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ARCHIVES OF ACOUSTICS
Vol. **37**, No. 2, pp. 219–225 (2012)

DOI: 10.2478/v10168-012-0029-0

Numerical Compensation of Air Absorption of Sound in Scale Model Measurements

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(*received December 2, 2011; accepted April 27, 2012*)

The absorption of sound in air represents one of the main problems of the scale model measurements. This absorption, especially at higher frequencies, is considerably greater than the value determined by the law of acoustical similarity between the full scale and the scale model. Different alternatives are applied for compensation of the excess air absorption including a numerical compensation. In this paper, a modified approach to numerical compensation is proposed. It is based on compensation of the sound decay only, and not background noise. As a consequence, there is no an increase of background noise in the compensated impulse response. The results obtained by the proposed procedure are compared to the corresponding ones obtained by the other procedures.

Keywords: acoustic measurement, scale model, air absorption, numerical compensation.

1. Introduction

Significant improvement in computer simulations of an enclosure acoustics (Bork, 2005; Alpkocak, Sis, 2010) has not superseded the scale models. They can be considered to be a complement to other tools applied for an enclosure analysis and design from an acoustical point of view (see, e.g. Carvalho, Silva, 2010; Kosała, 2011). Thus, the scale models still have a very special position in practice (Gade, 2007). Some applications are described in the literature (Xiang, Blauert, 1993; Boone, Braat–Eggen, 1994; Picaut, Simon, 2001; Ismail, Oldham, 2005; Picaut *et al.*, 2005).

Usage of the scale model yields some specific benefits, but also introduces certain problems. One of them is related to the measurements in the models, or more specifically, to the absorption of sound in air (Boone, Braat–Eggen, 1994; Picaut, Simon, 2001; Ismail, Oldham, 2005; Picaut *et al.*, 2005). This problem is especially important at high frequencies that are actually used in the models due to the fact that the absorption increases with frequency. There have been several alternative solutions to the air absorption. The alternative that has recently attracted an increased attention is a numerical compensation (Boone, Braat–Eggen, 1994; Picaut, Simon, 2001; Ismail, Oldham, 2005; Picaut *et al.*, 2005; Polack *et al.*, 1989; 1993; Akil *et al.*, 1994). Apart from the important advantages, this compensation shows certain drawbacks and limitations. Some of them are caused by the increase of background noise (Polack *et al.*, 1989; 1993; Akil *et al.*, 1994). In order to eliminate this artifact of the compensation, a modified approach to the numerical compensation is proposed and analyzed in this paper. Its efficiency is tested, and the results are compared to those obtained by the other procedures.

2. Air absorption in scale model

Application of a scale model implies that certain rules defined by the laws of acoustical similarity have to be fulfilled (Xiang, Blauert, 1993; Picaut, Simon, 2001; Polack *et al.*, 1989). In this way, it is possible to have the sound field in a scale model identical or similar to the field in the full scale enclosure (space). The rules are related to the lengths, time, and frequency, but also to the absorption of sound in the fluid (air) filling the model and full scale. Thus, the air absorption in the model, α_m, should be S times greater (S is the scale factor) than the absorption in the full scale, α, at the corresponding frequencies, f, (Xiang, Blauert, 1993; Polack *et al.*, 1989)

$$\alpha_m(Sf) = S\alpha(f). \tag{1}$$

During sound propagation through the air, it is absorbed (attenuated) as a result of two different mechanisms: classical and relaxation effects (Bass, Bauer, 1972; American National Standards Institute [ANSI], 1995). The most important factors of the air absorption become more significant at higher frequencies and for longer propagation distances. This is why the absorption is increased with an increase in frequency and propagation time.

