A NEW APPARATUS FOR MEASURING THE ABSORPTION COEFFICIENT FOR NORMAL INCIDENCE

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1 INTRODUCTION

At present the measurement of the sound absorption for normal incidence is usually done by means of a standing wave tube. The apparatus can be described as a tube of circular or squared cross section whose terminations are equipped respectively with a loudspeaker on one hand and with the sample to be tested on the other. The waves impinging the sample at a normal incidence are assumed to be plane and the stationary field inside the tube can be described as the superposition of one or more standing waves. To achieve the sound absorption coefficient of the material there are basically two methodologies known respectively as "standing wave ratio"[1] and "transfer function method"[2]. According to the former method a single microphone is moved along the tube to find the minima and the maxima of the sound pressure of the standing wave and the procedure has to be repeated for each pure tone of interest. Despite its simplicity this method requires a long time for its execution and has been replaced by the latter method. The "transfer function method" in fact is much faster and the algorithms are easily implemented by FFT analysers. The results are shown as curves of sound absorption α_n vs. frequency but the band values are not easily derived. On the contrary the α_n values often need to be reported as 1/3 or 1/1 octave values, both to compare with other procedures like the diffuse field methodology[3] or simply for acoustical design purposes. Anyway a drawback of the method seems the need of two microphones instead of the single microphone as in the "standing wave method".

The sound absorption coefficient of a test sample can be derived also by another method which is based on the measurement of the reverberation time inside the tube. In fact the transient properties of the sound field in a closed tube show that the succession of wave fronts obeys a simple exponential rule so that the impact of the sound absorption of the termination can be predicted analytically. Though a similar procedure was outlined by Beraneck[4], in this work the apparatus used and the processing involved are different. In particular, as it will be described in what follows, a new apparatus was developed with both ends possibly equipped by test samples. The sound is injected in the tube by a hole on the side. The algorithms and the signal processing are also sketched and the comparison with the "transfer function method" is shown for several types of sound absorbing materials.

2 THEORY

To examine the theoretical background of the method we can consider a tube whose terminations are both equipped by a sound absorbing material. That on the left is called α_L and that on the right is α_R. The tube length is *l* and its section is S. The sound power W is introduced in the tube by a small hole on the side which is connected via a small tube to a loudspeaker cabinet. The system is represented in Fig. 1

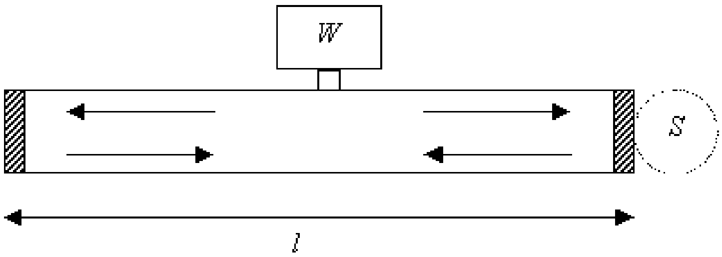

Figure 1. A sketch of the measurement apparatus with possibly two test samples at both ends.

The sound entering the tube moves in both directions and can be represented as a superposition of plane progressive waves that, under stationary conditions, build up the standing waves. The system is characterised by a mean free path for the waves equal to the length *l.* After each impact with the samples at the ends the intensity of the waves is reduced. The intensity of the first wave entering the tube can be indicated as:

$$I_0 = \frac{W'}{S} \qquad (1)$$

where the prime tells us that half of the sound power goes left and half goes right. After *n* reflections the modulus of the intensity can be expressed simply as:

$$I_n = \frac{W}{S}(1-\alpha_L)^{\frac{n}{2}}(1-\alpha_R)^{\frac{n}{2}} \qquad (2)$$

where half of the reflections are assumed on each sample. The order of reflection can be easily derived by the mean free path as *n=ct/l* (c being the sound velocity) and the expression (2) can be rearranged as follows:

$$I(t) = \frac{W}{S}e^{-\gamma ct} \textbf{ with } \gamma = -\frac{\ln[(1-\alpha_L)(1-\alpha_R)]}{2l}\textbf{;}\ \gamma > \mathbf{0}\ . \qquad (3)$$

