BUILDING A MODIFIED IMPEDANCE TUBE FOR MEASUREMENT OF SOUND TRANSMISSION LOSS AND ABSORPTION COEFFICIENTS OF POLYMER CROSS-LINKED AEROGEL CORE COMPOSITES.

By

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CHAPTER I

INTRODUCTION

Aerogels

Aerogels are mesoporous nanostructured materials with high surface area and their porosity can reach up to 99.0% [1]. Aerogels were first invented in 1931 by Steven S Kistler of the College of the Pacific in Stockton, California and are the lightest known solids on earth, with a minimum bulk density of 1.0 mg/cm^3 reported in the 2008 Guinness world records. Fig. 1 shows a picture of silica aerogel with a mass of approximately 1 g.

Figure 1 Traditional Silica Aerogel.

In aerogels, nanoparticles are connected together to form loosely connected network, leaving very high fraction of empty space (or porosity). Due to this high porosity, aerogles have exceptional properties such as low density, low dielectric constant, low thermal conductivity and high acoustic damping [2-4]. Traditional silica aerogels, however, have only found limited use as thermal insulation material when it is used as part of a composite, Cerenkov radiation detectors in certain types of nuclear reactors, in NASA's Stardust program for capturing hypervelocity particles in space, and as ultra lightweight thermal insulators (the aerogel had thermal conductivity 0.012 W/m·K in this case) aboard certain short-lived planetary vehicles such as the Sojourner Rover on Mars in 1997 in limited situations, such as the thermal insulation material for electronic box on board of all Mars Rovers. The slow commercialization of traditional aerogels is because aerogels have very low strength, they are brittle and hydrophilic (absorbing moisture in the environment, leading to final collapse due to capillary force). In 2002, Dr. Leventis's group at Missouri University of Science and Technology has successfully developed a method to encapsulate the traditional silica aerogel with crosslinker to form crosslinked silica aerogels, a kind of crosslinked aerogels to improve the mechanical strength without sacrificing other unique properties. Fig.2 (A) shows a SEM micrograph of traditional silica aerogel, and Fig. 2(B) shows a SEM micrograph of a crosslinked silica aerogel [5].

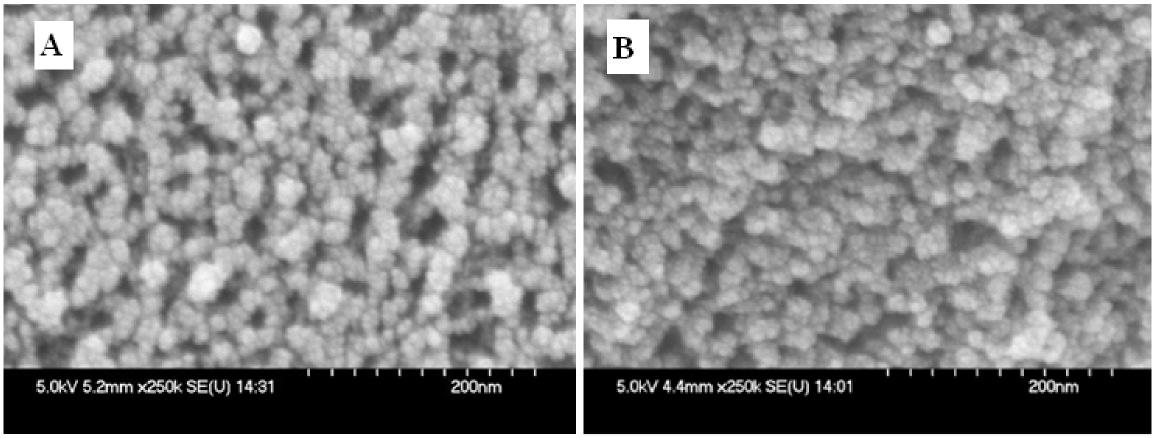

Figure 2 SEM image of (a) Aerogel (b) Crosslinked Aerogel

The nanomorphology of crosslinked silica aerogel is very similar to the traditional silica aerogel (Fig. 2A). Figure 3 shows the flexural strength as tested three-point bending.

