Design of the Specialized Tube for Sound Transmission Loss Measurements

Marko Ličanin, Ana Đorđević and Dejan Ćirić

***Abstract*—Traditional measurements of the transmission loss (TL) have been performed for many years using the proper infrastructure. This requires accessibility to the specialized facilities such as reverberation chamber, which is available only to a small number of laboratories throughout the world. In order to extend the possibility of transmission loss measurements to a wider population of acousticians, scientists are searching for new measurement methods and algorithms. This paper exploits a method for TL estimation using a modification of a standard impedance tube, usually applied for absorption coefficient measurements. Focus of this paper is on the design of the tube as well as its evaluation through repeatability of measurements. In addition, testing is using a sound insulation sample and the results of the obtained TL are discussed.**

***Index Terms*—Transmission loss, acoustic measurements, impedance tube, sound insulation, metamaterials**

I. Introduction

Several methods have been established for measuring the sound insulation of various structures. This has been performed in both laboratory and field environments, that is in-situ. According to different versions and updates of ISO 10534 standard [1], a number of indexes are defined, which offer various benefits of testing a partition insulation properties of any kind.

In a laboratory, sound transmission loss (TL) measurement is usually performed using two reverberation rooms with the pressure method or one reverberation room and anechoic chamber employing the sound intensity method. Sample material under test is mounted in between the two rooms, where a sound source is always placed in a reverberant room. The sample has to be large in order to avoid sound flanking at lower frequencies, where wave length could be higher than sample dimensions. These types of measurements provide the best precision. However, having such facilities, especially joint reverberation and anechoic chamber is a very rare commodity throughout the world.

An alternative to standard measurement of TL is related to using scale methods that do not require expensive specialized rooms. So far, there have been a few papers that exploit the use of a typical impedance tube, modified for this type of testing [2]. These are experimental setups and each of those has differences in design and mathematical approach for extracting the TL parameter. There are even several commercially available devices, such as one shown in Figure 1. Their price is still fairly high for the budget of smaller laboratories, and having the specialized software, they might not be suitable for scientific research.

Fig. 1. Commercially available transmission loss tube by "Akustik Dorschung Dresden"

This paper deals with design of the transmission loss tube as an alternative to standardized pressure and intensity methods in anechoic and reverberant rooms. Motivation for building such a device is the increase in demand of thin material solutions for sound insulation that will provide higher level of transmission loss at lower frequencies. These experimental compounds include metamaterials, and they are in an early stage of development in many current studies.

The tube is designed for a frequency range 100 Hz – 3 kHz. The process of building the device is explained and test measurements are performed. Repeatability of the impulse response (IR) measurements is analyzed and the results are discussed. Initial test measurements of a composed sound insulation sample are done and calculated TL is observed.

II. Theory of Transfer Function Method

A. Sound Field Representation

Sound pressure and normal particle velocities in the *upstream* and *downstream* segments in the tube are denoted in Fig. 2. Those parameters are expressed as four pole matrix:

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$$\begin{bmatrix} p \\ v \end{bmatrix}_{x=0} = \begin{bmatrix} T_{11} & T_{12} \\ T_{21} & T_{22} \end{bmatrix} = \begin{bmatrix} p \\ v \end{bmatrix}_{x=d} \quad (1)$$

where p and v are the complex pressure and complex particle velocity [3]. Elements T_{ij} are directly related to the acoustical properties of sample, and sound transmission coefficient and transmission loss can be expressed as [4]:

$$TL = 10\log\left(\frac{1}{4}\left|T_{11} + \frac{T_{12}}{\rho c} + \rho c T_{21} + T_{22}\right|\right) \quad (2)$$

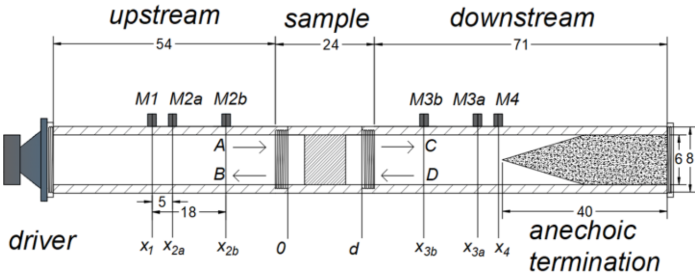

Fig. 2. Cross-section of the device. Microphone positions, construction elements and most important dimensions are shown.

