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1MAE504 ACOUSTICS

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Practical sessions report

TP1: Material Characterization (Impedance Tube)

Porous materials consist of rigid solid phases with channels or pockets dispersed throughout. These channels allow airborne waves to propagate through the materials. The narrow sizes of these channels lead to high levels of visco-thermal losses because of the extended contact between the solid and fluid phases. These losses can lead to highly effective absorption of acoustic waves. In the case of aircraft engine noise, acoustic liners are a more beneficial choice. Based on a resonator principle, nacelles are typically lined with a perforated plate covering a series of cavities. Liners function mainly on viscous losses at the perforations as opposed to utilizing thermal losses as well. The size of the cavities is used to control the resonant, and thus attenuated, frequencies of the liners.

The goal of this experiment is to investigate different acoustic materials, and using the knowledge of the main mechanisms, characterize them in terms of their useful ranges and possible practical uses.

Experimental setup

Among all the different measurement techniques that are aiming to quantify the performance of acoustics insulation, the impedance tube method stands out, because of its simplicity, along with short test times and small sample sizes.

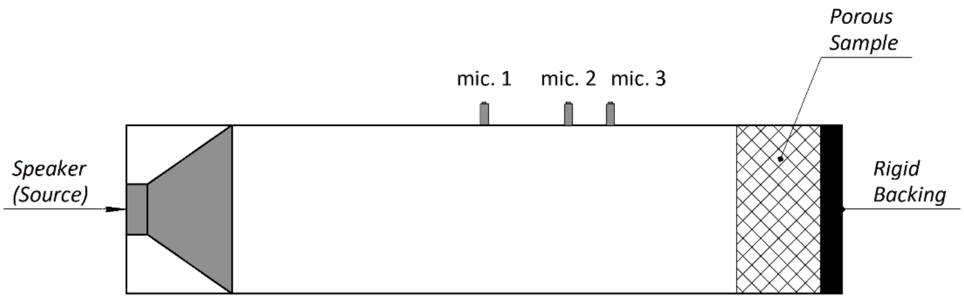

Figure 1 : Impedance Tube

Figure 1 represents the scheme of the experimental test bench. It consists of the:

- The sound source (speaker) that is mounted into one end of the impedance tube;
- 3 microphones, that are needed to detect pressure variations. This number of microphones is used to increase the accuracy of measurements;
- The sample, which acoustics characteristics we want to measure.

The loudspeaker produces random sound waves that cover a wide range of frequencies. These waves travel through the tube as plane waves, reach the sample, and reflect back.

Measurements

Samples to be measured that have been made into corresponding shapes are selected. The measurement parameters, such as sound wave frequency and sound pressure level, are set according to the experimental requirements. Firstly, the acoustic field in the empty impedance tube is measured when the material sample is not installed, so as to compare the subsequent measurements. Then, different samples are placed inside the impedance tube in turn in different measurements. The sound field is measured respectively to observe the performance of different configurations of sound absorbing materials.

Analysis

In general, a good liner material should have a high absorption coefficient over a wide range of frequencies. This means that it will effectively reduce the amount of noise transmitted through the liner. A graph with absorption coefficient $\alpha(\omega)$ as a function of frequency is analyzed, as the shape of the graph depends on the acoustic properties of the material tested. (See Figure 2) A rigid plate obviously does not absorb sound in the same way as a porous material, but instead reflects. Therefore, it has a low absorption coefficient across a wide range of frequencies. Minor fluctuations at certain frequencies are due to limitations of the setup or resonance as mentioned in the above sections.