The absorption of sound in air can be quantified by the attenuation coefficient (giving the pure-tone sound attenuation in dB/m caused by the air absorption) expressed as (ANSI, 1995)

$$\alpha = 8.686 f^2 \left(\left[1.84 \cdot 10^{-11} \left(\frac{p_a}{p_r} \right)^{-1} \left(\frac{T}{T_r} \right)^{1/2} \right] + \left(\frac{T}{T_r} \right)^{-5/2} \cdot \left\{ A \cdot e^{-2239.1/T} \frac{f_{rO}}{f_{rO}^2 + f^2} + B \cdot e^{-3352/T} \frac{f_{rN}}{f_{rN}^2 + f^2} \right\} \right), \tag{2}$$

where A and B are the constants equal to 0.01275 and 0.1068, respectively, p_r and T_r are the reference air pressure and temperature, p_a and T are the air pressure and temperature in the model, while f_{rO} and f_{rN} are the vibrational relaxation frequencies for oxygen and nitrogen, respectively.

The air absorption (sound attenuation) in a model is usually greater than required by the similarity law (Xiang, Blauert, 1993; Picaut, Simon, 2001; Polack *et al.*, 1989; 1993), Eq. (1). This excess of air absorption can be scaled or compensated for by applying different procedures. The usual procedures are related to filling the model by nitrogen (Gade, 2007; Xiang, Blauert, 1993) or drying the air to about 2% of relative humidity (Gade, 2007; Polack *et al.*, 1989). While some authors prefer these procedures, the others point out the procedures' limitations (they require special equipment and can be time-consuming; in addition, the accuracy of the air absorption compensation is not sufficient) (Picaut, Simon, 2001; Polack *et al.*, 1989).

3. Numerical compensation

As an attempt to overcome the limitations of the previously mentioned procedures, an alternative procedure called computer compensation was proposed more than 20 years ago (Polack *et al.*, 1989). This computer (numerical) compensation is based on the signal processing, where the energy response is compensated by the fast Fourier transform (FFT) (Polack *et al.*, 1989). Besides, a similar procedure based upon digital filtering of the time history of a signal has been developed (Akil *et al.*, 1994). Since the introduction of the numerical compensation into practice, it has been used in a number of different applications (Boone, Braat–Eggen, 1994; Picaut, Simon, 2001; Ismail, Oldham, 2005; Picaut *et al.*, 2005; Polack *et al.*, 1989; 1993).

Although these numerical compensation procedures have offered some significant advantages, they have introduced some additional problems and placed some other restrictions. One of the most important drawbacks is related to an artificial increase of background noise with time (Picaut, Simon, 2001; Polack *et al.*, 1989; 1993), as shown in Fig. 1. This represents an artifact of the numerical compensation. The noise increase is caused by the compensation of the whole impulse response (IR) of a scale model. However, a room IR generally consists of the sound decay and background noise. Only the first part of the response is modified (attenuated) by the air absorption in the described way, and only this part should be compensated for.

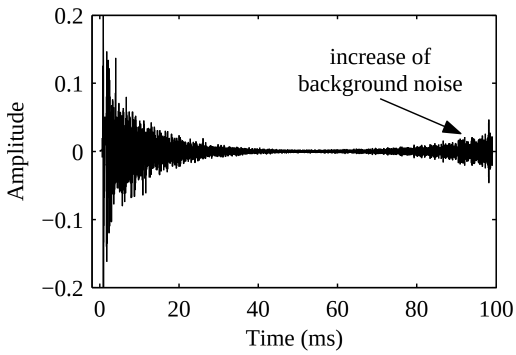

Fig. 1. Illustration of an increase of background noise caused by numerical compensation.

An approach to numerical compensation different from the ones presented in the literature (Polack *et al.*, 1989; Akil *et al.*, 1994) is proposed here. It is based on the compensation of the sound decay part of a scale model IR only, and not background noise. Since in room acoustics an IR is analyzed in octave or third-octave bands, the proposed compensation procedure can be applied to the IR filtered in such bands. Alternatively, it can be applied by using short-time Fourier transform (STFT).