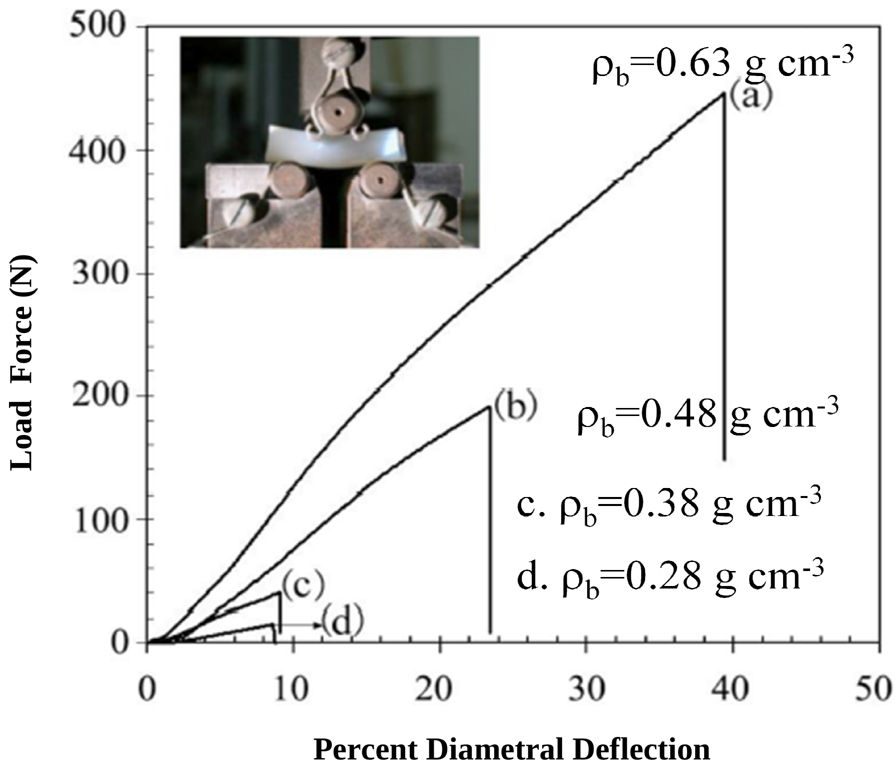

Figure 3 Typical load force versus diametral deflection curves for different crosslinked silica aerogel.

It is seen that there is a dramatic (4000 %) increase in the specific rupture strength, and a five-fold increase in specific Young's modulus due to cross-linking. Since the porous nanostructure is maintained in crosslinked silica aerogel, thus the properties possessed by traditional silica aerogels are maintained. The average thermal conductivity of crosslinked silica aerogel ranges between 0.03 to 0.04 W/m·K, which compares favorably with 0.02 W/m·K for monolithic silica aerogel. The X-aerogels have very high acoustic damping property similar to traditional aerogels.

The aerogel material used in this thesis is labeled as MP4-T45 ('P4' and T45 stand for 4 g of Pluronic P123 and 0.45g of TMB respectively) through this report. In this procedure, 4.0 g of Pluronic P123 was dissolved in 12 g of 1.0 M aqueous solution of nitric acid under magnetic stirring for 8 hours. Under vigorous stirring, 0.45 g of TMB was added to the solution for 30 min, samples were cooled to 0°C and 5.15 g of TMOS was added to the solution after another 30 min. After stirring for 10 min, the solutions were poured into a mold and a piece of Nomex Honeycomb placed inside the solutions. The mold was then covered and sealed with PTFE tape and kept inside the oven at 60°C for gelation. The sample was being monitored every 10 to 15 min until gelation time was established and the sample were aged at 60°C for 5 times the gelation time which was about 12 hours. The honeycomb embedded with MP4-T45 was removed from the mold directly into ethanol with the amount of 4 times the volume of the gels. Ethanol was changed 2 times at 8 hour intervals. The sample went through Soxhlet extraction using CH_3CN as solvent for 2 days to remove P123. Acetone was used to wash the sample 4 times with 8 h intervals. After washing with acetone, the sample was placed in acetone solutions of the di-isocyanate (Desmodur N3200). The volume of the solution was 4 times greater than volume of the sample with the ratio between acetone and N3200 of 12 g to 88 ml. After the Desmodur N3200 solution had reached the equilibration time (1 day) at room temperature, the sample together with the Desmodur N3200 solution were heated at 55°C for 3 days. Acetone was used again to wash the sample 4 times with 8 h intervals to remove the unreacted di-isocyanate. Finally, the sample was dried with Pentane.

Definition of Sound Transmission Loss

The physics of sound waves and interaction of sound waves with structures has been an extensive field of research from the past few decades. One of important characteristic property of research interest in this field of study is the sound transmission loss. Sound transmission loss (STL) is the accumulated decrease in the sound intensity of an outward propagating sound pressure wave. As a property of material, i.e., the ability of a material to block sound, STL is defined as the ratio of sound energy transmitted through a material to the sound energy incident upon it. STL is measured by mounting the sample between two reverberating test rooms. A broadband noise is generated in one of the rooms and the sound pressure levels are measured in the two rooms to determine the noise reduction and sound transmission coefficient (τ). The sound transmission coefficient (τ) is defined as the ratio of sound energy transmitted to the sound energy incident upon the material and the sound transmission loss is then simply defined as the sound transmission coefficient expressed in decibels:

$$STL = 10\times\log_{10}(1/\tau) \qquad \text{dB} \tag{1}$$

STL is frequency dependent, increases with the increase in frequency and is usually described (plotted) in 1/3 octave bands.

Problem and Description

Nature of problem

Two distinct kinds of problem arise in the study of sound transmission from one part of a structure (room/chamber) to another part (room/chamber). The first is illustrated by the example of motor or any other electric machinery mounted directly upon the building structure. Due to inevitable imperfections in the bearings and due to the periodic character of the torque exerted to the armature, vibrations are set up in the machine. These are transmitted directly through the structure on which it is mounted and hence conduction through solid structural members to remote parts of the building. These vibrations of the extended surfaces of walls, floors and ceilings, produce waves in air. The second kind of problem is the transmission of acoustic energy between adjacent rooms by way of intervening solid partitions, viz., walls, floors and ceilings. Transmission of voice or violin from one room to another is a typical case. The transmission of footsteps on the floor to the room below would come under the first type of problem.