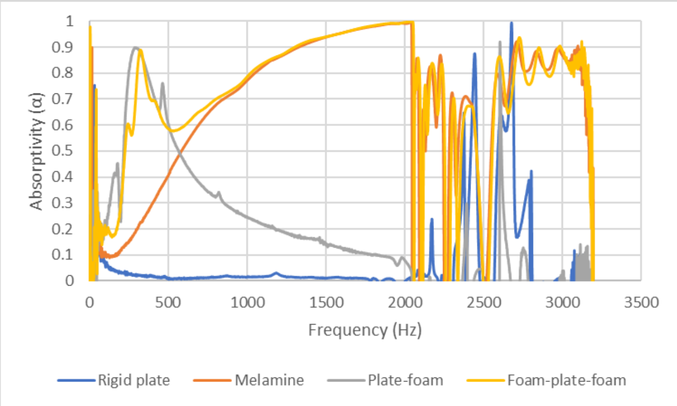

Figure 2 : Absorptivity spectrum for the rigid plate, melamine foam, and combination

The melamine foam can be seen in the graph, has a relatively high absorption coefficient at higher frequencies. The same foam with a combination of rigid-plate performed better at lower frequencies. It is because the combination acts as a mass-spring system resulting in a coupling effect, in which the melamine foam acts as a spring, and the rigid plate acts as a mass. While the rigid plate provides a barrier to sound transmission, the foam absorbs and dissipates sound energy as it compresses and expands in response to the sound waves. This creates a resonance effect that results in higher absorption at low frequencies relatively. As mentioned above, a good liner should be effective over a wide range of frequencies. An absorption coefficient of 0.5 is set as a minimum requirement over a wide range as this can be achieved now at frequencies 500-600 Hz and higher at other frequencies with different s respectively.

Considering the results above, a combination of “Foam-Plate-Foam” could be one possible configuration to achieve the goal. From the experiment results, the shape of the Alpha vs frequency graph in red shows the configuration performs well at both low and high frequencies. The initial foam layer absorbs the higher-frequency waves and the low-frequency waves that are further transmitted are damped by the plate-foam combined (coupling effect). A melamine foam with one surface that is encountered by the incident wave is cut into wedge shape as in the anechoic room to mimic the same. It did not perform better (see Figure 3), the absorption coefficient at 2000 Hz is not 1 as that in case of plane foam. This is because the wedges are designed for a certain cutoff frequency range in the anechoic room and mimicking the same doesn’t mean that it should perform better at all the frequencies. Another configuration of rigid plate, with one of the surface custom cavity designs, and a melamine foam is studied. This combination performed like the Foam-plate-Foam configuration.

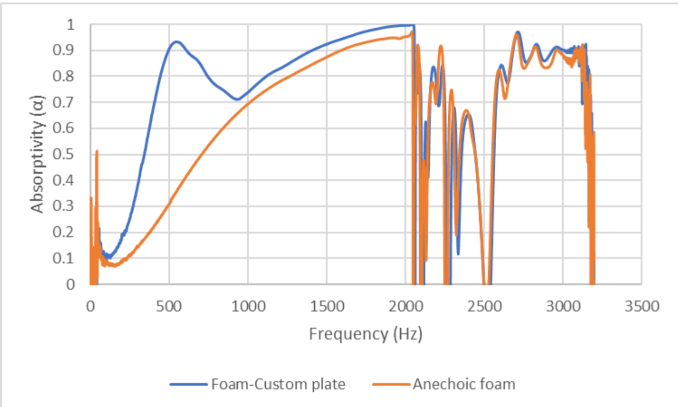

Figure 3 : Absorptivity spectrum of materials with different morphology

Taking this as an example of the acoustic liner, it is understood that instead of using different materials, the shape of the cavity can be modified to fit in the intended frequency band. One application of this can be in aircraft nacelle for reducing noise. As future engines will have thinner nacelles, absorbing low frequencies will be a problem as the total height of the cavity decreases. The shape of the cavity can be modified (see Figure 4) so that the effective height of the cavity doesn't change. The ratio of perforation height and cavity height, i.e., (h/H) is proportional to wavelength (λ). For larger wavelength applications the height of the perforation inlet can be modified as shown in the figure below.