The proposed procedure for numerical compensation of the excess air absorption is presented in Fig. 2. A measured scale model IR is first filtered in octave or third-octave bands. Then, in order to separate the sound decay from background noise, each of these filtered responses is truncated at the intersection of

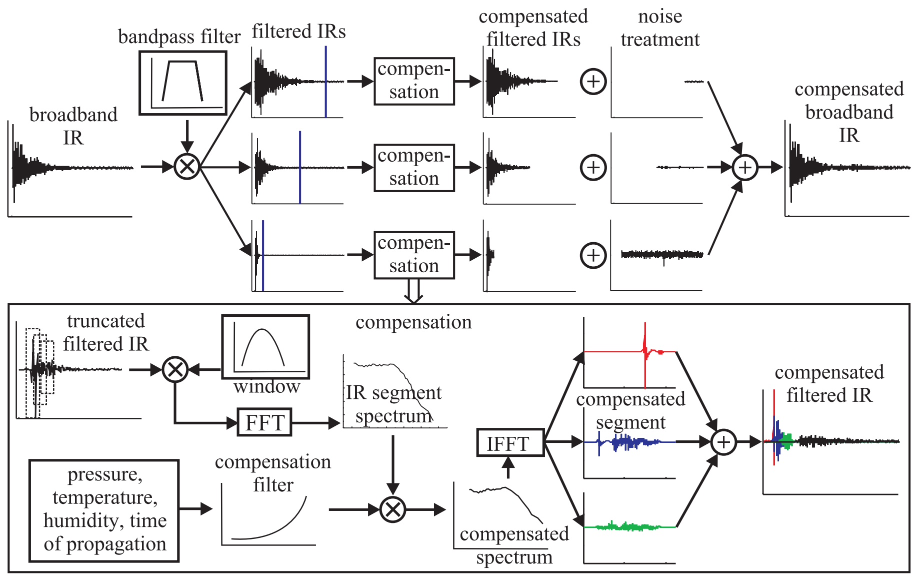

Fig. 2. Developed procedure for numerical compensation of excess air absorption of sound.

the sound decay and background noise (at the knee) (Ćiric, Milošević, 2005). The truncated IR is segmented by applying Hanning windows in the time domain with an overlap of 50%. Each segment is transformed to the frequency domain by the FFT.

Based on the position of the window (segment) in time and Eq. (2), the frequency characteristic of the compensation filter is calculated. The window position determines the mean time of sound propagation, that is, the mean propagation distance for that window. The frequency characteristic of the compensation filter is generated using the sound attenuation caused by the air absorption, defined by Eq. (2), for particular mean propagation distance (associated with that window), or more precisely the excess sound attenuation in reference to the attenuation specified by the similarity law. The compensation filter is constructed to compensate this excess sound attenuation. Multiplication of the segment spectrum by the corresponding compensation filter (its frequency characteristic) leads to the compensated (corrected) spectrum of the segment. The compensated segment of the IR in the time domain is obtained by the inverse FFT. Afterwards, all compensated segments are summed together to give the compensated sound decay part of the IR in a particular frequency band. The proposed compensation procedure is outlined by a pseudo-code algorithm given in Fig. 3.

The background noise part is treated separately, and appended to the sound decay part. Three alternative solutions are proposed for background noise. Since the numerical compensation increases the sound decay level, especially at higher frequencies, the first alternative for background noise treatment is related to adequate scaling (normalization) of the noise in each frequency band. This means that the background noise

(p - pressure, T - temperature, h - humidity, S - scale factor, s_j - propagation distance)

Input (broadband IR, p, T, h, S)

for i = 1:number of bands

  filtered IR = broadband IR * window-bandpass filter (i)

  truncated filtered IR = filtered IR from the IR beginning to the knee

  noise = filtered IR from the knee to the end of the IR

  % Compensation

  for j = 1:number of segments (until the end of the truncated filtered IR)

    IR segment = truncated filtered IR * window-time segment (j)

    IR segment spectrum = **FFT**(IR segment)

    Construct compensation filter based on p, T, h, S and s_j

    compensated spectrum = IR segment spectrum*compensation filter

    compensated segment (j) = **IFFT**(compensated spectrum)

  end for

  compensated filtered IR (i) = **sum**(compensated segment (j)); j = 1:number of segments