For determining the elements of the transfer matrix and sound transmission loss, complex sound pressure and velocities at both surfaces of the sample have to be determined as well. For the current derivation, we will consider sound pressure instead of the transfer functions, and denote it by P_1 to P_4. These are calculated using the following equations:

$$P_1 = Ae^{-jkx_1} + Be^{jkx_1} \quad (3)$$

$$P_2 = Ae^{-jkx_2} + Be^{jkx_2} \quad (4)$$

$$P_3 = Ce^{-jkx_3} + De^{jkx_3} \quad (5)$$

$$P_4 = Ce^{-jkx_4} + De^{jkx_4} \quad (6)$$

Equations (3)-(6) can be rearranged for solving the coefficients A to D. This provides the input data for the subsequent transfer matrix calculation:

$$A = \frac{j(P_1\,\mathrm{e}^{-jkx_2} - P_2\,\mathrm{e}^{-jkx_1})}{2\sin k(x_1 - x_2)} \quad (7)$$

$$B = \frac{j(P_2\,\mathrm{e}^{-jkx_1} - P_1\,\mathrm{e}^{-jkx_2})}{2\sin k(x_1 - x_2)} \quad (8)$$

$$C = \frac{j(P_3\,\mathrm{e}^{-jkx_4} - P_4\,\mathrm{e}^{-jkx_3})}{2\sin k(x_3 - x_4)} \quad (9)$$

$$D = \frac{j(P_4\,\mathrm{e}^{-jkx_3} - P_3\,\mathrm{e}^{-jkx_4})}{2\sin k(x_3 - x_4)} \quad (10)$$

In this paper it has been consider that the sample is backed by a perfectly anechoic termination, the coefficient D is equal to zero and the transmission loss can be calculated by dividing the coefficients C and A.

A major advantage of the transfer matrix approach presented here is that the transfer matrix elements represent only properties of the sample, and not of the measurement environment. Further, when those elements are known, the sound power transmitted by the sample can be calculated for any tube termination condition. When the calculation is based on a perfectly anechoic termination, as in (2), the corresponding transmission loss gives a true indication of the sample's TL performance.

For calculation of the transmission loss, the transfer function H_{xr} between the reference and other microphones is used

$$H_{1r} = \frac{p_r}{p_1} = \frac{P_r(\omega)P_{1c}(\omega)}{P_1(\omega)} \quad (11)$$

Because only 4 data acquisition channels are connected, the microphone 1 (M1) is used as a reference. Letter c in subscript denotes the calibration signal.

The coefficients A and C from the previous equations can be expressed in the frequency domain. This is done using the following equations:

$$A = \sqrt{G_{rr}}\,\frac{j(\mathrm{H}_{r1}\mathrm{e}^{jkx_2} - H_{r2}\,\mathrm{e}^{jkx_1})}{2\sin k(x_1 - x_2)} \quad (12)$$

$$C = \sqrt{G_{rr}}\,\frac{j(\mathrm{H}_{r3}\mathrm{e}^{jkx_4} - H_{r4}\,\mathrm{e}^{jkx_3})}{2\sin k(x_3 - x_4)} \quad (13)$$

where G_{rr} is the autospectrum of reference signal [5]. It is also necessary to compensate for the propagation delay between the loudspeaker and the various microphones to avoid the introduction of a time delay bias errors [6].

By dividing C over A, the transmission loss coefficient T_a can be calculated as:

$$T_a = \frac{H_{1r}e^{jkx_2} - H_{2r}e^{jkx_1}}{H_{3r}e^{jkx_4} - H_{4r}e^{jkx_3}} \quad (14)$$

Finally, the transfer loss function TL is:

$$TL = 10\log\frac{1}{|T_a|^2} \quad (15)$$

III. BUILDING PROCESS OF THE TL TUBE

Realization of the device is divided into two segments, the first being the preamplifier circuit, while the second is tube itself with mounting components. Amplifying the signal from microphone elements is necessary as they produce only about 5 mV and require polarization voltage up to 10 V. These transducers are electret ones, and even there are far better solutions with more costly condenser variants, they proved to be fairly decent in term of sensitivity and frequency response flatness.