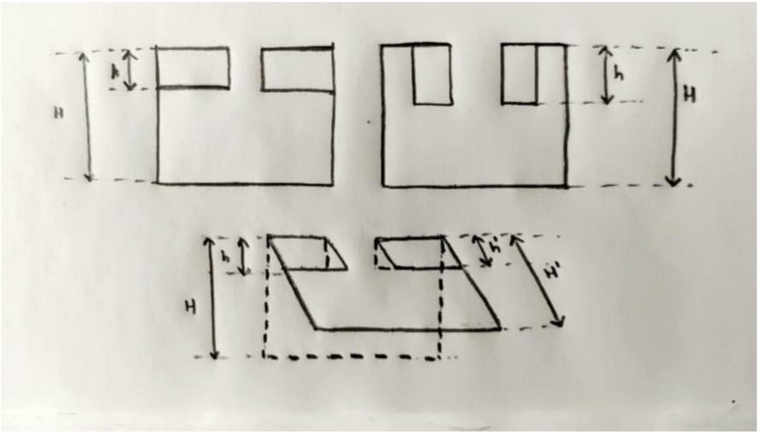

Figure 4 : Schematic of cavity liner morphologies

TP2: Material Characterization (Reverberant Room)

The aim is to determine the random sound absorption coefficient of materials. A reverberation room is a specialized testing facility designed to evaluate the acoustic characteristics of a given space or a material. The room is an enclosed space with highly reflective; non-parallel walls designed with a purpose to provide controlled environment (a diffused field) for measuring the reverberation time, which is the time it takes for sound to decay to 60dB after a sound source has been stopped emitting sound. As Sabine's theory suggests, the sound waves bounce of the surface of the room and continues to reflect until they lose their energy through absorption and the acoustic characteristics of a material can be determined using the following formulae:

Reverberation time

$$T = 0.161\frac{V}{A}$$

Acoustic absorption coefficient

$$\alpha(\omega) = \frac{P_{abs}(\omega)}{P_{inc}(\omega)}$$

Acoustic absorption coefficient of a material

$$\alpha_{mat} = 0.161V\left[\frac{1}{S_0T_0} + \frac{1}{S_{mat}}\left(\frac{1}{T_1} - \frac{1}{T_0}\right)\right]$$

<u>Experimental setup</u>

The experiment was conducted in a pentagon-shaped reverberation room with non-parallel walls coated with hard paint to facilitate reflection and prevent acoustic modes excitation. This ensured a diffused field was set up for accurate measurements. Additionally, a mini-reverberation room was prepared for measurement, which was a 10 times scaled-down version of the original room. Both the full-scale and mini-reverberation rooms had the same acquisition chain, which consisted of two pairs of transducers. The first pair of transducers acted as the source, while the second pair, two omnidirectional microphones, detected and converted the pressure into an electrical signal. The signal was then passed to the amplifier and data processing station to obtain the required intensity measurements. The materials whose absorption was measured were made up of foam.

Measurements

A total of six rounds of measurements were performed: four performed in the miniature reverberant chamber, and two performed in the large chamber. The first measurement for each chamber was performed to achieve a baseline of the standard behaviour of the chambers. For the miniature chamber, the first material (mat1) tested was a soft yellow foam with a 3D pattern of bumps across its surface. Mat1 measured 0.2m x 0.124m. The second material (mat2) tested was a section of melamine foam measuring 0.392m x 0.138m. The final test for the miniature chamber tested the combination of mat1 and mat2 arrayed next to each other in the chamber. The results for the miniature tests can be seen below in Figure 5. For the large chamber a large white foam section, measuring 2.54m x 1.305m, was tested. The results for the large chamber can be seen in Figure 5.