  % Treatment of noise part (noise) of the filtered IR

  if noise scaling

    Scale noise

  else if noise discarding

    Discard noise (noise = 0)

  else if decay extension

    Extend sound decay (compensated filtered IR (i)) until it reaches the noise level

  end if

  Append noise to the compensated filtered IR (i)

end for

compensated broadband IR = **sum**(compensated filtered IR (i)); i = 1:number of bands

Output (compensated broadband IR)

Fig. 3. Pseudo-code algorithm outlining the proposed procedure for numerical compensation (* denotes multiplication).

should be increased in such a way that its average level is equal to the level of the compensated sound decay at its end (alternatively to the average level of the very last part of the decay). The second alternative for background noise treatment is to discard this part of an IR. The third alternative is related to the decay extension where the compensated sound decay in every frequency band is extended so that it continues to decay until it reaches the corresponding (not

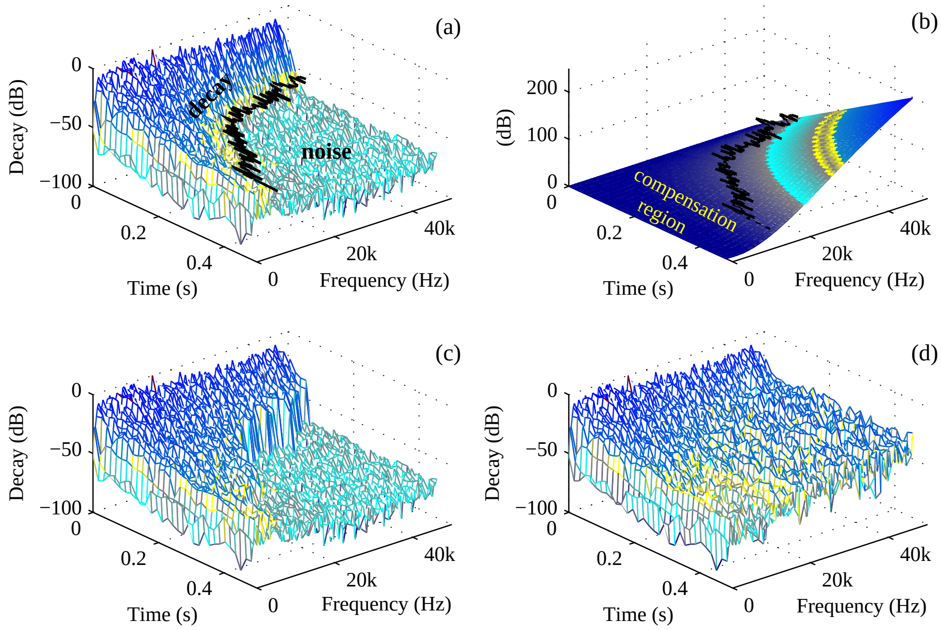

Fig. 4. Developed numerical compensation applied by using STFT: time-frequency diagram of IR with truncation curve separating the sound decay and background noise part (a), compensation filter with the curve separating the part applied for the compensation (compensation region) (b), time-frequency diagram of IR after compensation of the decay part only (c), and after background noise scaling (d).

scaled) background noise level. The mentioned extension can be done by modifying the first part of the noise to become a decaying function with the same decay rate as the sound decay. This first part of noise should be adequately scaled in order not to introduce a discontinuity when it is appended to the compensated decay. After processing of both sound decay and background noise part, the compensated IRs from different frequency bands are summed together to give the compensated broadband IR.

The proposed compensation procedure can also be applied in a somewhat different way. Thus, instead of bandpass filtering, a series of STFTs is successively calculated along the broadband IR of a scale model. In this way, a specific time-frequency diagram (STFT spectrum) is constructed (Fig. 4a). It consists of a series of spectra looking from the frequency domain, that is, a series of decay curves looking from the time domain. Each of these decay curves is processed by the compensation procedure in a similar way as presented in Fig. 2. So, a particular decay curve is first truncated at the knee. Then, the numerical compensation is applied to the sound decay part using the corresponding part of the compensation filter (for a particular frequency bin) shown in Fig. 4b. As a result, a compensated decay is obtained (Fig. 4c). Afterwards, the background noise part is processed (e.g. scaled), and appended to the compensated decay (Fig. 4d). This procedure can be represented in a simplified way by multiplication of the corresponding parts (decay and compensation region) of two time-frequency matrices (for the IR and compensation filter), shown in Fig. 4a and Fig. 4b. The compensated broadband IR can be recovered from the compensated time-frequency matrix by the inverse STFT.