Mechanism of Transmission in Walls

Let us consider two adjacent rooms/chambers A and B separated by a partition P with a schematic shown in Fig. 4. Sound is produced in the source room/chamber A, the major portion of the sound energy striking the partition will be reflected back into room/chamber A, but a small portion of sound will appear as sound energy in room/chamber B. There are three distinct ways in which the transfer of energy from A to B by intervening partition is affected

(1) If P is perfectly rigid, compressive air waves in A will give rise to similar waves in solid structure P, which in turn generate air waves in B.

(2) If P is a porous structure such as to allow the passage of air through it, the pressure changes due to the sound in A will setup corresponding changes in B by the way of the pore channels in the partition. A part of the energy entering the pore channels will be dissipated by friction, i.e., there will be loss of energy by absorption in transmission through a porous wall [6].

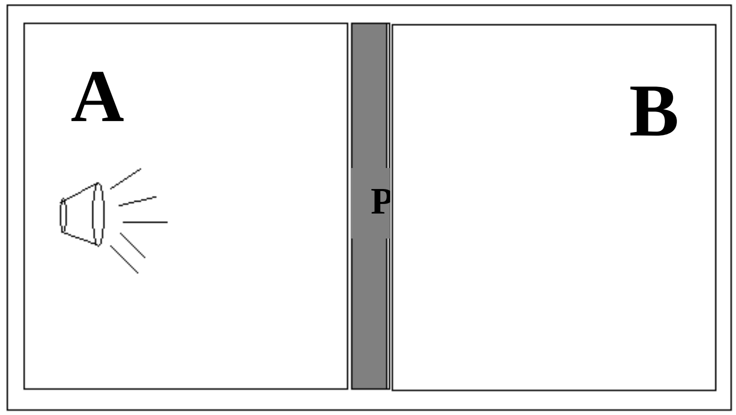

Figure 4 Illustration for showing the mechanism of sound transmission through walls.

(3) If P is non-porous and not absolutely rigid, the alternating pressure changes on the surface will setup minute flexural vibrations of the partition, which will in turn setup vibrations in B.

Of the three modes of transmission, the first is of negligible importance in any practical case, in comparison with the other two. When sound in one medium is incident upon the surface of the second medium, a part of its energy is reflected back into the first medium and a part is refracted. The ratio of the refracted portion of energy to the incident energy is the ratio of acoustic resistance of the first medium to that of the second. Hence it will be seen that for solid materials, the acoustic resistance is very high as compared to that of air, so that the sound entering the solid partition is a very small fraction of the incident sound and hence very little sound is transmitted through solid walls in this manner.

Measurement Techniques

I) **Method using Sound Pressure Level (SPL) and Reverberation Room**

This test method is a part of evaluating the sound insulating properties using the ASTM E-90(04). In this test method, the sound transmission loss is defined as the difference in decibels between the average sound pressure levels in reverberant source and receiving rooms, plus ten times the common logarithm of the ratio of the area of common partition to the sound absorption in the receiving room.

i.e., $$TL = SPL1 - SPL2 + 10 \times (S/A) \qquad (2)$$

where:

SPL1 – is the space/time averaged sound pressure level in the source chamber in dB with the reference pressure of 2×10^{-5} Pa.

SPL2 – is the space/time averaged sound pressure level in the receiving chamber in dB with the reference pressure of 2×10^{-5} Pa.

S – is the surface area of the test specimen (m^2).

A – is the total absorption in the receiving chamber with the test specimen in place (metric Sabines).

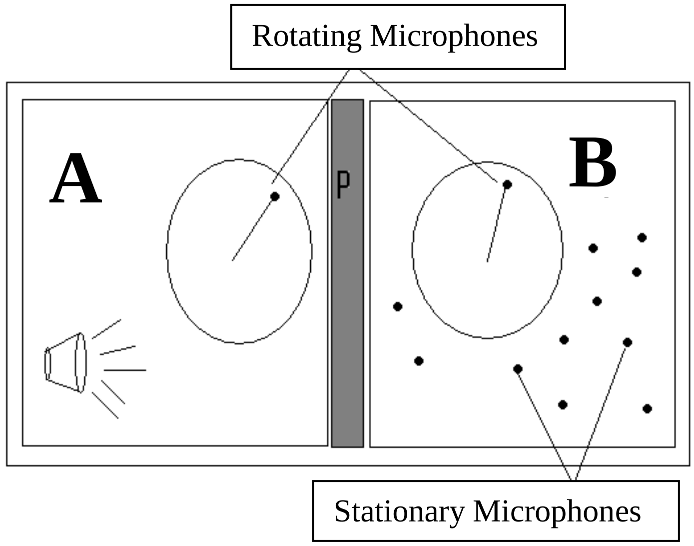

Figure 5 Illustration showing conceptual arrangement of a wall Sound Transmission Loss suite

If the room A in Fig.5 is considered to be the source chamber and B be the receiving rooms. Room A has moving microphones to measure the averaged SPL; the figure shows B has two types of microphones stationary and moving to measure the averaged SPL. However in practice, only one kind of measurement technique (moving microphone or stationary microphone) is used in both the rooms. While a continuous and constant power broadband noise is generated in the source room and microphones in position as shown in Fig.5, the space/time averaged SPL in the source and receiving rooms are measured to determine the STL using eq. (2) [7].