A. *Preamplifier Circuit*

In the core of the circuit, there is the SSM2019 self-containing audio amplifier chip, which proved itself in the devices that were realized in the past. [7] It has very good SNR and gives no distortion up to 60 dB of amplification. Only one resistor is required to adjust the gain, which is set to 30 dB using 330 Ω value. The preamplifier has 6 identical channels, but only 4 are connected to output as the same number of microphones are currently used on the TL tube. The entire circuit is powered by symmetric power source using two 9V batteries, which removes the necessity of constant voltage power source and filtering the 50 Hz power grid voltage frequency. The microphones are connected to the circuit using BNC connectors, and amplified output is sent to the recording interface through TS mono connectors and cables. The entire device is placed inside a box with an on/off switch for protection purpose. The device interior can be seen in Fig. 3.

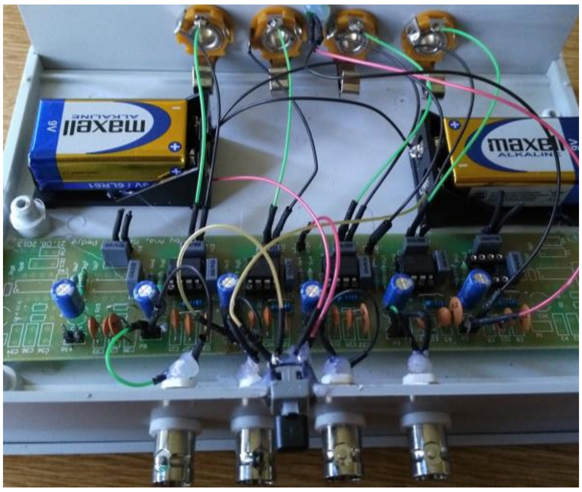

Fig. 3. Inside look of the preamplifier circuit box. Connectors, wiring, SSM2019 and battery power can be seen. Four channels are connected.

B. *Tube Design*

The transmission loss tube is composed of three sections, each being of the different length. The left section is called *upstream* [4, 5], and at its beginning a loudspeaker driver is placed. The right section named *downstream* is typically longer and has an anechoic termination. The middle section serves as a cartridge for the testing sample and has the smallest length (see Fig. 4)

Dimensions of the TL tube can be calculated based on several parameters that are correlated with the desired frequency range of the measurements [5]. Throughout the design process, focus is on the possibility of using the device for measurements in the frequency range 100 Hz – 3 kHz. Below this range, it is still very difficult to achieve good sound insulation without considering thick samples, according to the “mass law”. Above the mentioned frequency range, it is not so important to make the analysis, since most of the standard material, such as gypsum board coupled with brick wall, should provide decent sound insulation.

Fig. 4. Components of the tube, *upstream*, *downstream* and middle section tubes.

The upper cut-off frequency based on the size of the device is calculated as:

$$f_{up} = \frac{k \cdot c}{d} \tag{16}$$

where c is the speed of sound, k is a constant and d is diameter of the tube in meters [3, 4]. Frequency range also depends on mutual distances between a pair of the microphones in the *upstream* section. Since microphone positions are mirrored in relation to the center of the middle section, the same statement is also valid for the *downstream* side. Here, upper and lower cut-off frequency limits are calculated as:

$$f_{low} \geq \frac{c \cdot 0.05}{s}, \quad f_{up} \leq \frac{c \cdot 0.45}{s}, \tag{17}$$

where s is the distance between pair of microphones at one side. If s is considered to be 5 cm, than the lower and upper frequencies will be 343 Hz and 3 kHz respectively. The upper limit is thus satisfied, but lower frequency is not. If the distance is increased, the upper limit will be lower. To solve this, additional microphone can be used at both *upstream* and *downstream* sides. This microphone can be paired to the one of the microphones from the previous pair, and form a longer distance s. Considering (14), in order to achieve the lower frequency limit of 100 Hz, s must be higher than 17.15 cm. The upper limit is then 900 Hz. Since now, there are two sets of distances between microphones, giving different frequency ranges for proper measurement. Combining the data measured at both of the cases can give the results in the desired frequency range (100 Hz - 3 kHz).