4. Results

The developed numerical compensation procedure is applied in both proposed ways (bandpass filtering and STFT) to a number of IRs measured in different scale models. The results for two representative examples of the scale models (their IRs) are given in this paper. The first one whose IR is used in Fig. 4 is a one-tenth scale model of an irregularly shaped reverberation chamber with reflective walls, floor and ceiling made of glass. Its volume is 0.24 m^3, and it represents a room with long reverberation and almost uniform sound pressure level distribution. The second model whose IR is used in Figs. 5 and 6 is a one-tenth experimental scale model of a parallelepiped room of dimensions $0.8 \times 0.6 \times 0.47$ m^3. This model was developed for investigation of the influence of diffusivity, and its details can be found in (Šumarac–Pavlović, Petrović, 2010). In addition, the results obtained by the proposed procedure are compared to those obtained by other two procedures presented in the literature (Polack *et al.*, 1989; Akil *et al.*, 1994) as well as by commercial measurement software.

Typical results obtained by the proposed procedure using bandpass filtering and all three alternatives for

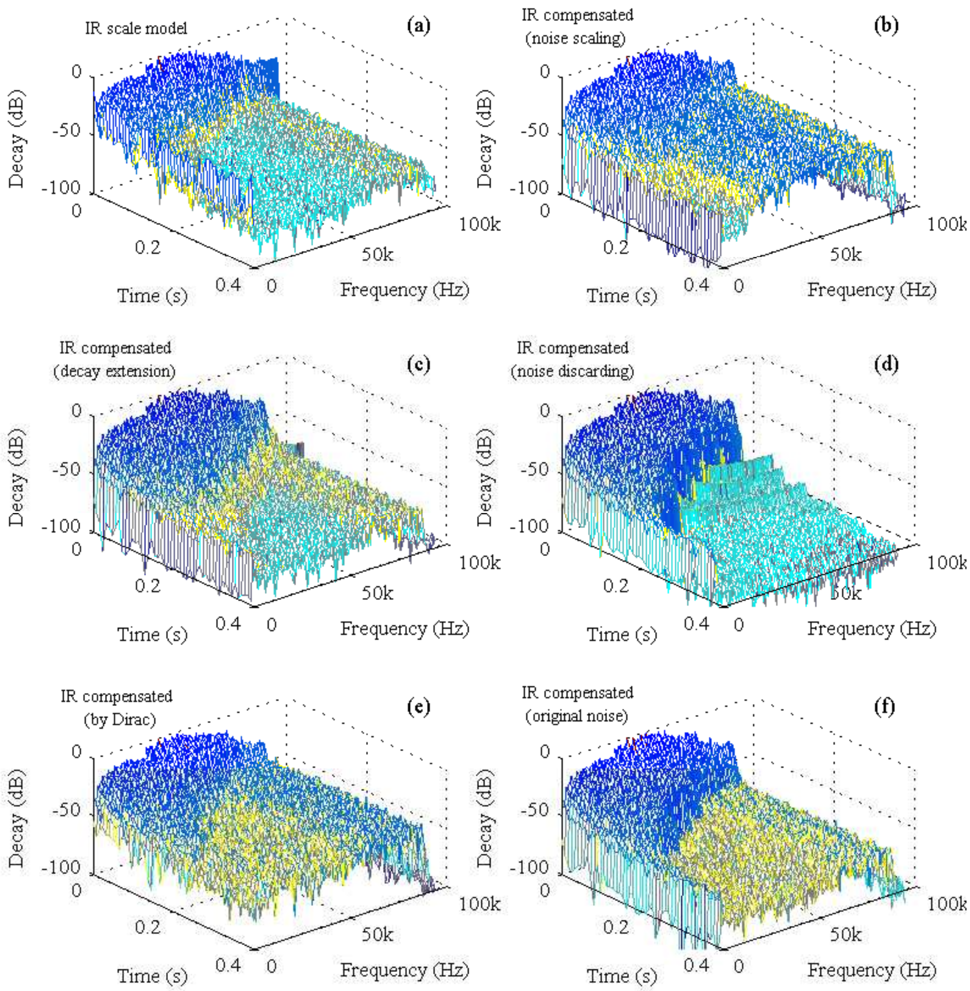

Fig. 5. Time-frequency diagrams of: IR measured in the scale model (a), IRs compensated by the proposed procedure with noise scaling (b), decay extension (c) and noise discarding (d), IR compensated by Dirac 3.0 software (e), and IR where only the sound decay is compensated by the developed procedure and original noise from the measured IR is appended (f).

noise processing are shown in Fig. 5. After applying the proposed compensation procedure to the IR filtered in third-octave bands, the compensated IRs from all frequency bands are summed together, as shown in Fig. 2. In addition, the excess air absorption is also compensated using the procedure incorporated into commercial software (Dirac 3.0) (see Fig. 5e).