II) Decomposition Method

This method utilizes the ratio between the incident sound energy to transmitted sound energy to determine the sound transmission loss of the specimen.

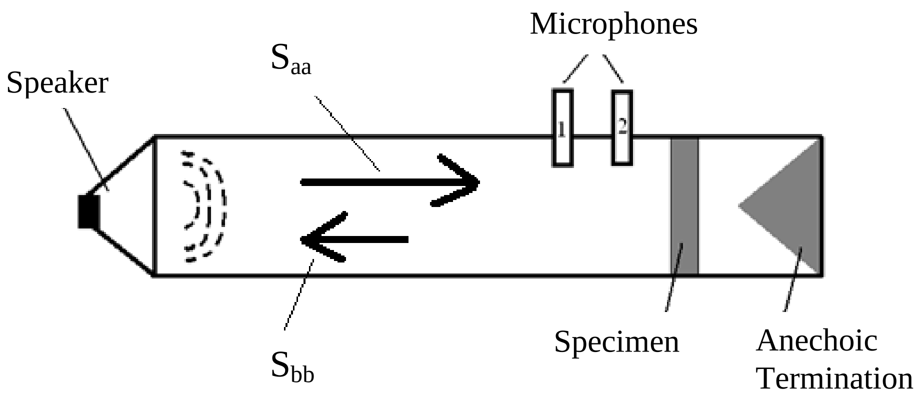

Figure 6 Illustration for Decomposition Method

For 1-dimensional sound produced for the configuration as shown in Fig.6 the sound transmission loss is calculated using the equation (3).

$$STL = 10 \log_{10} (W_i / W_t) \qquad (3)$$

where, $W = p^2 \times S / \rho \times c$ (4)

And p_i and p_t are determined using the auto and cross spectra from the response at mic1 and mic2. The amplitude of the incident sound pressure wave is found by: $p_i = (S_{aa})^{0.5}$.

The sound pressure can be decomposed into its incident and reflected spectra S_{aa} and S_{bb} respectively. By decomposition theory the auto spectrum of the incident wave is

$$S_{aa} = \frac{S_{11+} S_{22} - 2\, C_{11} \operatorname{Cos} K\, x_{12} + 2\, Q_{12} \operatorname{Sin} K\, x_{12}}{4 \operatorname{Sin}^2 K\, x_{12}}$$

Where S_{11} and S_{22} are the auto spectra of the total acoustic pressure points at 1 and 2respectively; C_{12} and Q_{12} are real and imaginary parts of the cross-spectrum between 1 and two, K is the wave number and x_{12} is the distance between the microphones [8].

III) SAE J1400

This method uses the same setup as shown in Fig.5. The noise reduction (NR) is measured using pressure microphones. NR is simply the volume averaged sound pressure level (SPL) difference between the source room and the receiving room. A frequency dependant correction factor is then subtracted from the NR to directly yield the STL. This correction factor is computed using the mass law for a homogeneous limp panel. The NR is measured with a reference panel in place between the two chambers and the correction factor (CF) is then calculated by subtracting the theoretical STL of the reference panel from the NR measured with reference panel in place.

The equation for the calculation of STL using this method is:

$$STL = 20\times\log_{10} w + 10\times\log_{10} f - CF \tag{5}$$

where:

w – is the surface weight of panel in Kg/m^2

f – is the frequency in Hz

IV) ASTM E336

This method measures what is known as field transmission loss (FTL). It is intended to measure the field transmission loss of panels that are installed in working environment. The same setup as described in the above method is used to measure FTL. The SPL in the source and receiving rooms are measured to determine the NR. A correction similar to that described in SAE J1400 method is calculated and the FTL is determined using the eq.(6)

$$\text{FTL} = \text{NR} - \text{CF} \tag{6}$$

V) Transfer Matrix Method

The transfer matrix method involves calculation of transformation matrix or the so called transfer matrix for the calculation of STL. This method involves in calculation of four elements of the 2X2 transfer matrix, which are unknown. The transfer matrix is

$$\begin{bmatrix} p_1 \\ v_1 \end{bmatrix} = \begin{bmatrix} A & B \\ C & D \end{bmatrix} \begin{bmatrix} p_2 \\ v_2 \end{bmatrix}$$

Where p1, v1 and p2, v2 are the pressure and velocities at two faces of the sample. The schematic of the setup is shown in Fig. 7 [9].

