TECHNICAL UNIVERSITY OF CLUJ-NAPOCA

ACTA TECHNICA NAPOCENSIS

Series: Applied Mathematics and Mechanics
Vol. 57, Issue I, March, 2014

CONTRIBUTIONS ON THE DEVELOPMENT OF A DEVICE FOR MEASURING THE SOUND-ABSORBING COEFICIENT

Alin Cosmin TOT, Mariana ARGHIR

Abstract: *This paper proposes you describe working and the components of a device for the determination of sound-absorbing coefficient published in the previous edition and essential elements in implementing the architecture processed and acoustic signals analysis undertaken*
Key words: *impedance toobe, acoustic signals analysis*

1. INTRODUCTION

This paper aims to describe working and the components of a device for the determination of sound-absorbing coefficient published in the previous edition and essential elements in implementing the architecture processed and acoustic signals analysis undertaken

2. Description of the device

Taking into account the specifics of data ISO 10534-2: 1998 concerning sizing device for determining the coefficient of phono-absorbent, resulted following dimensions listed in table 1 and shown in figures 1 or 2, for the following measuring ranges:

- 40-201Hz range takes place between the first and the last measuring point
- The range 81-735Hz takes place between 1 and 3
- 201-1500Hz Range is between 1 and 3 [1], [2]

The two microphones are placed in the impedance tube so that it is positioned on the inside diameter, to influence how much less acoustic phenomena inside the tube.Întrucît microfoanele utilizate necesită o

Dimensions of the device [1], [2].

X_1	X_2	X_3	X_4	D
0.36	0.445	0.57	0.78	0.12

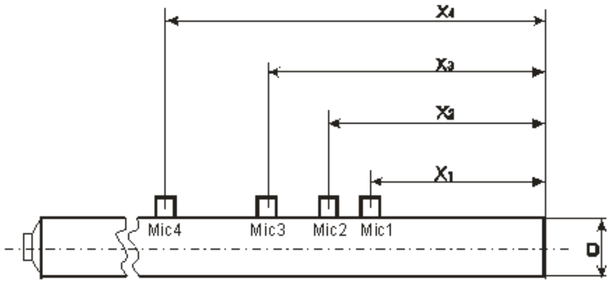

Fig. 1 Positioning of the microphone [1], [2]

Fig. 2 Vedere exterioară a tubului de impedanță

power supply output signal of microfonelor, has mounted an electronic circuit on the wiring test (Phantompower, fig. 3) with the task to ensure the electrical power required fuc□ionări microphones and also to provide alternative form of semalului, which will be taken over by the achiziție board.

Purchasing card illustrated in Figure 4, converts analog signal into a digital signal via a converter with a resolution of 16 bits, thus 16^2 value ranges between negative pressure and the maximum pressure.

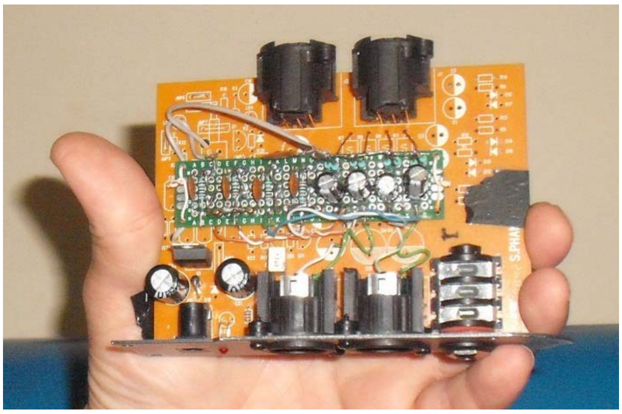

Fig. 3 Phantompower

Fig. 4 Acquisition board [3]

Fig. 5 Audio-frequency amplifier

Fasten the inserted inside the tube impedance is achieved, starting from the fuc□ii generator made in Matlab programming environment tramite digital signals to the sound card is charged with transforming the signal into analog signal.

This in turn sends its semalele by an audio-frequency amplifier (fig. 5) able to deliver a high enough power to put in motion the loudspeaker mounted inside the tube of inpedance.