When the developed compensation procedure is compared to the one from commercial software (Dirac 3.0), it can be observed that both procedures change the IR, especially at higher frequencies. However, there are certain differences between the compensated responses, but also between the decay curves based on Schroeder backward integration (Schroeder, 1979), as shown in Fig. 6. Thus, in the curves in third-octave bands obtained by the developed procedure, a difference between the decay slopes of the compensated and original response (curve) continually increases with frequency. On the other hand, the compensation carried out by the commercial soft-

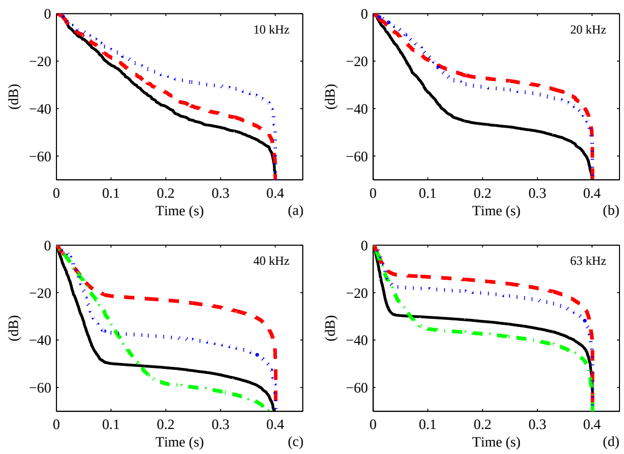

Fig. 6. Decay curves in the third-octave bands obtained by the backward integration of the original IR of the scale model with $S = 10$ (solid lines), IR compensated by the developed procedure with noise scaling (dashed lines) and decay extension (dash-dotted lines), and IR compensated by the commercial software – Dirac 3.0 (dotted lines).

ware is not consistent in different frequency bands. The decay curves of such a compensated response can have strange curvatures at the beginning, and almost the same slope as the curves of the original response in the major part of the decay, especially at higher frequencies.

5. Discussion

As a consequence of specific time-frequency dependency of the air absorption, that is, of the compensation filter (gain), during the compensation greater changes occur in the segments (parts) positioned away from the response beginning, and at higher frequencies. The gain of the compensation filter for the instances coincident with the end of an IR, and for the highest frequencies can be even several hundreds of dBs. This causes enormous increase of background noise if the numerical compensation is applied to the whole IR including background noise, as in the procedures from the literature (Polack *et al.*, 1989; Akil *et al.*, 1994). The decay of an IR of several tens of dB becomes almost negligible to this noise increase. Such a compensated IR is useless at higher frequencies, and additional processing is necessary in order to overcome the problem introduced by the mentioned noise increase (Polack *et al.*, 1989; Akil *et al.*, 1994).