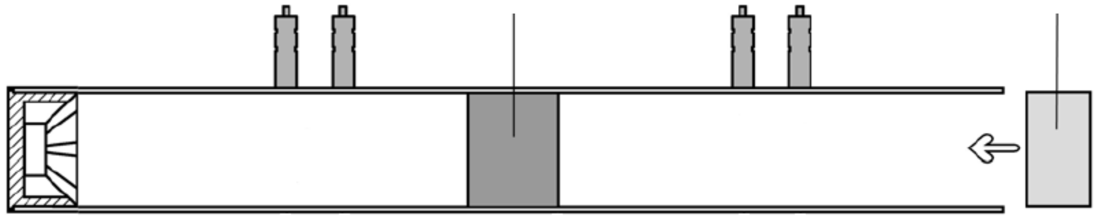

Figure 7 Modified Impedance Tube for Four-Microphone Method.

On determining the four unknowns, the STL is calculated using the eq. (7)

$$\mathrm{STL} = 10 \times \log \{0.25 \times \left(abs(A + (B/(\rho \times c)) + (\rho \times c \times C) + D)\right)^2\} \tag{7}$$

CHAPTER II

THEORY

Basic Model for Development of Transfer-Matrix

Consider the sound transmission in a layer of acoustic medium under study, of thickness L as shown in Fig. 8. Let the input side be X=0 and X=L be the output side.

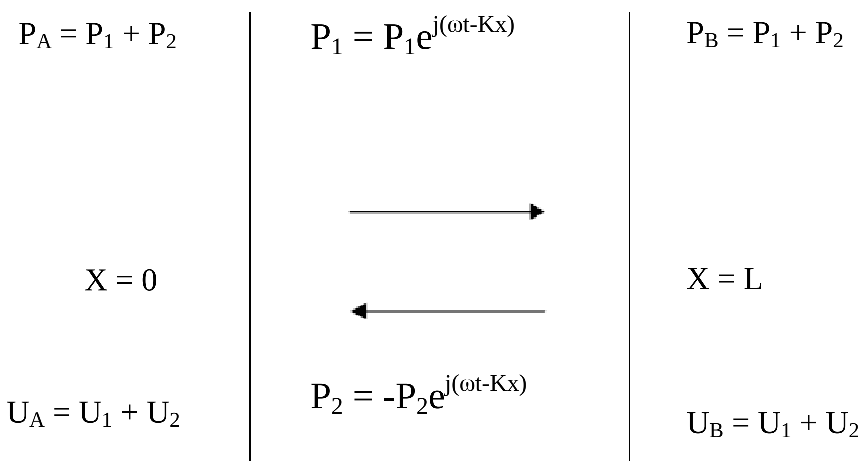

Figure 8 Illustration for sound pressure and velocities

The right travelling wave is,

$p_1 = P_1 \times e^{j(\omega t-kx)}$ and $u_1 = P_1 \times e^{j(\omega t-kx)} / (\rho \times c)$

Where: p_1 and u_1 are the pressure and velocity at position x.

Similarly, the left travelling wave is given by,

$p_2 = P_2\times e^{j(\omega t-kx)}$ and $u_2 = -P_2\times e^{j(\omega t+kx)} / (\rho\times c)$

Now, at X=0 i.e., at the input side, using the boundary conditions:

$P_A = P_1 + P_2$ and $\rho\times c\times U_A = P_1 – P_2$

And at x=L, the pressure and velocities become:

$P_B = P_1\times e^{-jkL} + P_2\times e^{jkL}$ and $\rho\times c\times U_B = P_1\times e^{-jkL} - P_2\times e^{jkL}$

Adding the above two equations:

$P_1 = 0.5\times e^{jkL}(P_B + (\rho\times c\times U_B))$

Subtracting the two equations:

$P_2 = 0.5\times e^{-jkL}(P_B - (\rho\times c\times U_B))$

Using the values to determine P_A and U_A:

$P_A = P_B\times \text{Cos } kL + j\times r\times c\times U_B\times \text{Sin } kL$ and

$U_A = (j/\rho c)\ P_{B\times}\ \text{Sin } kL + U_B\times \text{Cos } kL$

Writing the above equations in matrix form:

$$\binom{\mathrm{Pa}}{\mathrm{Ua}} = \begin{bmatrix} Cos\ kL & j\rho c\ Sin\ kL \\ \frac{j}{\rho c} Sin\ kL & Cos\ kL \end{bmatrix} \binom{\mathrm{Pb}}{\mathrm{Ub}} = \begin{bmatrix} A & B \\ C & D \end{bmatrix} \binom{Pb}{Ub}$$

Where the matrix ABCD is known as transfer/transformation matrix while the equations relates the input and output pressure and velocity variables with the subscripts ‘a’ and ‘b’ being the input and output sides respectively.

Development of Transfer Matrix for Calculation of STL Using Four Microphones Method

Now, this basic formulation is extended for our four microphone impedance tube method, which consists of two variables at each side of the tube. Fig.9 shows the setup with four microphones and the left and right travelling waves with marked intensities on both sides of the sample.