Having this in mind, the distances between the microphones of 5 cm and 18 cm are chosen. To ensure anechoic termination, *downstream* tube is set to be longer, so enough of absorption material can be stuffed inside (40 cm of acoustic polyurethane foam). This is shown in Fig. 2.

For the purpose of reasonable size and weight of the device, 6 cm inner diameter is chosen. Thickness of 1 cm provides good sound insulation and prevents sound leakage. The tube is excluding the loudspeaker 150 cm long, and even with relatively small inner dimensions, it is still heavy (approximately 12 kg).

In order to have a flexible microphone mounting solution, special microphone casings are made. The elements are then inserted into these casings to ensure perfect fit, and ease of the mounting and demounting. This setup gives freedom of upgrading the device in some later stage by inserting a better quality condenser transducers. The microphone elements inside the casings are shown in Fig. 5.

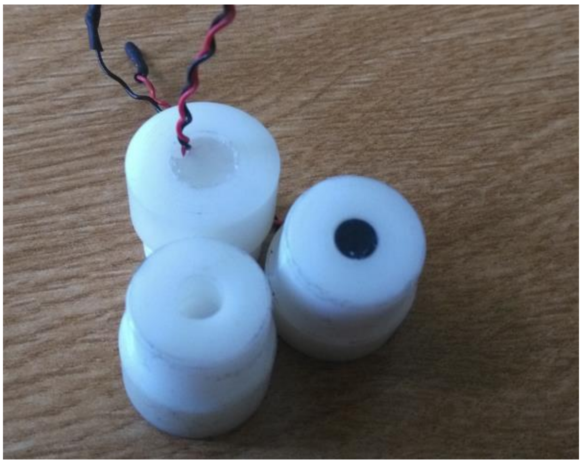

Fig. 5. Microphone elements fitted to the casing. Wires movement is restricted with silicon glue.

IV. Evaluation

A. Measurement Setup and Procedure for TL Calculation

The transmission loss tube is connected to the battery powered preamplifier circuit via shielded cables. Once amplified, signals are fed to the recording interface M-Audio Profire. Recordings are performed using 2 channels at the time and Adobe Audition software. In the excitation part, output of the recording interface is connected through the power amplifier to the driver attached at the *upstream* end of the tube. The driver is a bass loudspeaker with a kevlar cone, particularly intended for frequencies below 5 kHz. The *swept sine* signal of the length of 15 s is then reproduced and recorded at various microphone positions. The measurement setup is presented in Fig. 6. The recorded measurements are imported into Matlab, where IRs are extracted and the results are further analyzed.

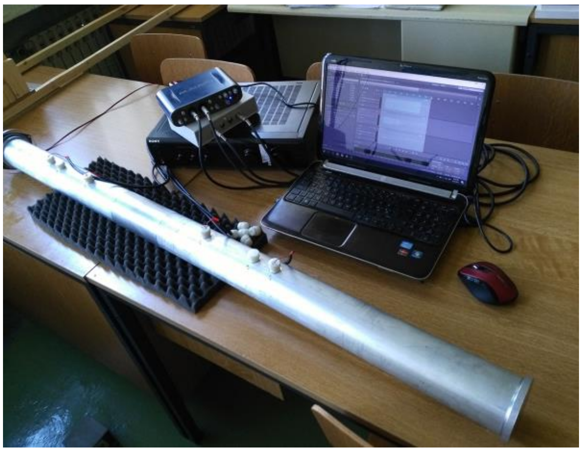

Fig. 6 Measurement setup for recording inside the tube.

B. Repeatability of Measured Responses

Latency is typically a time delay caused by hardware and software of the recording interface and computer. It has a negative effect in time sensitive measurements, and it is in most cases considered as not predictable [7]. Fig. 7. presents the latency of the IRs measured 5 times by all 4 microphones placed at the position of the reference microphone (microphone 1). It is notable that variation is rather high. The results are here presented in reference to the least delayed IR.