One of the main drawbacks of the numerical compensation procedures from the literature – increase of background noise, does not exist in the developed procedure. However, even in this procedure, if a frequency range of the measured IR is great (e.g. 96 kHz), and if the IR is long enough, the maximum values of the compensation filter gain can be significant in comparison to the dynamic range of the IR. During the compensation, the IR dynamic range is decreased, since the sound decay (level) is modified by the time-frequency dependent increase. In that regard, the compensation filter gain directly represents a decrease in the IR dynamic range. This weakness (limitation) caused by high sound attenuation represents a general weakness of any numerical compensation procedure (Gade, 2007). Thus, for every particular scale model, there is an upper cutoff frequency up to which the numerical compensation is viable. This cutoff frequency depends on the propagation distances found in the model (IR) and dynamic range of the IR.

Adopting a certain dynamic range as a threshold (the smallest acceptable dynamic range of the IR obtained after numerical compensation), the cutoff frequency can be calculated by an iterative procedure. The highest frequency of the model IR is set to be the cutoff frequency. Then, the dynamic range that can be achieved after compensation is calculated based on the actual dynamic range and compensation filter gain (for particular length of the model IR up to the knee) at that frequency. If the obtained range is smaller than the adapted threshold, another frequency, lower than the previous one, is set to be the cutoff frequency. The procedure is repeated until the frequency is found yielding the range achieved after compensation equal to or greater than the threshold. This frequency is the cutoff frequency for the adopted threshold (dynamic range).

Applying an adequate technique for the IR measurements, such as swept sine technique, and taking care to obtain as large as possible the IR dynamic range, the mentioned cutoff frequency can be close to 100 kHz. This implies that the frequency range (of the full scale) is limited to about 10 kHz for a tenth-scale model. If a model scale factor is larger, the frequency range is further reduced. In case that such a frequency range is not sufficient, other procedures for air absorption compensation should be applied, such as filling the model by nitrogen (Gade, 2007; Xiang, Blauert, 1993).

Although the first two alternatives for background noise processing, noise scaling and discarding, enable certain important benefits (easy application, IR and decay curves of a common shape without discontinuity for noise scaling, and decay curves with larger dynamic range for noise discarding), they introduce some other weaknesses, too. So, discarding of the noise part leads to a discontinuity of an IR, that is, abrupt stop of the sound decay. The broadband IR composed of the narrow-band IRs of different lengths could have a shape in form of steps, especially close to the end. On the other hand, the noise scaling results in an artificial frequency dependent increase of background noise. The noise is more increased at higher frequencies. In such a way, the frequency characteristic of background noise becomes unnatural, since the noise levels become higher at higher frequencies. Both mentioned weaknesses can be overcome by the third alternative for noise processing (the decay extension). The extension applied here is similar to the one presented in the literature (Polack *et al.*, 1993). Nevertheless, there are some principal differences – an IR is truncated before the compensation here, only the sound decay part is compensated for, and the sound decays until it reaches the original noise level existing before the compensation. By applying the decay extension, instead of noise scaling (increase) or discarding, the noise level remains the same.

6. Conclusions

The modified procedure for numerical compensation of the excess air absorption is presented. It can be applied in two ways – by using bandpass filtering and STFT. This procedure differs from the ones presented in the literature, since only the sound decay part of the IR is compensated for and not background noise. In this way, one of the main drawbacks of the numerical compensation related to background noise increase is eliminated. Two noticed weaknesses of the proposed procedure (unnatural increase of background noise at higher frequencies, and possibly small dynamic range of the compensated IR at higher frequencies representing a general weakness of numerical compensation) can be mitigated by the decay extension. The broadband IR compensated by the proposed procedure can be used for both evaluation of acoustic qualities of auditoria and auralisation with certain limitations related to the frequency range (scale factor).

Acknowledgments

This research is supported by the Ministry of Science and Technological Development of Serbia through the project No. 36026. The authors would like to thank the anonymous reviewers for their constructive reviews and many helpful comments. They also thank Professors Miomir Mijić and Dragana Šumarac–Pavlović who provided impulse responses measured in the models.

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