Acoustic verification of rectangular reverberation chamber using impulse sound source

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Abstract: There are various methods for determining the acoustic quantities of materials, machines, equipment, and their sub-assemblies. A suitable measurement procedure for a given purpose depends on the frequency range of interest and the required precision. Nowadays, most laboratories deal with the effective measurement management system which ensures that the measurement processes are fit for their intended use, and furthermore, it leads to managing the risk of incorrect measurement results. However, this management system is mainly focused on the generic requirements and routine metrological confirmation of measuring equipment. This approach is essential especially in the large organization performing measurements, although it does not ensure the correct measured data. In the case of acoustic laboratories, in addition to the precision of measuring equipment, has equal importance the properties of environment where the measurement takes place. This paper describes an experimental process to identify the acoustic properties of reverberation chamber which proportions, shape and construction material of boundary surfaces are not in compliance with standards in every detail. In the following, the subsequent acoustic treatments to extend the measured frequency range, are discussed.

Key–Words: reverberation chamber, verification process, impulse sound source, reverberation time, decay curve, eigenfrequency

1 Introduction

One of the most fundamental factors affecting the inside enclosures is an excessive noise generated by equipment of environmental engineering [1]. High sound pressure levels can damage the hearing and cause dangerous neurovegetative reaction patterns [2]. From this perspective, it is necessary to approach the issues from the viewpoint of the entire chain of noise transmission into the buildings. It includes features of the sound source in terms of temporal, spectral and spatial aspects, of the sound propagation, and of individual sound transmission paths, and furthermore with noise receptiveness of individual humans.[2] However, these features are not easy to measure; moreover, with respect to all requirements of gripping and placing the sound source. To determine acoustic parameters of these machines with higher accuracy, such as sound power, must be performed the measurements in the acoustic laboratory.[3] Certain types of acoustic measurements can be carried out in an environment which is free of reflections or environment which uses reflection in many directions as reverberation rooms [4].

Reverberation rooms are useful for measurement of sound power, in particular, the sound power of machines that operate in long cycles, since the alternatives, the sound intensity method, and the free field method, are not very suitable for such sources [5]. However, the round robin evidence has shown that inter-laboratory reproducibility of acoustic data is difficult to achieve and high levels of measurement uncertainty remain even when design and construction process was carried out in accordance with recommended practices [6] [7]. These maladies occur most frequently at the low frequencies [8]. Ways of dealing with these facts are based primarily on their own test laboratories identification including measuring, data processing, and re-derive the acoustic properties.

This paper presents the key findings, proposed improvements, and results of the verification the newly build Laboratory of Environmental Engineering at Tomas Bata University in Zlin, Faculty of applied Informatics (FAI UTB). This laboratory consists of Universal Compensate Calorimetric chamber which also serves as Reverberation chamber. This chamber is used among other things to measure energy parameters and sound power level of equipment commonly used in buildings.

2 Reverberation chamber

The design of reverberation chamber is very complex concernment; moreover, when a multipurpose facility is concerned. The proposal of reverberation chamber depends on the prime purpose of usage. Acousticians use reverberation chamber to conduct various acoustical tests, including absorption and scattering coefficient measurements of acoustic materials, and determination of sound power levels of noise sources using sound pressure. European standards [9] and [10] are based on the already known facts in spatial acoustic and can be used as brief guidelines for the design of reverberation rooms. However, these standards assume diffuse field when the requirements, the volume, not rectangular shape and the sound absorption of the surfaces of the reverberant room, are satisfied. Following these assumptions could cause the sufficient diffuse sound field, but nevertheless, it is necessary to know the frequency range where the sound fields are adequately diffuse and where not. The measured data between the sound source and the diffuse sound field could be afflicted with great uncertainty in the predominantly non-diffuse section [11].

In enclosures, the many waves can travel at the same time even they can have a different shape. The eigenfrequencies have the major impact on the distribution of sound because when the frequency of excitation sound is similar as eigenfrequency, it evocates resonance and hence the acoustic properties of the space in terms of sound transmission from the source to the observer (microphone) are highly dependent on frequency [11]. In order to ensure a good sound transmission, it is necessary to establish on the observed frequency band, a maximum eigenfrequencies, as many as possible. This implies that the lowest eigenfrequency should be much smaller than considered frequency band. Unfortunately, this condition is met only for larger rooms (min. 200 m^3) [12].