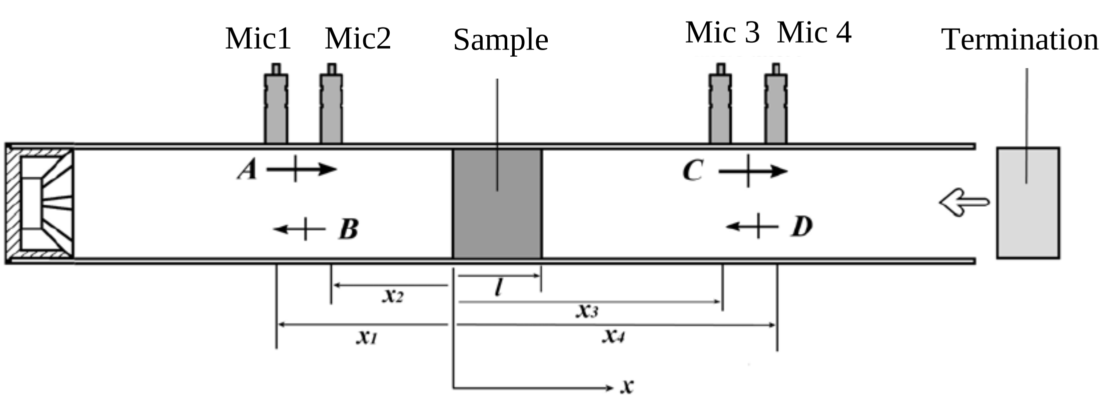

Figure 9 Setup for Four-Microphone Method

As shown in Fig.9, the approach used here consists of a loud speaker to generate plane wave field in the tube and the microphones are used to measure the transfer function between the signal provided to the loudspeaker and the sound pressure level at the four microphone positions. But for the current derivations we will be considering sound pressure instead of the transfer functions and denote them with P_1 to P_4. Writing the equations for the complex pressures at all the four positions:

$$P(x,t) = \begin{cases} \left(A(\omega)e^{-jkx} + B(\omega)e^{jkx}\right)e^{j\omega t}, & x \le 0 \\ \left(C(\omega)e^{-jkx} + D(\omega)e^{jkx}\right)e^{j\omega t}, & x \ge l \end{cases} \tag{8}$$

$$v(x,t) = \begin{cases} \frac{A(\omega)e^{-jkx} - B(\omega)e^{jkx}}{\rho \times c}\, e^{j\omega t}, & x \le 0 \\ \frac{C(\omega)e^{-jkx} - D(\omega)e^{jkx}}{\rho \times c}\, e^{j\omega t}, & x \ge l \end{cases} \tag{9}$$

i.e., $P1 = (Ae^{-jkx1} + Be^{jkx1})e^{j\omega t}$ (10)

$P_2 = (Ae^{-jkx2} + Be^{jkx2})e^{j\omega t}$ (11)

$P_3 = (Ce^{-jkx3} + De^{jkx3})e^{j\omega t}$ (12)

$P_4 = (Ce^{-jkx4} + De^{jkx4})e^{j\omega t}$ (13)

Where P and v are the complex pressure and complex particle velocity respectively, A, B, C and D are the complex amplitudes of the plane wave components, ρ is the ambient fluid density, c is the ambient sound speed, ω is the angular frequency, $k = (\omega/c)-j\alpha$ is the complex wave number and l is the thickness of the sample. Equations 10-13 yield the unknown coefficients A to D in terms of complex sound pressures and the distance of microphones form the sample i.e.,

$$A = \frac{j\left(P_1 e^{jkx_2} - P_2\, e^{jkx_1}\right)}{2\, Sin\, k\, (x_1 - x_2)}\ , \quad B = \frac{j\left(P_2 e^{-jkx_1} - P_1\, e^{-jkx_2}\right)}{2\, Sin\, k\, (x_1 - x_2)} \tag{14}$$

$$C = \frac{j\left(P_3 e^{jkx_4} - P_4\, e^{jkx_3}\right)}{2\, Sin\, k\, (x_3 - x_4)}\ , \quad D = \frac{j\left(P_4 e^{-jkx_3} - P_3\, e^{-jkx_4}\right)}{2\, Sin\, k\, (x_3 - x_4)}$$

These coefficients A, B, C and D are required for determining the elements of the transfer matrix. Now, from the basic formulation we know that,

$$\begin{bmatrix} \mathrm{P}_1 \\ v_1 \end{bmatrix} = \begin{bmatrix} \mathrm{T}_{11} & \mathrm{T}_{12} \\ \mathrm{T}_{21} & \mathrm{T}_{22} \end{bmatrix} \begin{bmatrix} \mathrm{P}_2 \\ v_2 \end{bmatrix} \tag{15}$$

Where the subscripts 1 and 2 are the locations at surface of the sample i.e., x=0 and at x=L.