For this situation, the latency does not affect the precision of the measurements, as it must be eliminated in the process of the transmission loss calculation. All of the extracted IRs are set to start at the same time position (recording and propagation latency) and further processing is done on these time shifted responses. The latency between the channels can be removed if an interface with more parallel channels is used.

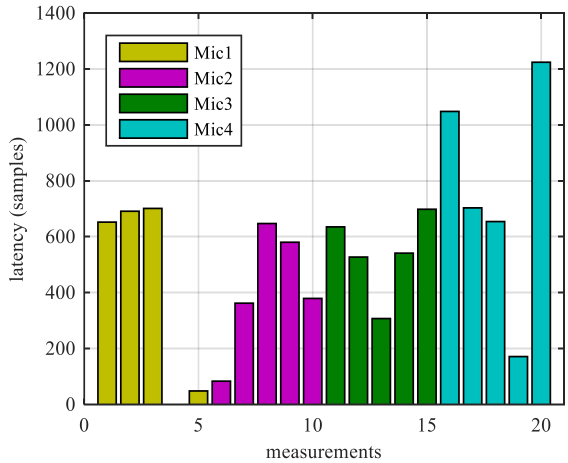

Fig. 7. Repeatability of IRs (measured by all 4 microphones placed in the position of the microphone 1 and repeated 5 times) regarding relative latency presented as a difference from the least delayed IR. Sampling frequency of 44.1 kHz was used.

C. Repeatability of IR magnitude

Analysis of this step is very important as it can provide information about inconsistencies in the gain between the channels in the preamplifier circuit, combined with the differences in the microphone element sensitivities [8]. For this purpose, the IRs are extracted from 5 repeated measurements by 4 microphones in the position of the microphone 1. Maximum of the magnitudes is then determined and mean (average) values of those magnitudes (separate mean value is determined for each microphone) are calculated. Differences of the maximums from the average value is expressed in percentage and shown in Fig. 8.

It can be observed that the greatest deviation from the average value, of 5 measurements for microphone 4, is less than ±0.1 %. It is a very good result. In Fig.8 results are not compared between microphones, but rather repeatability of measurements at each microphone position is shown. To completely remove existing differences between all recording positions, a calibration files are stored and used in all subsequent measurements. The same study is repeated for minimum values as well as dynamic ranges of IRs. The results are similar.

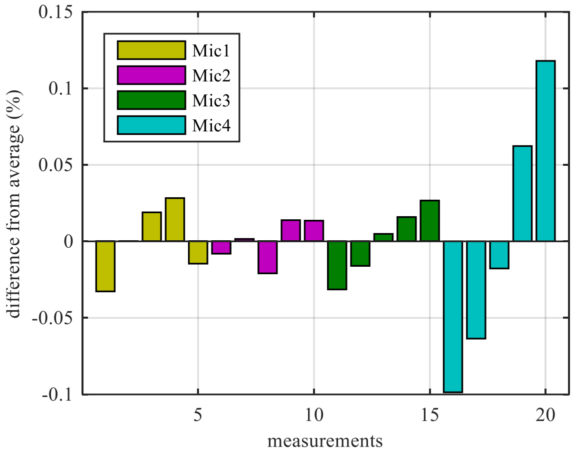

Fig. 8. Repeatability of magnitude of IRs (measured by all 4 microphones placed in the position of the microphone 1 and repeated 5 times) calculated as a difference from the average magnitude value.

D. Repeatability of IR frequency response

Fig. 9. shows the frequency responses of all 4 microphones, measured at position M1, under the same conditions. As in the case of the magnitude repeatability, it can be seen that consistency of frequency responses for different microphones is very good. All of the responses follow almost the same trend throughout the entire frequency range. Notches and peaks in the responses are caused by reflections and resonances in the tube, as for the repeatability measurements the anechoic termination is not placed at the end of the *downstream* tube.