Equation (3) gives an estimate of the intensity amplitude of the peak occurring at time t in the energetic impulse response in the tube. It is to be noted that the position within the tube does not matter since we are dealing with plane waves whose amplitude is not decreased by any divergence in the travel but just by discrete energy losses at the boundaries. A further step can be achieved by collecting all of the singular energy contributions (regardless of their sign) in a backward integration procedure known also as Schroeder plot[5]. This transformation gives us the proper energy decay of the sound in the tube:

$$I_{tot}(t) = \int_t^\infty I(\tau)d\tau = \frac{W}{\gamma cS}e^{-\gamma ct} \qquad (4).$$

It is to be noted that the physical energy units of equation (4) are consistent with the time integration. Equation (4) is the decay of the sound energy so that it is allowed to implement the definition of reverberation time expressed by:

$$\frac{I_{tot}(t)}{I_{tot}(t_0)} = 10^{-6}\ . \qquad (5)$$

From the above definition the reverberation time is indicated by $T = t - t_0$ and after simple calculations the algorithm is completed as follows:

$$e^{-\gamma c(t-t_0)} = e^{-\gamma cT} = 10^{-6}$$
$$T = -\frac{2l \cdot 6\ln 10}{c\ln[(1-\alpha_L)(1-\alpha_R)]}. \tag{6}$$

The absorption coefficient at one end can then be easily calculated by equation (6) if the reverberation time and the other absorption coefficient are known:

$$\alpha_L = 1 - \frac{1}{1-\alpha_R} e^{\left(-\frac{2l \cdot 6\ln 10}{cT}\right)} \tag{7}$$

Equation (7) is the base of the new method for the determination of the absorption coefficient by measurement of the reverberation time.

3 MATERIALS AND METHODS

The new measurement apparatus consists in a perspex tube equipped with two sample holders at the ends each having a movable metal backing. On the side of the tube there are six holes to insert the microphone by a firm seal. A loudspeaker is located in a cabinet made of perspex and folded with sound absorbing material: a small perspex tube connects the front of the speaker to the main tube. The dimensions of the main system components are resumed in Tab. 1.

Max tube length (*m*)	1,45
Internal tube diameter (*m*)	0,1
Thickness of the sample holder (m)	0.03
Cabinet volume (m^3)	0.24x0.20x0.24
Length of small tube (*m*)	0.04
Internal diameter of the small tube (*m*)	0.01

Table 1. Dimensions of the system components.

Fig. 2 and Fig. 3 represent respectively the project and the construction of the measurement apparatus. The dimensions of the tube, its length and section, were chosen to comply with the formula $f_{max} \cdot l \leq 200$ fixing the upper frequency of the waves propagating in the tube and the lower one was achieved by a sufficiently long tube coupled with a loudspeaker with a suitable response in the lower range. The useful frequency range of this configuration of the apparatus is thus from 100 Hz to 1600 Hz.

The components of the measurement chain are: a B&C 6 PEV 13 loudspeaker whose frequency response is optimal in the range of interest, a sound card MOTU2408, an amplifier B&K2706 and finally a ¼ " B&K4939 condenser microphone with a B&K5934 signal conditioner. Fig. 4 shows the set up for the measurements and the connections.

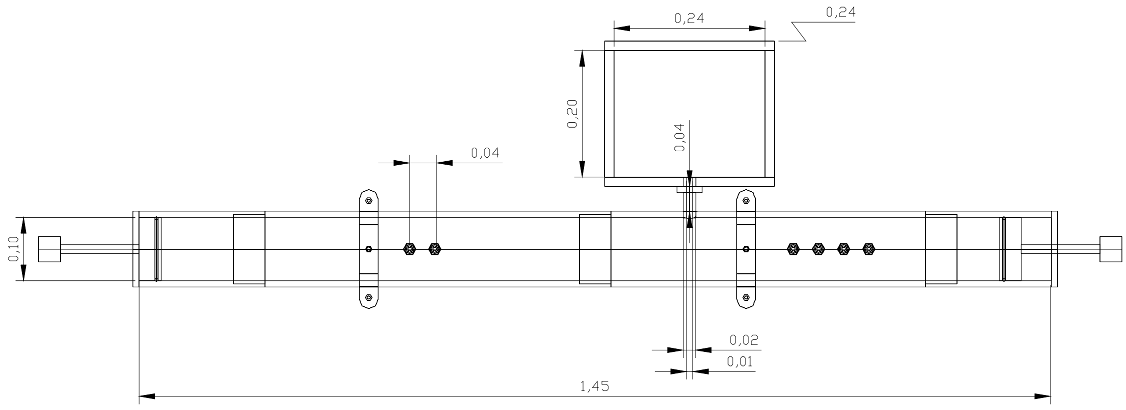

Figure 2. The new apparatus: design (dimensions in meters).