Besides the sound transmission, the eigenfrequencies have an effect to temporarily occurring phenomena, ie. at the beginning the sound source excitation and its end because the sound energy growth due to reflections from the boundaries or expires due to absorption on the walls and in the air. It depends on what frequencies is energy transmitted and how these frequencies correspond to the eigenfrequencies of the room. To achieve monotonic dependence of rising and falling sound energy on the observed frequency, it is necessary to ensure that even the lowest frequencies must contain the highest number of eigenfrequencies [4]. This can only be achieved by suitable design the room dimensions.

The reverberation chamber at FAI UTB has a rectangular shape with dimensions 6,66 x 4,30 x 3,27 m and a volume of $93,66 m^3$. According to the wave-based acoustics approach, the proportions, and shape of the chamber could appear to be a tricky problem. Its proportions should be selected so that the ratio of any two dimensions does not equal or closely approximate an integer [9]. In this case, however, the chamber dimension ratio it is almost the integer. This can result in the improper decomposition of eigenfrequencies from the chamber response at low frequencies. This is mostly because of unevenly distribution of sound energy as described above.

Fig. 1: Reverberation chamber at FAI UTB in Zlin.

The surrounding surfaces of the room consist of different materials, therefore, the sound absorbing qualities of the walls, floor, and ceiling can vary considerably within a frequency range of a few hertz. Particularly, ceiling and three walls are consists of PUR panels, remaining wall is formed by Cetris Farmacel panels and the floor is covered by an antiskid metal plate. This multi-material concept is a rather awkward considering the fact very complicated evaluation of reverberant properties of the inner space at the design stadium. In the case of rectangular shape, the accuracy and reliability of results would be impaired by the interference of the direct sound with sound reflected from the boundaries [11].

3 Verification description

The aim of the verification process was to specify the reverberant properties of acoustic test chambers with respect to the prime purpose of usage. This is a desirable goal of room acoustical efforts since differences in reverberant properties can cause significant errors in measured data [11]. For example, the absorption coefficients determined in reverberation chambers are commonly exaggerated by diffraction effects and care must be taken, where it is critical, to correctly take such effects into account. The appropriate proof can be that absorption coefficient tables published in the literature often contain values greater than 1 which is meaningless.

It is possible to conduct the evaluation of the acoustical behaviour of the chamber by different ways. First one established through theoretical modelling, second one using numerical simulations and third one based on the real measurements [13]. Description of a sound field in enclosures and evaluation of their reverberant characteristics are not simple, and several theoretical methods of different complexity have been developed for this. Numerical simulations are very useful, however, the accuracy of the prediction models is highly dependent on the quality of the input data, including the absorption coefficient of surfaces which often can not be determined correctly [12]. These two ways have the advantage to be able to obtain preliminary results when it is still on the drawing-board.

The verification described in this paper is based on a calculation of basic parameters resulting from wave equation and several series of measurements which will be evaluated by impulse response of the chamber. The first series of measurements were performed with an empty chamber and with each other series was conducted a partial acoustic treatment. Afterward, from the results were calculated remainder parameters which characterizing the acoustic properties. This method was chosen because no absorption coefficient date was available for the individual surface materials for the chamber where the verification was carried out.

To assess correctly the acoustic properties of the chamber in terms of signal transmission, it is important to know the number of eigenfrequencies N at a certain frequency. A detailed description of how to calculate these values with all correction terms is in [11]. The corrected expression of the number of modes with eigenfrequencies up to a frequency f is

$$N = \frac{4\pi}{3} V \left(\frac{f}{c}\right)^3 + \frac{\pi}{4} S \left(\frac{f}{c}\right)^2 + \frac{L}{8}\frac{f}{c} \tag{1}$$

Where is V volume of the chamber $[m^3]$,
S area of all walls $[m^2]$,
L sum of all edge lengths $[m]$,
c sound velocity $[m/s]$.

From this equation yields the important verdict that the number of eigenfrequencies is independent on the size of the individual dimensions of the room but only on size of its volume.

The average density of eigenfrequencies on the frequency axis, i.e. the number of eigenfrequencies per Hz at the frequency f , is

$$n = \frac{dN}{df} = \frac{4\pi V}{c^3} f^2 + \frac{\pi S}{2c^2} f + \frac{L}{8c} \tag{2}$$

As can be seen from Figure 2 the calculated density of eigenfrequencies in the chamber is at the low frequencies (under 100Hz) a little bit sparse. This means that the signal transmission at those frequencies will be very unstable; moreover, the inappropriate ratio of chamber dimensions probably disturbs a diffuse field due to insufficient time density of wave reflections.