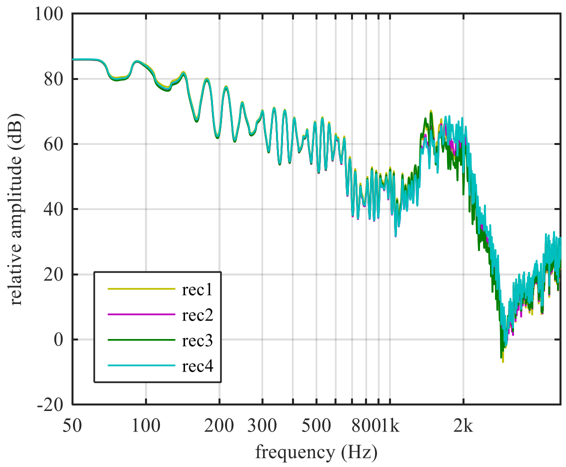

Fig. 9. Repeatability of IRs frequency response of 4 microphone, measured in position M1.

V. TL Measurements

A. Tube Without the Sample

In order to evaluate the transfer function method and behavior of the tube, the TL measurements are carried out without any test sample inserted in the middle section. The obtained results are used to determine a correction factor for the sound transmission loss measurements. During the measurements, cylindrical anechoic termination, composed of poliuretan foam of length of 25cm was placed at the end of the *downstream* tube. This represents approximation of anechoic condition, and for the future research this results will be compared with results were the reflections are considered.

The responses measured by pairs of microphones at 5 cm distance (*M1–M2a* and *M4–M3a*) and at 18 cm distance (*M1–M2b and M4–M3b*) are taken for the TL calculation in accordance with the approach described in section II. In every measurement, the response of the reference microphone (*M1*) is measured with the response of another microphone. This is done for the purpose of calibrating previously discussed differences in latency. For each microphone (position), 5 measurements are performed, latency is removed and IRs are averaged. The effects of differences in measurement channels including microphone sensitivities are eliminated using the calibration responses. The microphone pair with a larger distance (18cm) provides more reliable results at lower frequencies, see Fig. 10. The measurements done by the pair at 5 cm distance give more reliable values in the middle and high frequency range.

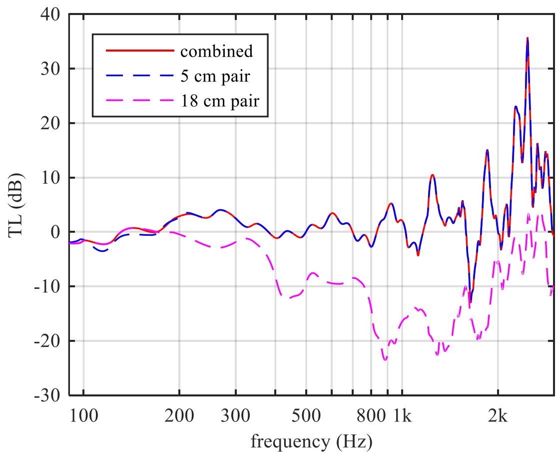

Fig. 10. Combined TL of two different microphone pairs.

By combining those two microphone pairs the TL is determined in the desired frequency range. Since the tube is empty in this case, it can be expected that the values of the responses measured in both the *upstream* and *downstream* tube are similar except differences caused by different positions of the microphones. This can be observed in Fig. 11. In theory, when the tube is empty and ratios of pressures at two microphone pairs in the *upstream* and *downstream* tube are the same, TL is equal to 1. In the obtained results, there are variations of TL (± 5dB up to 1 kHz), as shown in Fig. 10, caused by the resonances in the tube. Thus, further mathematical study is necessary in order to entirely remove the influences of the tube in calculation of the TL parameter.

B. Tube with sample

For a final set of measurements, a sample material is inserted in the middle part of the tube. It is made as a “sandwich” consisting of two layers of geo textile and a layer of visco-elastic membrane in the middle. Pieces were glued using silicon. In this case only the measurements done by the microphone pairs at 18 cm (*M1 – M2b* and *M4–M3b*) distance are used. TL of the measured sample is corrected using TL obtained for the empty tube. The results are presented in Fig. 12.