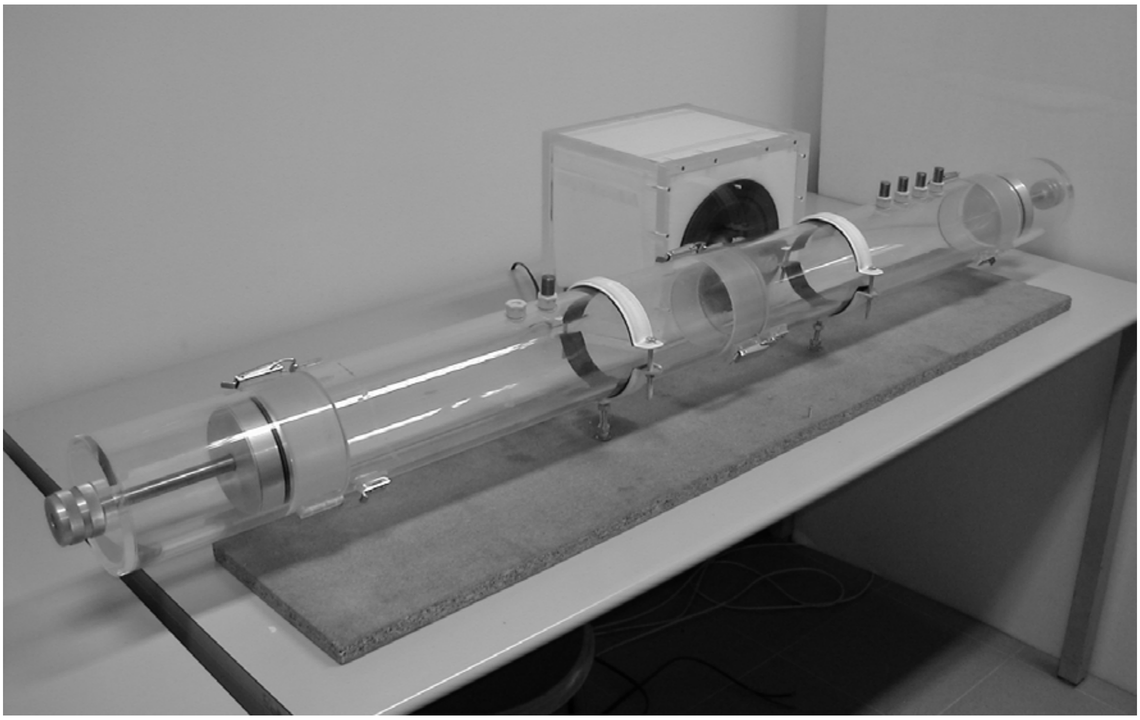

Figure 3. The new apparatus as it appears once built.

The reverberation time in the tube is obtained by means of the preliminary measurement of the impulse response. To achieve the impulse response a methodology known as "exponential sine sweep" was implemented. For this specific system and in order to profit from a large S/N ratio a 25s was used between 80 Hz and 2000 Hz. All of the operations are done with the Aurora package in the Cooledit environment. Finally the reverberation time is calculated as 1/3 octave band values and the absorption coefficients are calculated in the same bands.

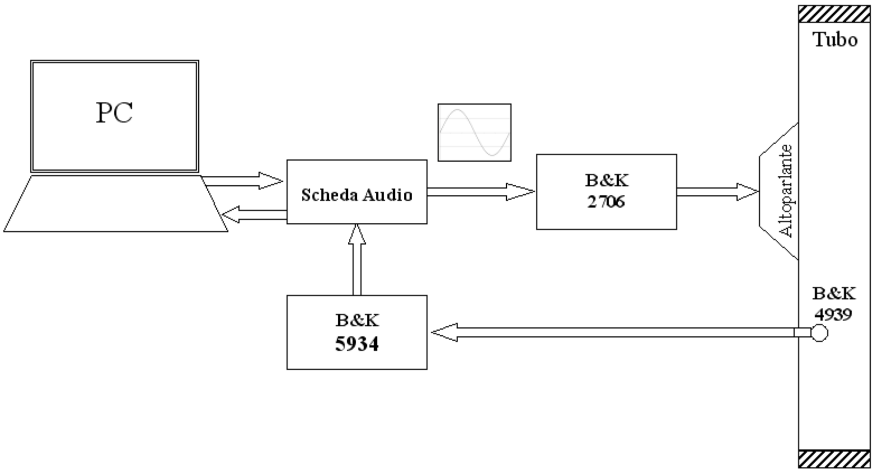

Figure 4. The measurement chain.