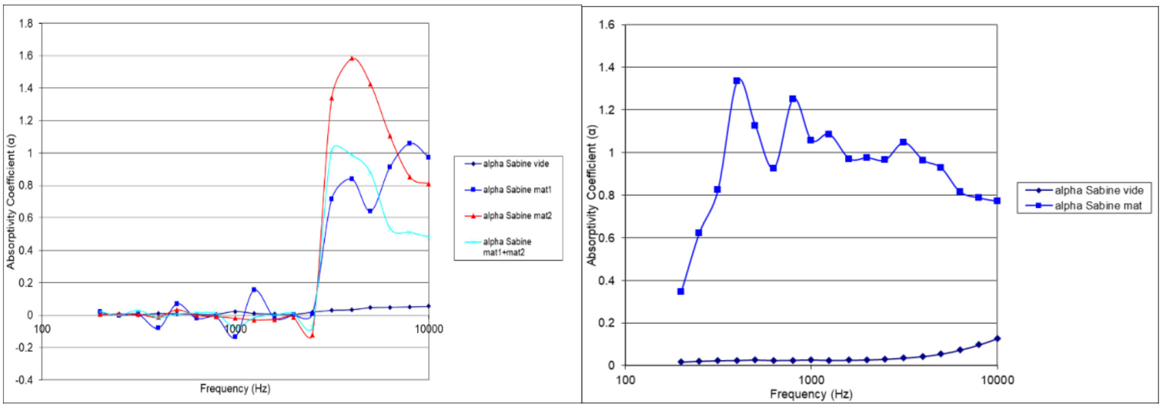

Figure 5 : Reverberant room results from Sabine theory for miniature room (left) and large room (right)

Analysis

Despite all its merits, Sabine's theory could exceed the value 1 of the absorption coefficient. It happens because Sabine assumes that there is only one energy value all over the room, which is not really physical, since this requires an infinitely fast speed for an information about absorption. For this reason, it is better to use the Eyring method, that will give us values in the correct range from 0 to 1.

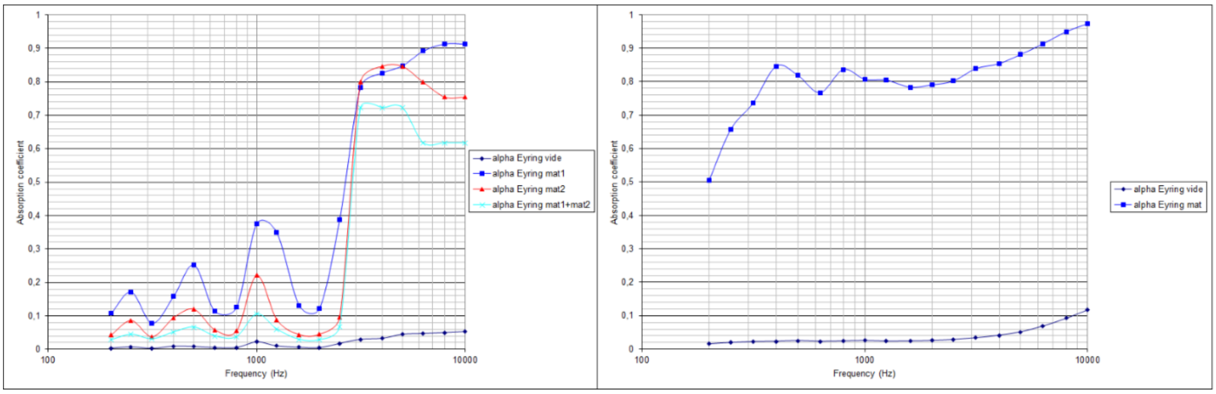

Figure 6 : Reverberant room results by Eyring for the miniature room (left) and the large room (right)

Figure 6 represents an absorption coefficient obtained with from the Eyring method. As expected, this approach provides us more appropriate results compared to Sabine's theory. By analysing the obtained results, firstly the influence of a cut-off frequencies is clearly seen. For the big chamber (Figure 6- right) this value equals to 200 Hz, so this phenomenon does not affect our measurements. In a contrary, the small reverberation chamber (Figure 6- left) is around 10 times smaller than the big one and so its cut-off frequency equals to 2000 Hz. Because of that, there is no diffuse field in the chamber, and we can clearly identify resonance mode.

As for the absorption coefficient, its trend is to increase along with the frequency for both chambers. Generally, porous materials are good to damp high frequencies, however, several interesting events were observed. For the small room, both materials had identical before the frequency of 5000 Hz, where the pyramidal absorber foam presented a sharp rise in absorption coefficient. It happens because of the presence of a pyramid-like structure 5-6 mm high. This height corresponds exactly to frequencies above 5000 Hz, allowing sound waves to interact with this shape. Also, it's worth mentioning that the combination of these 2 materials in the room, provided worse performance than these materials separately.