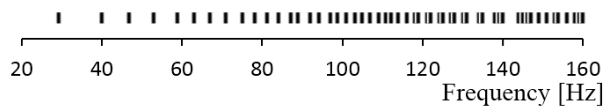

Fig. 2: Spectrum of eigenfrequencies at low frequencies for reverberation chamber with a volume of 93,66 m3.

Sufficiently uniform sound field is defined by the requirement that on average three eigenfrequencies fall into one resonance half-width. The limited frequency separating both cases was developed by Schroeder.

$$f_s \approx \frac{5500}{\sqrt{V\langle\delta\rangle}} \approx 2000\sqrt{\frac{T}{V}} \tag{3}$$

Where is T reverb. time $T = 6,91/\langle\delta\rangle$ $[s]$,
δ damping constant $[-]$.

Below this frequency the modal density is low and particular modes can be decomposed from the room response, thus in multimode resonance systems, the Schroeder frequency fs marks the transition from individual, well-separated resonances to many overlapping modes [14].

4 Methods

Reverberation time is the main parameter determining the indoor reverberation characteristics, thus, it is primarily used to assess the acoustic quality of enclosures [14]. For the purpose of analysing the structure of the impulse response characterizing the sound transmission through the chamber, this parameter was observed.

4.1 Measurement equipment

The standard equipment for this measurement is depicted schematically in Figure 3. The output voltage of the microphones and preamplifiers is fed to a multichannel Nor850 and then to a sound power module software which running the state-of-the-art Windows platform. This module enables connecting a number of individual measuring units through various communication channels including both standardized LAN and USB interfaces. Wireless communication through Bluetooth or WLAN is also available. It is very important to appropriately set the value of the sampling period. This setting controls the period length of each sample along the decay. In this case was chosen the lowest one, 5 ms.

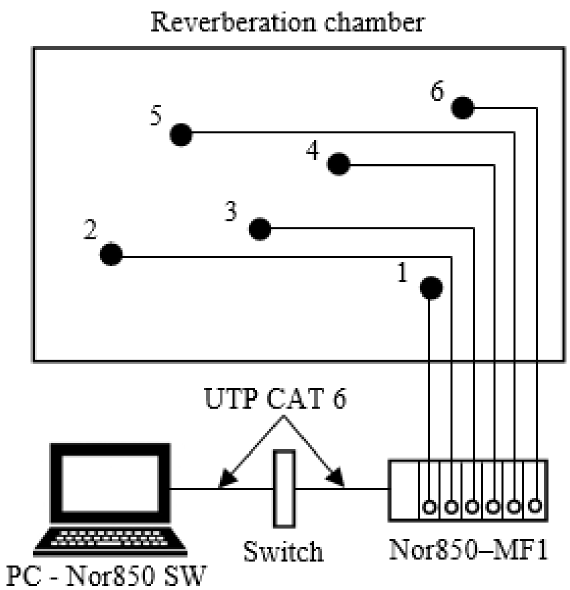

Fig. 3: Scheme of measurement equipment (points 1-6 are consist of microphone GRAS GS-40AF and preamplifier Nor1209).

4.2 Measurement process

The measurement was performed under conditions which are standardly used for determination of sound power levels of noise sources using sound pressure. The chamber was maintained at a temperature of 23C and relative humidity of 50% and the range of mid frequencies, for which all measurements, was taken, extends from about 50 to 10 000 Hz. The following procedure is not required to comply with the same order.

First of all, was measured the background noise level in the room with the all devices turned off by measuring the time-averaged sound pressure level. In the case of acoustic laboratories, it is not so frequent, however, when the room impulse response is contaminated with high-level background noise the resulting accuracy is substantially reduced because of a distortion of decay curve slope during the late decay [14]. Secondly, the determination of the impulse sound generator suitability was done. In this case, the chamber was excitation by a pistol shot. Each third-octave bandwidth was examined, more specifically, the standard deviation and the amplitude of excitation peak of a sound pressure measured in six observation points. The minimum amplitude size depends on the chosen sound pressure drop which will be evaluated, usually 35 dB [15]. Finally, the last part of the process is to obtain several different decay curves.