Therefore using the boundary conditions, from equations 10 -13 we have

$P_a = A + B$ and $v_a = (A – B)/\rho c$

$P_b = Ce^{-jkl} + De^{jkl}$ and $v_b = (Ce^{-jkl} + De^{jkl})/\rho c$

From eq. 14 it is clear for us that we have four unknowns with only two equations [10]. This criterion is satisfied by using two conditions in the testing method, an open ended tube and a closed anechoic termination as shown in Fig.6 and the unknowns of the matrix can be easily calculated.

Therefore considering two conditions with two different types of terminations, eq.14 can represented as

$$\begin{bmatrix} \mathrm{P_a} \\ v_a \end{bmatrix}_{x=0} = \begin{bmatrix} \mathrm{T}_{11} & \mathrm{T}_{12} \\ \mathrm{T}_{21} & \mathrm{T}_{22} \end{bmatrix} \begin{bmatrix} \mathrm{P}_b \\ v_b \end{bmatrix}_{x=L} \tag{16}$$

Where the subscripts 'a' and 'b' denote the two different termination conditions. Hence the elements of the transfer-matrix can be now easily calculated as:

$$T_{11} = \frac{1}{P^a_{x=L} \times v^b_{x=L} - P^b_{x=L} \times v^a_{x=L}} \times \left\{P^a_{x=0} \times v^b_{x=L} - P^b_{x=0} \times v^a_{x=L}\right\} \tag{17}$$

$$T_{12} = \frac{1}{P^a_{x=L} \times v^b_{x=L} - P^b_{x=L} \times v^a_{x=L}} \times \left\{P^a_{x=0} \times P^b_{x=L} - P^b_{x=0} \times P^a_{x=L}\right\} \tag{18}$$

$$T_{21} = \frac{1}{P^a_{x=L} \times v^b_{x=L} - P^b_{x=L} \times v^a_{x=L}} \times \left\{v^a_{x=0} \times v^b_{x=L} - v^b_{x=0} \times v^a_{x=L}\right\} \tag{19}$$

$$T_{22} = \frac{1}{P^a_{x=L} \times v^b_{x=L} - P^b_{x=L} \times v^a_{x=L}} \times \left\{P^b_{x=L} \times v^a_{x=0} - P^a_{x=L} \times v^b_{x=0}\right\} \tag{20}$$

With all the coefficients and elements of the transfer-matrix determined, the sound transmission loss is calculated using the formula [11]:

$$\mathrm{STL} = 10\log\left(\frac{1}{4}\left|\mathrm{T}_{11} + \frac{\mathrm{T}_{12}}{\rho c} + \rho c\mathrm{T}_{21} + \mathrm{T}_{22}\right|^{2}\right) \tag{21}$$

CHAPTER III

TESTING FOR SOUND ABSORPTION COEFFICIENT: TWO MICROPHONE METHOD

As described in the theory of formulation for the four-microphone method, which involves in calculation of complex unknowns, using the complex acoustic pressure and particle velocities which are to be measured using two microphones. As a matter of convenience and to overcome the errors and difficulties involved in performing the four-microphone method directly, a simpler and less complex method is tested which yields the sound absorption coefficient of the material being tested. This is the two-microphone method and makes the working and understanding of the four-microphone method easier and simpler. This method also forms the starting point for the development of the four microphones method. The equipment required for both the methods is almost the same and hence we will describe this method in detail.

Scope, Significance and Standard

This test method was developed with ASTM E-1050 as reference which describes the two-microphone method or the transfer function method for measuring the sound absorption coefficient of acoustical materials. This test method can be applied to measure sound absorption coefficient of absorptive materials at normal incidence i.e., 0^0.

Normal incidence sound absorption coefficients are useful in basic research and product development of sound absorptive materials like aerogels. This method is quite useful in situations where the material being tested in placed in cavities just like the present case of impedance tube. This method involves the use of impedance tube (with two microphones), a digital oscilloscope, and measurement of complex pressures using the microphones, from the source speaker which has an input from white noise generator.

Equipment and Requirements

The apparatus is a hollow cylinder or tube, which holds a loud speaker at one end and the test sample at the other. Two microphones are mounted at a specified distance apart, long the length of the tube. The output of the microphone is connected to a multi-channel oscilloscope, for data acquisition and further processing/calculations [12].

Tube

The tube can be constructed from materials including metal, plastic, cement or wood. It must be assured that the interior walls of the tube are smooth in order to maintain low sound attenuation for plane waves. The interior section of the tube can be circular or rectangular with constant dimension from end to end. The tube must be straight and it's inside surface shall be smooth, nonporous and free of dust to maintain low attenuation. The tube construction shall be massive so that the transmission through the tube walls is negligible. The tube should also be sufficiently long to ensure plane waves are fully developed before reaching the microphones and test specimen. A minimum of three tube diameters must be allowed between sound source and nearest microphone. The sound source generates non-plane waves along with plane waves and this distance of three tube diameters between the sound source and the nearest microphone ensure that the non-plane waves subside before reaching the nearest microphone. To achieve the low background noise, the tube must be constructed of heavy materials and must be sealed properly at all openings so that transmission loss is low.