3. Details of testing and working

Working testing system for determining the sound-absorbing coefficient, to take into account two aspects:

1. Computational testing aggregate to deliver results in areas of both borderline field acoustic frequency, achiziționînd signals in the range of frequency between 40 and 1600Hz on all delivered by the measure, as well as the ability of the system to deliver the values of sound-absorbing coefficient close to 0 with goal aggregate, as well as in the area of maximum 1, carried out with a sample is deemed to have a high coefficient of absorșieso I chose a sample made of sponge 90 mm thick.
2. Taking into account that at the moment the device is realized in a test phase of functionality is necessary to limit the complex phenomena of acoustic waves by changing the type of rapid oscillation generated by the speaker and the period in which the acoustic signal is generated to be as short as possible in order to obtain a more constant value. Thus, the acoustic signal generated it took 0.2 seconds (minimum interval analysis recommended by ISO 10534-2: 1998).

4. Signal processing and data processing

A sound pressure from inside the tube of the impedance is achieved with two microphones that convert sound pressure in the magnitude of the voltage directly proportional to it. Electric (analog) signal is picked up by an analog-to-digital conversion, later reaching into a processing system. Once introduced into the signal processing system is useful for extracting the required sequence analysis. Such cuts, a relevant sample analysis. To minimize errors arising when the signal useful sample drawings, place it on a form of progressive what mitigation starts from the center of the sample taken and ends at ends with value 0 (Figure 6) [4].

Once implemented this procedure on each record retrieved, carry out an analysis FFT (Fast Fourier Transform) (Fig. 7).

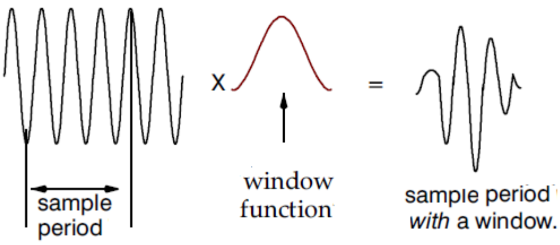

Fig. 6 Application of window [4]

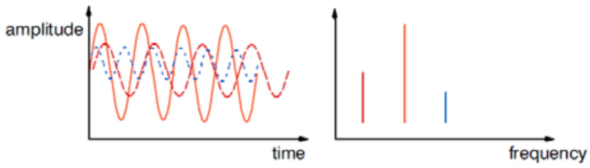

Fig. 7 The FFT signal processing [4]

With the help of Matlab programming environment has generated a signal composed of three elements with a period of 0.2 seconds, of which the first and third period has zero second value period located in the middle of them is made with a sinusoidal oscillation constant with a given frequency, thus a total of 0.6 seconds. The first range with value 0 was inplementat to eliminate any noises occurring in the early trasmisiei the signal generated. The last range with valore 0 was mounted so as to be able to observe residual acoustic oscillations (ECHO). As shown in Figure 8, the raw form of the signal picked up by one of the MIC it is noticed that at the beginning of the corresponding oscillator signal time interval approximately 0.22-0.3 seconds there is an unevenness of stationary waves, and towards the end of the segment oscillator that is the time period between 0.43-0.5 seconds appears again a stationary wave unevenness resulting from the echo effect.

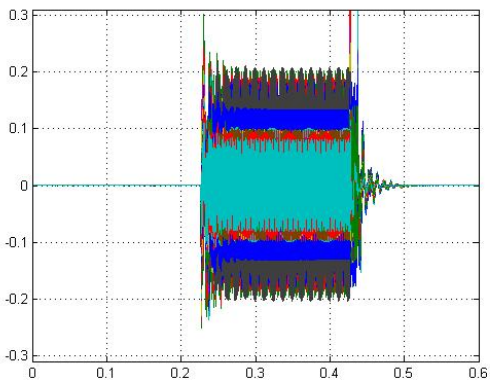

Fig. 8 Raw signals

In conclusion, the figure 8 we can deduce that the relevant determinations, i.e. the area where the stationary waves have a well-defined shape is comprised between 0.3 to 0.43 seconds. Therefore he cut this interval as shown in Figure 9 (signal from one microphone), the totalcatches.