If the sound field were completely diffuse, the decay should be the same throughout the chamber [15] [16]. Since this ideal condition is hardly ever met in real enclosures, it is advisable to carry out several measurements for each frequency band at different microphone positions. This is even more important if the quantity to be evaluated is the early decay time, which may vary considerably from one place to the next within one inner space [11]. For this reason, the measurements were taken at least 3 locations of impulse source and with at least 6 microphone positions. The experimental arrangement consisted of

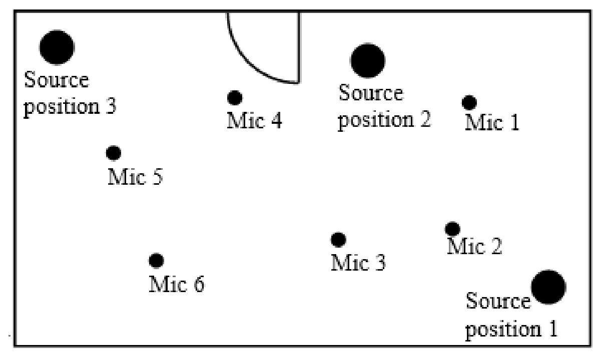

Fig. 4: Placement of microphones and sound source in each measurement series.

a 1/2 omnidirectional microphones and worker with pistol inside the chamber. The subsequent analysis was achieved conveniently using a multichannel as a real-time analyser (Nor850 MF-1). Usually, the sound pressure level in the measure decay curves does not fall in a strictly linear way but contains random fluctuations which are due to complicated interferences between decaying normal modes [11].

The reverberation time was calculated using a linear least-squares regression [15] of the measured decay curve, from a level 5 dB below the initial level to 25 dB below, using a linear regression of ensemble averaged decay curves. It means that the decay curve is approximated by a straight line $\hat{L}_i$ [dB], at the time $\hat{T}_i$ [s] is defined as

$$\hat{L}_i = a + bt_i \tag{4}$$

Where is a value $\hat{L}_i$ at the time t=0 $[dB]$,
b line tilt $[dB/s]$.

According to linear least-squares regression method is estimation of value a and a line tilt b determine by following equations:

$$a = \bar{L} - b\bar{t} \tag{5}$$

$$b = \frac{\sum_{i=1}^{n} (t_i L_i) - mt\bar{L}}{\sum_{i=1}^{n} \left(t_i^2\right) - mt^{-2}} \tag{6}$$

Where

$$\bar{L} = \frac{1}{n} \sum_{i=1}^{n} L_i \tag{7}$$

$$\bar{t} = \frac{1}{n} \sum_{i=1}^{n} t_i \tag{8}$$

Estimation of reverberation time $\hat{T}$ is given by

$$\hat{T} = \frac{-60}{b} \tag{9}$$

Another method of determining the decay curve consists in the reverse-time integration of squared impulse response which is described in [17].

Another desired parameter was to determine the absorption coefficient for inner space of the chamber. A couple of authors have suggested that alternative reverberation time formulations should be used to calculate the absorption coefficients from the reverberation time measurements. Using more exacting reverberation time equations may help predictions, but it does not prevent the inaccuracies to re-derive the absorption coefficients, due to edge effects and the non-diffuse nature of the reverberation chamber [12]. To derive practical conclusions from this fact, it could happen that results will be undervalued where the density of eigenfrequencies is sparse. To re-derive the sound absorption coefficient we favored the use of the Millington reverberation time formulation.

$$T = 0,164 \frac{V}{-\sum_{i=1}^{n} S_i \ln\left(1 - \alpha_i\right) + 4mV} \tag{10}$$

Where is m the molecular absorption coefficient of air

Because of the chamber consists of multi-material boundaries and sound absorbing qualities can vary, it was assumed like the chamber has consisted of only single material. A full expression for the in Equation 10 is

$$\alpha = 1 - e^{\left(-\frac{0,164V}{T \cdot S} + \frac{4mV}{S}\right)} \tag{11}$$

5 Results

The main objective of measured results presented in the following was to examine crucial acoustic properties of empty reverberation chamber at FAI UTB in Zlin for acoustic measurements and consequently propose some improvements. In other words, it was necessary to determine whether the inter-laboratory space is able to perform acoustic data with required reproducibility and specify the frequency range in which a given reverberation chamber can be used.

5.1 Background noise

The background noise was measured in several days to be eliminated any temporal noises. Results and standards requirements are shown in Figure 5. As mentioned above the supporting constructions are consists of PUR panels. In fact, the entire chamber is surrounded by another compensation chamber formed by those same panels and thus considerably reduce the noise transmission between inner and external environment. As can be seen from the Figure 5 the background noise values are predominantly below the requirement values, nevertheless, this can not be achieved at the mid frequency 630 Hz. This phenomenon is most likely caused by the fact that the sandwich structure of PUR panel acts as oscillating plate and therefore at the resonant frequency sound will pass easier through the partitions, and so the sound insulation will be poorer [12]. Generally, it is important that the absorbing material is lightly packed, otherwise, it can form a vibration path bridging between the two partitions; this could greatly reduce the sound attenuation properties [12]. However, the results indicated that the background noise only slightly influences the evaluation of reverberation times T_{20}.