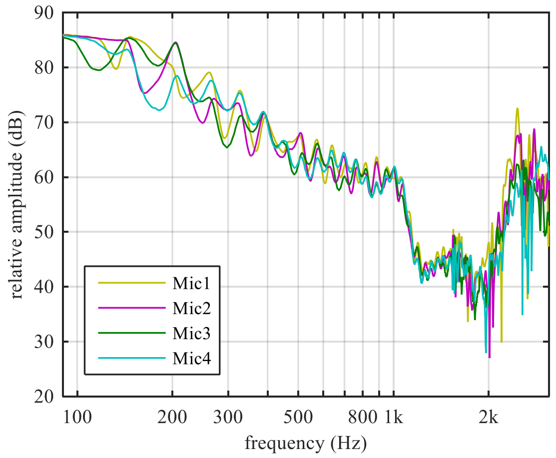

Fig. 11. Frequency response recorded at 4 microphone positions in the case of empty tube

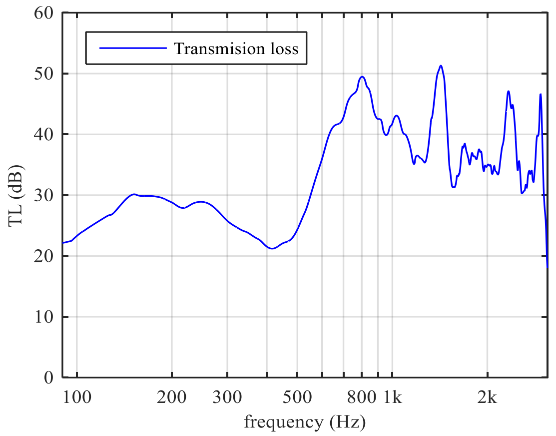

Fig. 12. TL calculated using microphone pairs *M1–M2b* and *M4–M3b*

This indicates that the sample reduces the values of sound pressures (responses) in the *downstream* part of the tube. This can be seen in Fig. 13, where the frequency responses of the *upstream* and *downstream* measurements are presented.

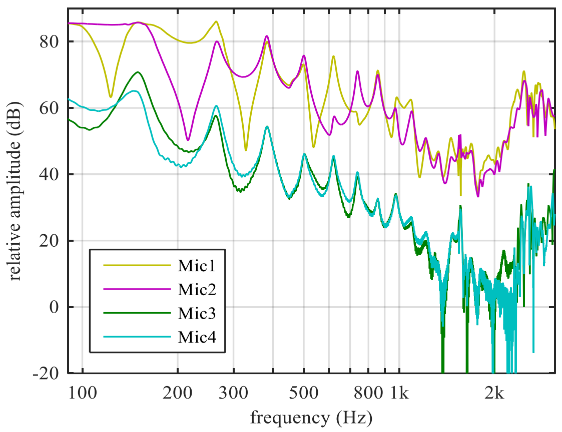

Fig. 13. Frequency responses of signals (IRs) recorded at 4 microphone positions in the case of sample placed in a middle section of the tube

VI. Conclusion

This paper analyses the possibility of using an impedance tube for transmission loss measurements in situations where sophisticated facilities are not available. Design of the tube is presented with regards to the frequency range intended for metamaterial measurements. To test the realized device, repeatability of the latency, IRs magnitude and frequency response is observed using the responses measured by different microphones. The initial results show that the measured signals have very good consistency, which increases precision of the measurements.

According to the mathematical approach described in section II, transmission loss measurements and calculation are realized. Test of the transmission loss of the empty tube shows the behavior of the tube itself and its frequency response. Using these measurements, the correction signals can be calculated and used in the TL calculation. The procedure is repeated in the case when the testing sample is placed inside the middle section of the tube.

Realization of this device and analysis of the results are initial steps towards building a prototype for future measurements. Even though that hardware parts have been finished, there is yet a lot to improve regarding processing of signals and eliminating various impacts such as effects of repositioning of the microphones, differences among the microphones and measuring channels in general, resonances of the tube, etc. This serves as a motivation for future analysis and research in order to achieve a high reliability of the measurements.

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