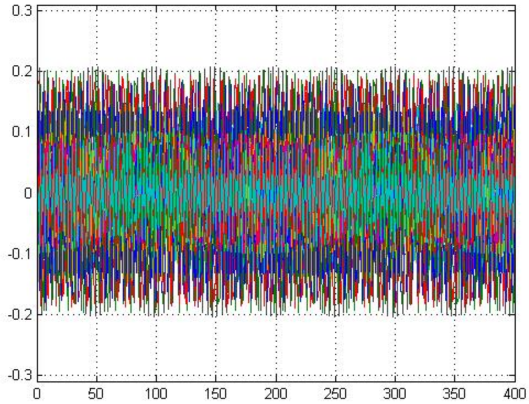

Fig. 9 Useful signals

Once cut the useful analysis, the implementation of other phases of signal processing to eliminate certain differentiated as signals phase that may have to begin and end with value 0 to determination with a better accuracy zero of FFT and focus sections. It has thus been introduced function window which carry out a gradual attenuation of the Center listed first (with 1 value) sample that has been applied and ending with its ends (value resulting in a 0). The result of the application of this analytical sample function picked up by a microphone, which is illustrated in Figure 10.

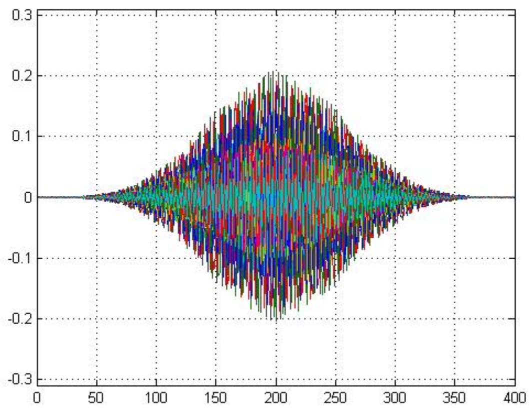

Fig. 10 Application effect of window

The result of Fourier analysis, shown in Figure 11 represents the amplitude of the sound wave in the rapport with its frequency in a specific measuring point located on the impedance tube. Theoretically, if it would mean amplitudes obtained on each point on the axis of the image indicates the frequency, this graph should be a mitigation of the noise amplitude with increasing frequency, and in areas where the measure has been achieved for this graph coincides with node powered by stationary wave form value to be 0.

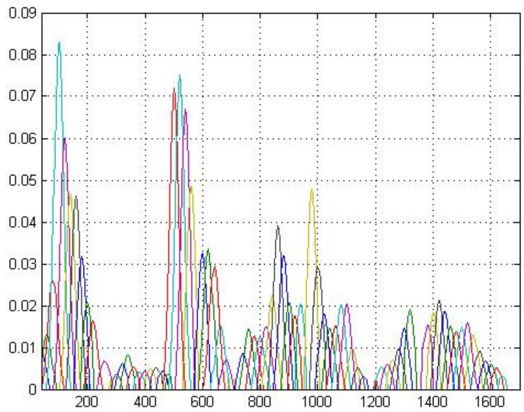

Fig. 11 FFT analysis

The calibration factor, (Hc) is achieved by successive measurements between two measurement points by changing the position of the microphones between measurement points and is determined by the relationship [1] [2] [4].

$$H_c = (H_{12}^{I}/H_{12}^{II})^{0.5} \qquad (1)$$

H_{12}^{I} – the first sense of measure
H_{12}^{II} - data collection in reverse

Figure 11 represents the calibration factor achieved by reversing the microphones between points of measurement.

Calibration procedure is carried out through the implementation of stocking sizes Hc in the transfer function thus H12 [1] [2] [4].

$$H_{12} = \frac{\tilde{H}_{12}}{H_c} \qquad (2)$$

Where, (H_12) transfer function that requires calibration

H12 transfer function represented in Figure 12 is effected by the ratio of the measured sound pressure between two measurement points.