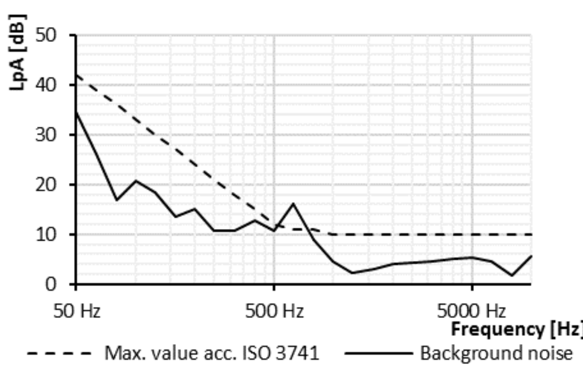

Fig. 5: Graph of measured and permissible levels of background noise.

5.2 Suitability of impulse sound source

The measurement has been made as simple as possible, focusing on the peak sound pressure level, in order to obtain a reference value, as well as on recording the waveform of the sound wave emitted by the explosion.

The determination of suitability of pistol shot as impulse sound source was done for three observation locations. According to the measured data (see Figure 6.) can be said that the pistol filled with a bullet will provide a typical N-pattern sound wave. The explosive speed is high enough to enable almost instantaneous release a high amount of energy for all frequency range (50 - 10000Hz) in very short time. More about the comparison of different impulse sound sources is described in [18].

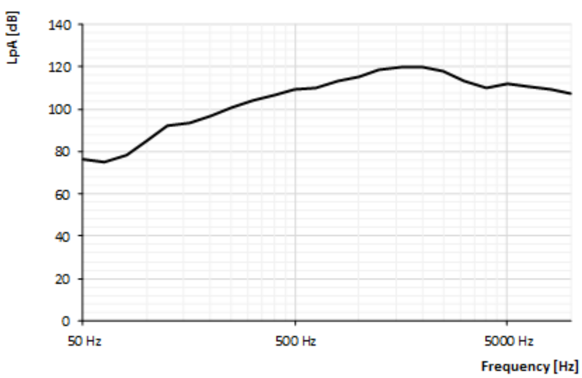

Fig. 6: Graph of maximum peak sound pressure level of pistol shot.

5.3 Analysis of sound decays

To properly characterize reverberant properties of the chamber, eighteen decay curves were obtained for each mid frequency band at different impulse sound source positions. Calculation results present in a clear way that in the analysed room reverberant response of the chamber and an initial amplitude of a sound decay strongly depend on the sound frequency [8], as indicated in Figure 7 and 8. They illustrate in each graphs only two curves because more of them would evocate confusing results.

To assess the repeatability was very important to observe the initial slope of decay curves.[19] In order to determine the decay times, the linear regression, according to the chapter 2.2, was used for finding fit lines to appropriate parts of the decay function. It is important to note that both decay times characterize a decrease in the sound energy level during the initial decay of sound. As can be seen from Figure 7, there are significant differences between each regression lines. For 50 Hz the excitation maximum peak sound pressure level was so low that the calculation could not be correctly executed. At frequencies, 63 to 125 Hz can be observed quite wide variances of the slope of the regression lines. This indicates that it is extremely important to average over many source positions if the sound power of a source with a significant content of pure tones is to be determined in a reverberation chamber. However, this frequency range is not suitable for performing laboratory measurements with high accuracy. The frequencies 160 and 200 Hz also shows weaknesses as well though not so heavy. Nonetheless, there is no doubt that the inner space for these frequencies is not sufficiently diffuse.

Equally important was to compare the smoothness of the reverberation decay for the third-octaves from 50 Hz to 10 kHz. In general, the smoothness of the decay increases as frequency is increased, this can be seen in both Figures 7 and 8. The sound power emitted by an impulse source in a lightly damped chamber cannot be expected to be unaffected by the reverberant sound field even when it is placed far from the walls of the chamber, especially when the rectangular shape is assumed. This can be seen, for example, 50 Hz a large wave fluctuations are noted suggesting that in this case the beating effect is present. The beat information on the low-frequency reverberation decay makes possible a judgment on the degree of diffusion [14]. This effect increases the relative standard deviation of calculated reverberation times.