4 EXPERIMENTAL RESULTS

4.1 Optimisation of the methodology

The apparatus was investigated to determine the optimal configuration as a combination of the tube length and of the microphone positions. First of all the microphone positions were numbered as reported in Fig. 5 and three different tube lengths were selected according to the positions of the sample holders respectively at 1,16m, 1,30m and 1,39m.
Some problems were reported in the energy decays at lower frequencies for the shorter tube length. In that case the decays showed a double slope. This problem was solved by extending the tube length to the maximum value. With this configuration the double slope was shifted downwards at frequencies out of the range of interest.

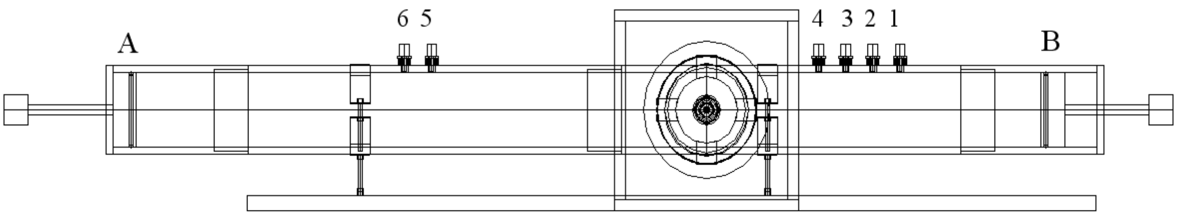

Figure 5. Numbering of the microphone positions.

Once the tube was set at the maximal length a set of measures of the reverberation time was done without any sound absorbing materials in the sample holders A and B. The results are reported in Fig. 6 and show a good consistency even if the values are not the same as theoretically predicted. Of course the derivation that led us to equation (7) did not take into consideration that the tube has its proper modes according to its length. Their frequency can be calculated and their density increases with frequency. The modes strongly characterise the response of the system but, given a suitable S/N ratio at each position for each frequency band, their impact on the calculation on the reverberation time can be limited. This is what is demonstrated by the results in Fig. 6.

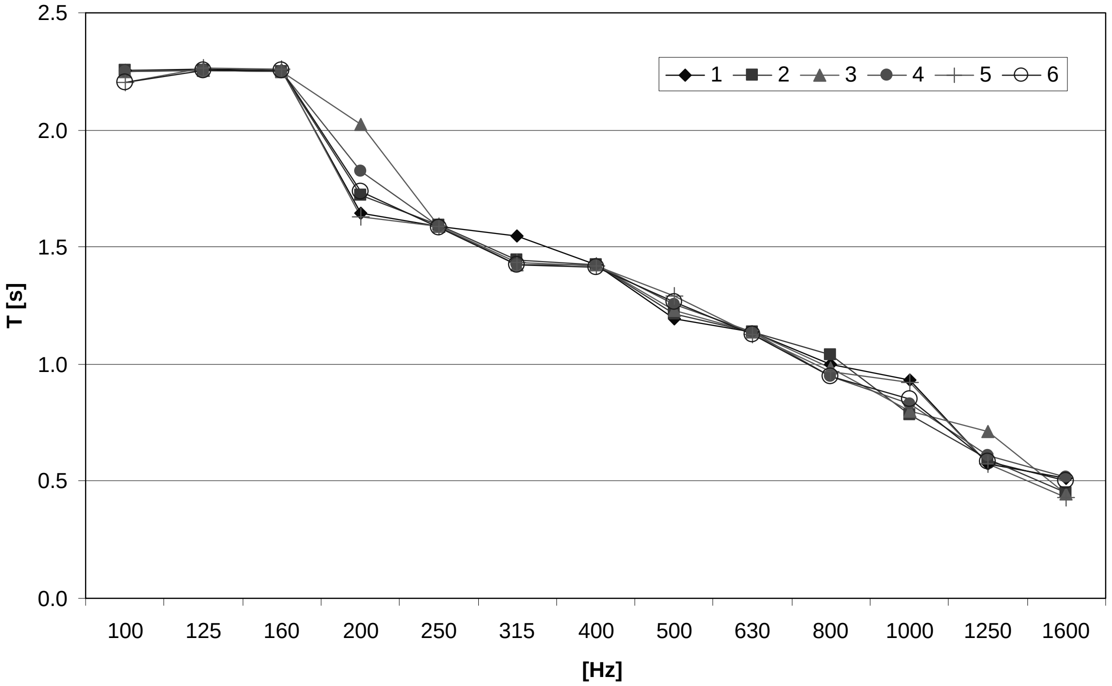

Figure 6. Results of T_{20} within the tube at different positions for tube length at 1,39 m.