TP3: Source Characterization (Anechoic Room)

Anechoic chambers are designed to minimize background noise, creating an ideal environment for measuring acoustic wave intensity. This allows for the study of acoustic wave propagation and interaction assuming the 'free field' conditions. The aim of this experiment is to determine the constraints of sound intensity probes when they are not aligned with the direction of wave propagation. Additionally, the experiment aims to describe the behavior of the pressure field between two sources for both dependent and independent cases, with the goal of reducing the intensity of acoustic waves.

Experimental setup

The study under consideration was conducted in an anechoic chamber specially designed to prevent sound reflections from any surface. Typically covered with foam wedges and fiberglass, which traps sound waves and prevents them from bouncing back and interfering with the original sound. The cut-off frequency is 90 Hz. The experimental setup inside this room comprised of two controlled speakers and a motor-fan as acoustic sources. A movable sound intensity probe of the 'p-p' type which consists of a pair of microphones is used to assess the pressure gradient and velocity of sound waves. The readings are on the screen during the experiment, but it is important to note that the use of two microphones is required to estimate the gradient accurately.

Figure 7 : Anechoic chamber

From the Euler equation:

$$u_r' = -\frac{1}{\rho_0}\int_{-\infty}^{t}\frac{\partial p}{\partial r}d\tau \cong -\frac{1}{\rho_0}\int_{-\infty}^{t}\frac{p_2' - p_1'}{\Delta r}d\tau$$

The active acoustic intensity, $I = \langle p'u' \rangle$, the radial active intensity at the center of the "p-p" probe is given by:

$$I_r \cong -\frac{1}{2\rho_0 \Delta r}\langle (p_2' + p_1')\int_{-\infty}^{t}(p_2' - p_1')d\tau \rangle$$

Measurements

The measurements primarily focus on the sound pressure level and sound intensity level.

$$L_p = 20log_{10}\left(\frac{P}{P_{ref}}\right)$$

$$L_I = 10log_{10}\left(\frac{I}{I_{ref}}\right)$$

where $P_{ref} = 20e^{-6}\ Pa$ and $I_{ref} = 1e^{-12}\ W/m^2$. First, the acoustic information of the anechoic chamber with no sound source turned on was measured. Second, the noise performance of two loudspeakers and a broadband noise fan were measured independently in turn, and their noise performance when interacting with each other was measured as well. When they are interacting with each other, the positions of the microphones are moved correspondingly so as to be set aligned with the direction of sound propagation and at the centre of the two sound sources.

Table 1 : Experimental results of the sound pressure level in dB (mean value of microphone 1 and 2) and sound intensity level

Active Sources	Sound Pressure Level (dB)	Sound Intensity Level (dB)
Speaker 1	47.58	47.13
Speaker 2	47.08	47.19
Fan	46.70	46.57
Speaker 1 & Speaker 2	52.86	34.41
Speaker 1 & Fan	49.92	42.12

Analysis

Each of the individual sources tested provided roughly equivalent sound pressure levels at the frequency of interest, which in our case was 800Hz, as can be seen above in Table 1. Thus, the expected result from constructive interference of two sources would be a decibel change of 6dB due to the logarithmic nature of the scale. However, when tested together only the two loudspeaker sources provide the expected an approximate 6dB change. For the case of the broadband fan and a loudspeaker, the decibel change is less than expected. The two loudspeakers are controlled by the same current source and thus receive the same input signal. As a result, the signals each respective loudspeaker produces are correlated with the other loudspeaker. This translates to a phase agreement which allows for the assumption that constructive interference occurs at the same location of the signal. The pressure amplitude of the correlated signals is identical, thus the expected result of 6dB was achieved. For the case of the fan, it is independently powered, and the signal it produces is uncontrolled. In this case, there is no correlation between the loudspeaker signal, and the broadband noise at the designated frequency, and thus a phase difference between the two sources. The difference in phases causes the variation from the expected 6dB result, as the sum of the pressure amplitudes varies with respect to time, given the phase difference.