For frequencies 63 and 80 Hz a characteristic property is a considerable different between decay times in the early and late stages of the decaying sound. A fundamental change of the response consists in a rapid decrease in a pressure in the initial stage of a sound decay. When the excitation impulse from the pistol shot energizes the inner space of the chamber, it excites one mode, and an instant later another mode. While the response shifts to the second mode, the first mode begins to decay. Before it decays very far, however, the random noise instantaneous frequency is once more back in the first mode, giving it another boost. All the modes of the room are in constant agitation, alternating between high and somewhat lower levels, as they begin to decay in between stimuli. It is strictly a random situation, but it can be said with confidence that each time the excitation noise is stopped, the modal excitation pattern will be somewhat different. These phenomena belong to the modal theory briefly described in chapter 2 and it's well described in [8]. For the frequencies in Figure 8, the analysed time domain decreases almost linearly. These decays provide information that inner space is more stable than frequencies below 250 Hz. Within the mid-frequency 200 Hz of third-octave band will be

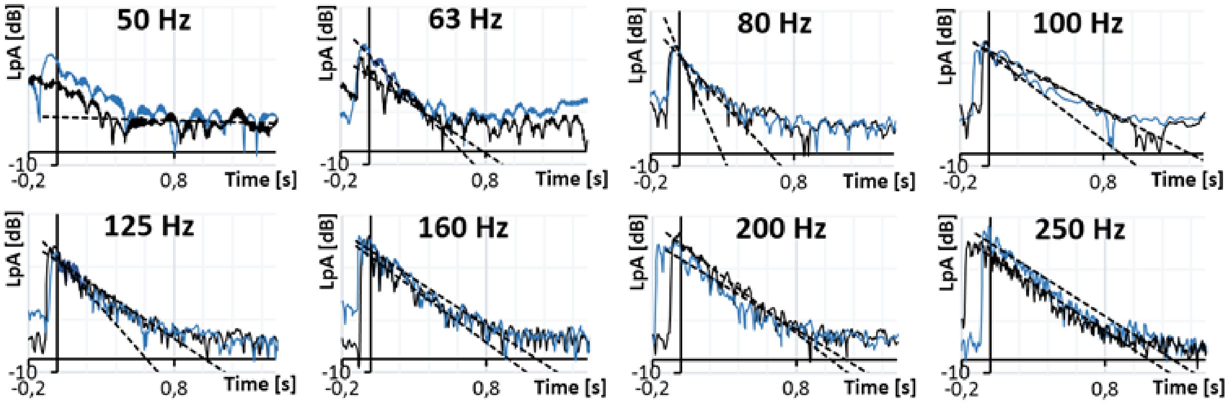

Fig. 7: Temporal decay of level LpA of pressure envelope for mid-frequencies of third-octave bandwidth about 50 to 250 Hz.

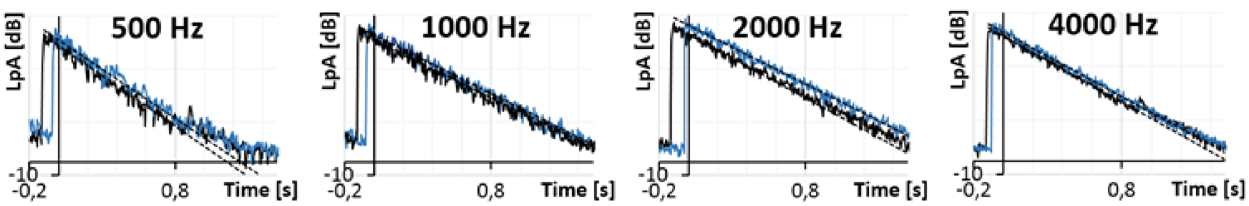

Fig. 8: Temporal decay of level LpA of pressure envelope for mid-frequencies of octave bandwidth about 500 to 4000 Hz.

located region where the transition boundary between the modal region where the room modes are well separated and the region of diffuse behaviour of a sound field, known also as the Schroeder region, where there is a high overlap of the room responses [20].