As a result of the measures in Fig. 6 the position 6 was adopted for the further steps of the research. Then it was decided to put the sound absorbing material in A and leave the B holder free of test samples. After this part of the study the procedure for obtaining the absorption coefficient was established as follows: measurement of the reverberation time for the empty tube, measurement of the reverberation time with the test sample in A and implementation of equation (7). In order to go through the last step a sound absorbing value for the empty tube is calculated by the same equation (7) by setting α_L equal to α_R.

4.2 Repeatibility

A further step in the development of the method was the testing of the repeatibility. To this aim the absorption coefficients of 15 test samples consisting of porous and fibrous materials of various thickness and densities were measured. Table 2 reports the gross characteristics of the samples.

N°	*Name*	*Density [Kg/m^3]*	*Thickness [m]*	*Typology*
C1	Melammina	10	0,030	porous
C2	Melammina	10	0,060	porous
C3	Polyuretan	30	0,025	porous
C4	Polyuretan	-	0,020	porous
C5	Polyuretan	-	0,030	porous
C6	Polyester fibres-type1	67,5	0,012	fibrous
C7	Polyester fibres-type1	66	0,050	fibrous
C8	Polyester fibres-type1	38,5	0,060	fibrous
C9	Polyester fibres-type1	16,5	0,030	fibrous
C10	Polyester fibres-type1	50	0,010	fibrous
C11	Polyester fibres-type2	50	0,010	fibrous
C12	Polyester fibres-type2	40	0,030	fibrous
C13	Glasswool	21	0,045	fibrous
C14	Rockwool	55	0,050	fibrous
C15	Feltro	73	0,015	fibrous

Table 2. Classification and characteristics of the samples used to test the repeatibility.

Firstly the reverberation time in the empty tube was measured and then, as outlined above, the sound absorption coefficient of a test sample was calculated. For each material a set of measurements was done and the reverberation time was obtained as a mean of the measured values with the proper standard deviation. From the reverberation times the calculations produced for each test sample an average α with its related standard deviation σ. In Table 3 such standard deviations have been averaged and are showed according to their frequency band.

[Hz]	100	125	160	200	250	315	400	500	630	800	1000	1250	1600
$\overline{\sigma}$ *10^{-4}	2	2	2	2	4	13	13	10	5	10	13	5	4

Table 3. Averaged standard deviations.

It can be concluded that the repeatibility of the measurements of sound absorption with the new method based on the reverberation time is quite satisfactory since the average values of the standard deviations is as small as 6*10^{-4}.

4.3 Validation

Moreover also the consistency of the new method with the previous method based on the transfer function (TF) was investigated. The same group of 15 materials as above was employed. For each method a curve of results can be traced. In the new method the points are the α values for the measured 1/3 oct. bands whereas in the TF method the band values must be extracted from the continuous curve produced by FFT analysis. In this case they nominally represent only the centre frequency with no contribution of the rest of the band. This gives and indicative character to the comparison even if a central value is generally well representing the whole band behaviour.
Anyway the comparison took into consideration the average differences between the two curves for the group of 15 materials and the results are reported in Table 4.

[Hz]	100	125	160	200	250	315	400	500	630	800	1000	1250	1600
$\Delta\alpha$*10^{-2}	4.2	2.7	1.4	2.3	2.3	5.2	5.3	5.4	3.6	6.8	3.5	4.6	3.8

Table 4. Average discrepancy from the two methods for the 15 samples.