Reflection coefficient shown in Figure 13, is accomplished with the following relationship [1] [2] [4].

$$r = \frac{H_{12} - H_i}{H_R - H_{12}} e^{2jk_0 x_1} \qquad (3)$$

r – reflectance factor
H_{12} – the transfer function
H_i – the transfer function of the incident wave
H_r - the transfer function of the reflected wave
k_0 – wave number
x_1 – the distance between the sample and the last microphone

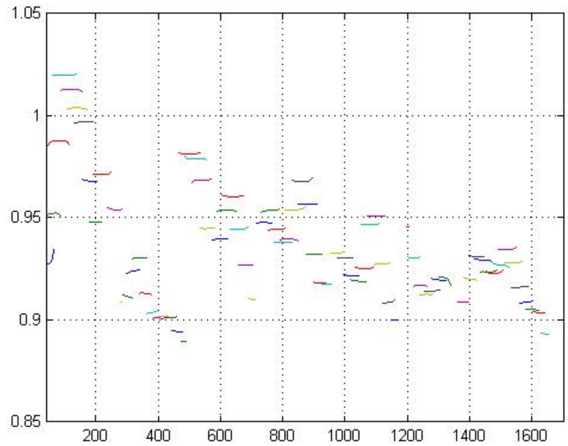

Fig. 11 The calibration factor H_c

The absorption coefficient shown in Figure 14 (represented only one longing for the measuring range), has been achieved through the implementation of the equation below.

$$\alpha = 1 - |r|^2 \qquad (4)$$

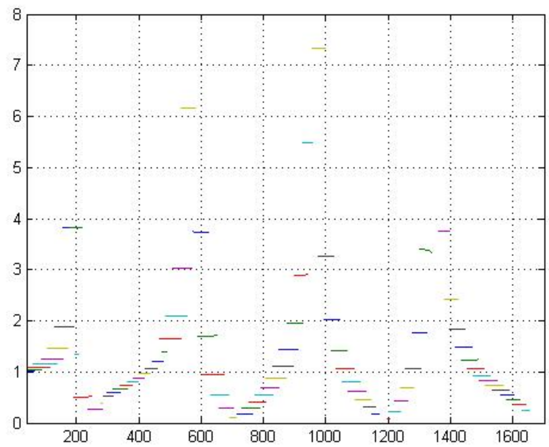

Fig. 12 Funcția de transfer H_{12}

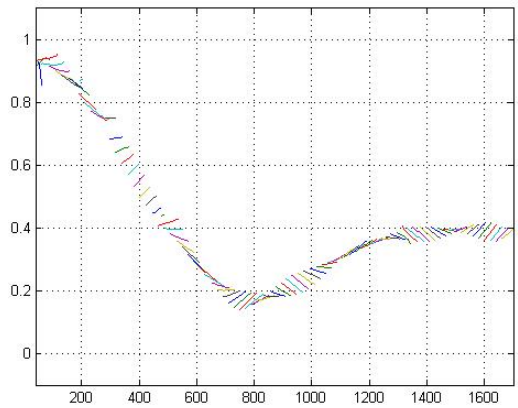

Fig. 13 Coeficientul de reflexie

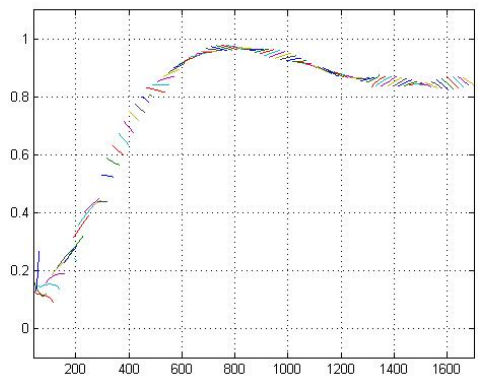

Fig. 14 Coeficientul de absorbţie

Whereas the absorption coefficient shown in Figure 14 (accounted Miss for one interval of measurement) is performed for several samples taken, has been achieved on their mediation resulting graph shown in Figure 15 and 16 for the three ranges of results as defined by the measuring points located on the impedance tube collected in the range 40-1500 Hz. So the three graphs in Figure 15 and 16 are:

1. The blue line represents the absorption coefficient achieved between points x 1 and x 2 of the impedance tube. In both cases, the results tend to be more relevant although satisfies ISO 10534-2: 1998, the measuring range has been exceeded.
2. the measuring range defined by the points x 1 and x 3 represent the green line as shown in figures 15 and 16, coinciding with the chart described above (made between points x 1 and x 2) in the frequency range 40-1000-1500Hz and 700Hz, although these measurement points were intended frequency range 81-735Hz.