5.4 Acoustic treatment

It is obvious that a diffuse sound field will not be established in enclosures whose walls have the tendency to concentrate the reflected sound energy in certain regions [11]. In an attempt to avoid this regrettable phenomenon, it was applied a combination of two different geometric types of boundary diffusers (the tent and semicylinder type) and curved plexiglasses as hanging diffusers (see Figure 1). Generally, the diffusers can be used to break up the reflected wavefront and so disperse the focus while still maintaining the acoustic energy and avoiding absorption, however, it should be remembered that diffusers surface can cause significant diffraction from its edges [8]. The amount of diffraction depends on the wavelength and the size of the obstacle [21]. To minimalize the diffraction effect is recommended to accomplish an acoustic treatment. For boundary diffusers, this means using two different diffusers abreast and arranging them randomly or pseudorandomly on the wall. This will reduce periodicity and so improve dispersion [12].

The proposal of all three types of diffuse elements is based on the results from the first series of measurement with an empty chamber. From these values was calculated limited frequency f_s=243 Hz; however, it was much more than we expected. Hence, it was desirable to make the diffuse elements partially able to absorb the sound waves and as a result, the reverberation time will be shorter than it was before. The tenth type of boundary diffusers is made of plywood and plexiglass and the semicylindrical type is formed by plywood and KG pipe with an antivibration coat. The second series of measurement was per-

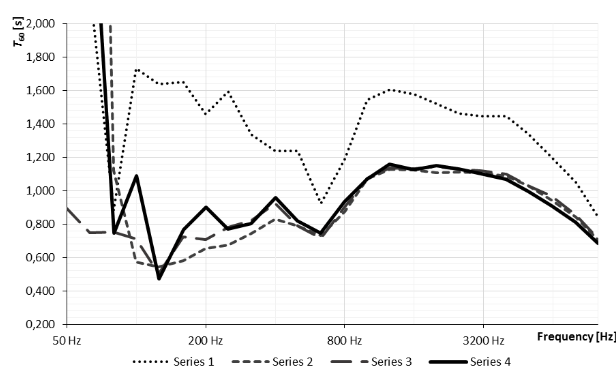

Fig. 9: Reverberation time curve for each measurement series (range 50 - 10000 Hz).

formed only with boundary diffusers arranging randomly on the wall. The total number of diffusers were 46 pieces (34x the tenth type, 12x semicylinder type) with total reflecting area 38,11 m^2. As can be seen in Figure 10 there is a significant decline in the overall curve of reverberation time (series 2). At that moment, the Schroeder frequency f_s has been bellowed the value 200 Hz. Nevertheless under 250 Hz is T_{60} less than another critical value according to the standard [9]. For the given space, this amount equals 0,73 s and indicates the minimum reverberation time at all measured frequencies. Therefore it was necessary to reduce the number of boundary diffusers (series 3). This treatment resulted in an increase T_{60} values at low frequencies (bellow 400 Hz). Considering the construction characteristics of hanging diffusers, we assumed that T_{60} values will continue to grow. As can be seen (Figure 9, series 4), the results confirm our assumption.

According to Equation 11 was calculated acoustic absorption for the entire space of the chamber (see Figure 10).

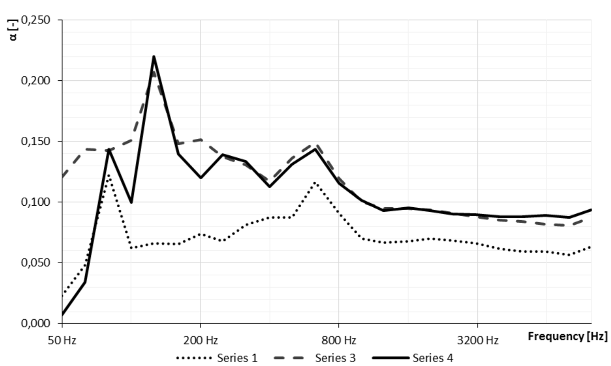

Fig. 10: The sound absorption coefficient for each measurement series (range 50 - 10000 Hz).

6 Conclusion

Reproducibility of acoustic data measured in acoustic laboratories is difficult to achieve for wide frequency bandwidth. For acoustic laboratories that perform measurements with high accuracy, it is necessary to know the frequency range where the sound fields are adequately diffuse and where not. In this paper were compared the decay curves obtained in several positions. These decays are not identical, and differences can be attributed to the non-uniform character of an acoustic field through the chamber. In particular, the fluctuations in the decays result from beats between closely spaced modes. Because the excitation level of the modes is constantly shifting, the form and degree of the beat pattern shift from one decay to another depending on where the random excitation happens. There is much information in each decay, and acoustical flaws can often be identified from aberrant decay shapes. It is obvious that a diffuse sound field will not be established in enclosures whose walls have the tendency to concentrate the reflected sound energy in certain regions. For this reason, diffusers which have been made, help to establish a diffuse sound field by continuously redistributing the energy in all possible directions. Upon each reflection is the fraction of energy diffusely scattered into non specular angles while the remaining energy is reflected into the specular reflection angle in the usual way. The diffuse energy is ambient energy spread throughout the room volume. The number of diffuse elements has been optimized during the verification process; however, the dispersion of measured data at low frequencies is still evident.

