal for both excitation signals. The differences in PNR values between both signals are caused by differences in the frequency spectra of the signals. Similar to the sweep measurements the amplitude of the MLS was increased in steps of 5 dB. For lower PNRs this leads to an increase of PNR by the same amount. In case of higher PNRs, however, the increase is less than 5 dB. The noise detection algorithm identifies the nonlinear spikes as noise floor, leading to an increasing noise floor for higher output amplitudes. The evaluated reverberation time does not change for higher output levels. This indicates that the nonlinear components in the impulse response are still small compared with the linear signal part.

The systematic errors depending on the PNR are similar for sweeps and MLS. The random deviations between both excitation signals hardly ever exceed 1–2%. One reason for the similarity of the results is the control of the measurement conditions. The lack of impulsive noise in the measurement and only small nonlinear components (due to the high-quality loudspeaker) ensure that situations where the excitations display a different behavior do not occur.

6. Conclusions

The performances of the five noise compensation methods differ significantly. It is shown that most methods lead to systematic errors. However, these errors can be predicted. It must, however, be mentioned that the performance of

Table I. Overview of the investigated noise compensation methods and their properties.

	Noise sensitivity	ISO compliance	Limitation of results
Method A	high	yes	no
Method B	medium	yes	no
Method C	medium	yes	yes
Method D	low	no	no
Method E	low	no	yes

the three methods allowed by ISO 3382 also deviate significantly. Using no noise compensation (Method A) results in a large overestimation of the reverberation time depending on the total length of the impulse response. Using a noise compensation method reduces the error significantly. The ISO compliant method to truncate the impulse response at the intersection time and correct for the truncation (Method C) has the advantage that results for reverberation times are discarded automatically, if the peak signal to noise ratio is insufficient. The systematic error for T_{20} is always $< 8\%$. Method E (subtraction of noise, truncation and correction) showed the smallest sensitivity to noise. The systematic error is negligible and for insufficient PNRs the results are discarded automatically. However, due to the subtraction technique this method is not compliant with the ISO standard. Table I provides an overview of all methods.

In general the clarity index C_{80} is less sensitive to noise. Again Method A (no noise treatment) shows the greatest noise sensitivity. The different noise compensation techniques (Methods B to E) are quite similar. Definition D_{50} and sound strength G are even less affected by noise.

Theoretically the knowledge of the systematic errors allows a manipulation of the measured reverberation time by choosing a suitable noise compensation method and adjusting the measurement parameters (impulse response length or output amplification). To avoid unclear or ambiguous manipulations and allow an estimation of the noise induced error the following specifications should be included or modified in the ISO 3382 standard:

- The application of a noise compensation technique should be mandatory or recommended. In the latest version of the standard noise compensation is optional.
- The subtraction of the noise technique proposed by Chu should be included as a possible noise compensation technique.
- The peak signal to noise ratio and the applied noise compensation method should be included in the measurement report, so that an estimation of the systematic error is possible afterwards.
- The proposal on how to calculate the correction of truncation should include the safety margin above noise floor as proposed by Lundeby *et al.* to reduce the error.

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