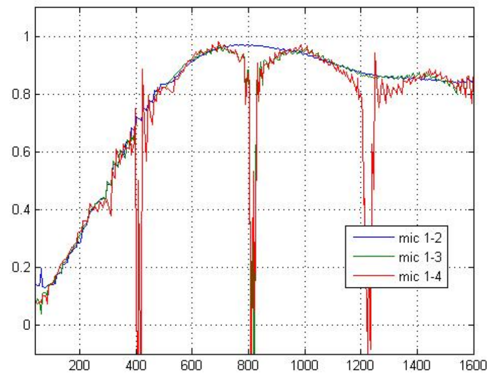

Fig. 15 Mediated absorption coefficient for the 3 measured in the range of 40-1500 Hz frequency, carried out with sample

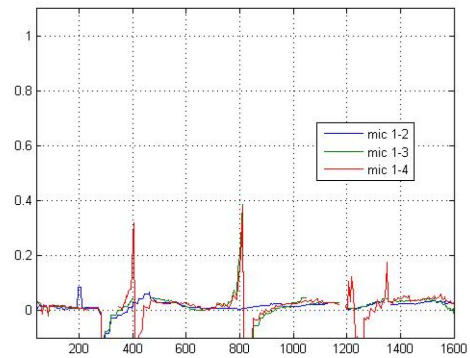

Fig. 16 Mediated absorption coefficient for the 3 measured in the range of 40-1500 Hz frequency, without evidence

3. Drawn with red figures 15 and 16 is the result of the determination of the absorption coefficient taken between measurement points x 1 and x 4. As shown, the result of this determination has a slightly different face shape of the two charts above but coincides in the range of 40-400 Hz frequency, although that position was designed to measure the range 40-201Hz.

5. CONCLUZII

1. As shown in figures 15 and 16 the result graph of absorption between measuring points x 1 and x 2, has the same value as the other two trend graphs results without major disturbance measurements. Thus we conclude that the determination of the absorption coefficient can be achieved between

points x 1 and x 2 on the whole range of frequency 40-1500Hz.

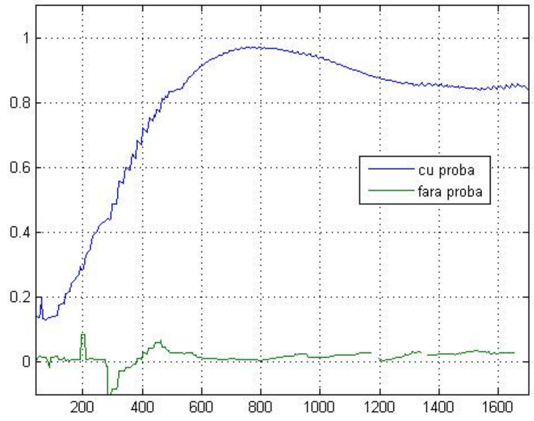

Fig.17 Graph of absorption between measuring points x 1 and x 2, with and without sample

2. As can be seen in Figure 17, although the equations for determining the dimensional relationships between the microphones indicated by ISO 10534-2: 1998 places the range x 1, x 2 as being valid in the range of 201-1500Hz frequency, experimental results show that this period may be exceeded.
3. The mediation results obtained at the end of the process of the warm, is a viable alternative for determining the results because this solution does not prejudice in any way the result of the relationship of determination of the reflectance factor, which allows a deviation of lint result of microphones or pickup system of acoustic signals as long as this deviation is constant.

6. REFERENCES

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CONTRIBUŢII PRIVIND REALIZAREA UNUI DISPOZITIV DE DETERMINARE A COEFICIENTULUI FONOABSORBANT

Rezumat: *Această lucrare îţi propune descrie funcţionalitatea şi componentele unui dispozitiv de determinare a coeficientului fonoabsorbant publicat intr-o ediţie precedentă precum şi elemente esenţiale in implementarea arhitecturii de procesate şi analiză a semnalelor acustice preluate*

Alin Cosmin TOT, PhD Student, Dept.: Mechanical Systems Engineering, Technical University of Cluj-Napoca**,** e-mail: tot.alin.cosmin@gmail.com **,** Romania

Mariana ARGHIR, Prof. Dr. Eng., PhD Student, Dept.: Mechanical Systems Engineering, Technical University of Cluj-Napoca**,** e-mail: Mariana.Arghir@mep.utcluj.ro**,** Tel.: (+)0264. 401.657, Romania