The work could be extended to include the effects of edge diffraction, which previously in this paper have been shown to cause significant estimation errors in measured data. The work also needs to make a simulation model of the wave distribution in the chamber, to further problem frequencies optimization.

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References:

[1] P. Drabek, “Testing acoustics parameters for devices of technical environment in calorimetric chamber,” Master’s thesis, Tomas Bata University in Zlin, 2014.

[2] K. C. P. F. A. Everest, *Master Handbook of Acoustics*. The McGraw-Hill Companies, 2009, vol. 5.

[3] A. C. Gade, “Acoustics for symphony orchestras; status after three decades of experimental research,” *Building Acoustics*, vol. 18, no. 3-4, pp. 181–206, 2011.

[4] L. Beranek, *Acoustics*. Acoustical Society of America, 1996, vol. 5.

[5] F. Jacobsen, “The sound field in a reverberation room,” Master’s thesis, Technical University of Denmark, 2011.

[6] J. A. D. T. Bradley, M. Muller-Trapet and M. Vorlander, "Comparison of hanging panels and boundary diffusers in a reverberation chamber," *Building Acoustics*, vol. 21, no. 2, pp. 145–152, 2014.

[7] V. W. W. Weise, "Using round robin test results for the accreditation of laboratories in the field of building acoustics in germany," *Building Acoustics*, vol. 12, no. 3, pp. 189–206, 2005.

[8] A. S. A. Prato, F. Casassa, "Reverberation time measurements in non-diffuse acoustic field by the modal reverberation time," *Applied Acoustics*, vol. 110, pp. 160–169, 2016.

[9] "Iso 3741 acoustics - determination of sound power levels and sound energy levels of noise sources using sound pressure - precision methods for reverberation test rooms," European Committee for Standardization, Prague, CZ, Standard, 2010.

[10] "Iso 354 acoustics measurement of sound absorption in a reverberation room," European Committee for Standardization, Prague, CZ, Standard, 2003.

[11] H. Kuttruff, *Room Acoustics*. Spon Press, 2009, vol. 5.

[12] P. D. A. T. J. Cox, *Acoustic Absorbers and Diffusers*. Taylor And Francis, 2009, vol. 2.

[13] M. Meissner, "Prediction of reverberant properties of enclosures via a method employing a modal representation of the room impulse response," *Archives of Acoustics*, vol. 41, no. 1, pp. 27–41, 2016.

[14] ——, "Evaluation of decay times from noisy room responses with pure-tone excitation," *Archives of Acoustics*, vol. 38, no. 1, pp. 47–54, 2013.

[15] "Iso 3382-2 acoustics measurement of room acoustic parameters part 2: Reverberation time in ordinary rooms," European Committee for Standardization, Prague, CZ, Standard, 2008.

[16] P. Z. C. Passero, "Statistical comparison of reverberation times measured by the integrated impulse response and interrupted noise methods, computationally simulated with odeon software, and calculated by sabine, eyring and araupuchades formulas," *Applied Acoustics*, vol. 71, no. 12, pp. 1204–1210, 2010.

[17] M. R. Schroeder, "New method of measuring reverberation time," *Acoustical Society of America*, vol. 37, no. 3, pp. 409–412, 1965.

[18] H. D. M. Horvat, K. Jambrosic, "A comparison of impulse-like sources to be used in reverbertion time measurements," *Acoustics 08 Paris*, vol. 17, no. 3, pp. 409–412, 1965.

[19] F. F. J. Nannariello, "The prediction of reverberation time using optimal neural networks," *Building Acoustics*, vol. 9, no. 1, pp. 5–28, 2002.

[20] M. Meissner, "Influence of wall absorption on low-frequency dependence of reverberation time in room of irregular shape," *Applied Acoustics*, vol. 69, no. 7, pp. 583–590, 2008.

[21] E. Deines, "Acoustic simulation and visualization algorithms," Ph.D. dissertation, Vom Fachbereich Informatik der Technischen Universitat Kaiserslautern zur Erlangung des akademischen Grades, 2008.