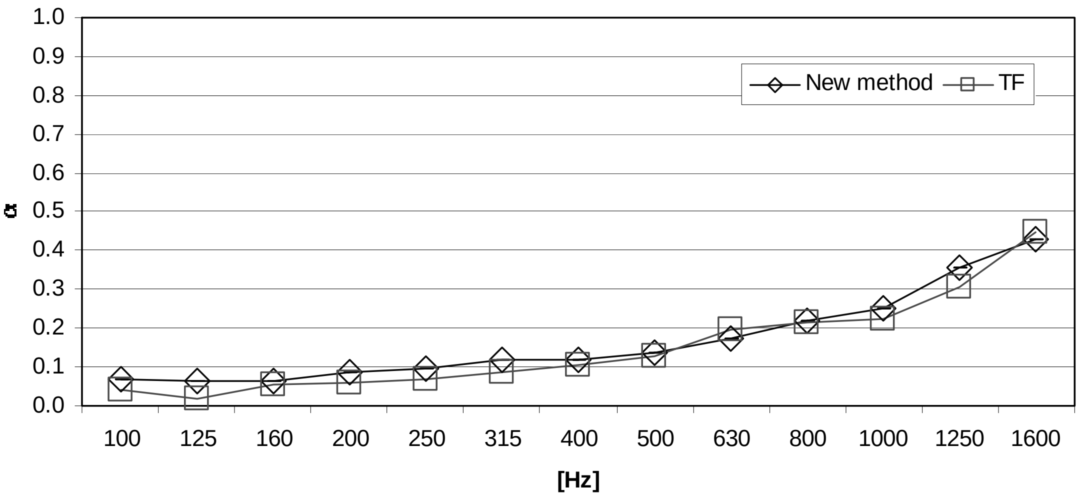

Figure 7. Sound absorption coefficient of sample 3 (Polyuretan thickness: 25mm, density: 30Kg/m3).

Taking the average of the values in Table 4 gives $3.9*10^{-2}$ which is a satisfactory result when compared with the usual accuracy expected from sound absorption data. Furthermore it is to be noted that no anomalies or discontinuities were reported in the curves even if the set of materials included both light and heavy sound absorbing layers. Two typical results of the comparisons are showed in Figs 7 and 8.

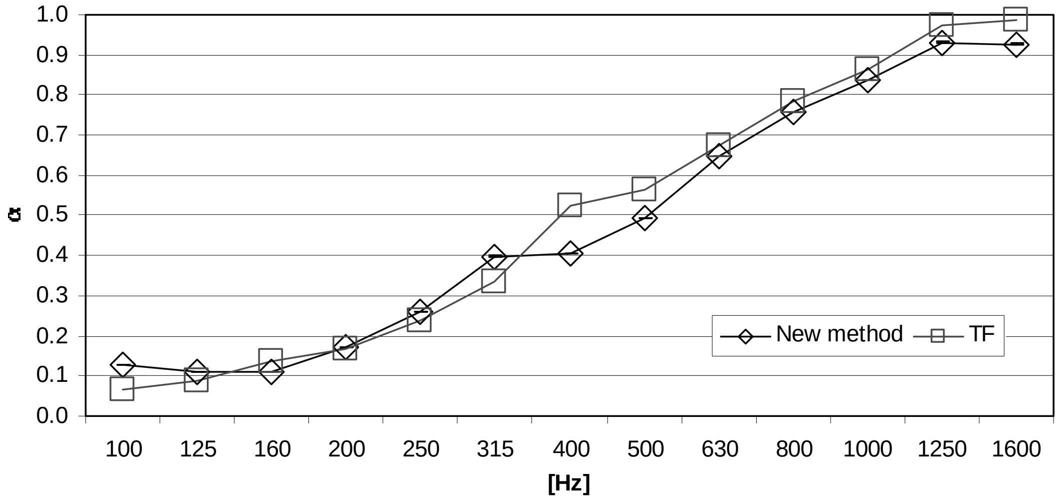

Figure 8: Sound absorption coefficient of sample 8 (Polyester fibers-type1, thickness: 60mm, density: 38.5Kg/m3).

5 CONCLUDING REMARKS

The present work reported the theoretical and experimental development of a new methodology for making the measurements of sound absorption for normal incidence in a tube. Differently from the most common method based on the transfer function, the new method uses one microphone only and is based on the measurement of the reverberation time. A new apparatus was built consisting in a tube where the test samples can be placed at both ends. Moreover the set-up was tested regarding the repeatibility and the consistency with the TF methodology.

In both cases the results are quite satisfactory and the impact of the proper modes of the tube, which give a typical coloration to the sound inside it, can be limited by a sufficient S/N ratio in each band.

This methodology is still under development and a 1/12 octave band analysis is being tested. It is also programmed to develop the measurement of the complex reflection coefficient to obtain the specific impedance of the material.

6 REFERENCES

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5. M. Schröeder, *"New method of measuring reverberation time"*, J. Acoust. Soc. Am., 37, 409-412 (1965).