Sound Source and Signal Strength

The sound source should provide enough sound energy over the entire frequency range of interest. The working frequency range is:

$$f_l < f < f_u \quad (22)$$

where:

f – is the operating frequency, Hz,

f_l – is the lower working frequency of the tube, Hz and

f_u – is the upper working frequency of the tube, Hz.

The upper frequency limit, f_u depends upon the diameter of the tube and the speed of the sound. In order to maintain plane wave propagation, the upper limit is defined as:

$$f_u < Kc/d \tag{23}$$

f_u – is the upper frequency limit, Hz,

c – is the speed of sound in tube, m/s,

d – is the diameter of the tube, m and

K = 0.586.

The lower frequency limit f_l depends on the spacing of the microphones and accuracy of the analysis system. It is recommended that the microphone spacing exceed 1% of the wavelength corresponding to the lower frequency of interest.

It is best to conduct the plane wave measurements well within these frequency limits in order to avoid cross modes that occur at high frequencies when the acoustical wavelength approaches the sectional dimension of the tube.

Accurate measurements of material absorption require that the sound field be substantially larger than the background noise inside the tube. The minimum signal to noise ratio will occur at the minima of the standing waves, these minima can be as much as 25 dB below the maximum levels in the tube. The standards recommended that the level of sound in the tube be at least 10 dB greater than the background noise level, but 20-30 dB is preferred. Taken together, these conditions require a background noise level in the tube in the 10-20 dB range. To achieve the low background noise, the tube must be constructed of heavy materials and must be sealed properly at all openings so that transmission loss is low.

Microphones and Microphone Spacing

Two nominally identical microphones are mounted along the length of the tube. The microphones must be flush mounted with the inside wall of the tube and isolated from the tube to minimize sensitivity to vibration. The diameter of the microphones must be small in comparison with the spacing between microphone ports and also to minimize the spatial averaging at higher frequencies across the diaphragm face. The standard recommends the diameter of the microphone to be 20% lesser than the wavelength for the highest frequency of interest. A large spacing between the microphones enhances the accuracy of the measurements; however, the microphone spacing must be less than the shortest half wavelength of interest.

$$S << c/2\times f_u \tag{24}$$

where:

s - is the microphone spacing, m

c - is the speed of sound, m/s and

f_u – is the upper frequency limit, Hz.

It is recommended that the maximum microphone spacing, s, be 80% of $c/2f_u$. The minimum distance between the sound source and the nearest microphone has been described in earlier section and the minimum distance between the specimen and the closest microphone depends upon the surface characteristics of the specimen. Care must be taken, if the microphones are switched, when the microphones are removed from their port, so that the original mounting geometry is maintained when they are replaced.

Test Signal

It is recommended that the test signal be random noise having a uniform spectral density across the frequency range of interest. The spectral line spacing of the test signal should be compatible with the analysis bandwidth. Alternative test signals also may be used if they have an equivalent spectral density. These alternative signals include pseudo-random noise and swept or stepped sine generation. In the course of this work, white noise was used as a test signal because it gave better results as compared with other signals.

The sound source shall generate sufficient signal at both microphone locations such that the measured signal in each test frequency band is at least 10dB greater than the background noise. The technique of ensemble averaging has the effect of reducing uncertainties due to the variance of random noise. 16 averages were used. Analysis is made on the blocks of data as a time record of finite length. In the course of this work a

Hamming window was used for the measurement of transfer function spectra since it produced relatively better results than the other windows.

Consideration of the Attenuation Constant

The incident and reflected waves that propagate within the tube are subject to attenuation due to viscous and thermal losses. This effect causes the loci of the pressure minimums to shift asymmetrically in the standing wave pattern as distance from the specimen increases. In order to account for the tube attenuation, work was carried out using the equations involving the calculation of the reflection coefficient. The modification was made by replacing the real variable k by the complex wave number $k = k_r - j\times\alpha$. Here k_r is the real component of the wave number where and α is the attenuation constant.

where:

$$\alpha = 0.02203\,\frac{\sqrt{f}}{c\times d}\ \text{ and } \mathrm{K_r} = \frac{2\times\pi\times f}{c} \tag{25}$$

Experiments were conducted and calculations were performed considering the attenuation effect and also without considering the effect, from the results it was observed that there was very little difference using the attenuation constant since the value has negligible effect, the attenuation constant for air being very low.

Other Equipment

For the signal processing, the required signals are recorded using the flush mounted microphones which are then amplified using a microphone pre-amplifier in this course of work. The amplified output of the microphone is then connected to the digital oscilloscope to obtain a data. The digital oscilloscope must be a two channel (a four channel is used to incorporate for the four microphone method). The digital oscilloscope is connected to a desktop computer to perform the digital frequency analysis including the calculation of transfer function [13].

Theory

As described earlier, the two microphone method, also known as the transfer function method uses the transfer function determined from the ratio of the cross spectrum to the input auto spectrum. This could also be calculated directly from the complex ratio of the Fourier transform of the acoustic pressure at the microphone nearest to the test specimen to the Fourier transform of the microphone nearest to the sound source. For a single measurement both yield same result. Fig. 10 shows the schematic of the tube, sound source and microphone setup for this test.

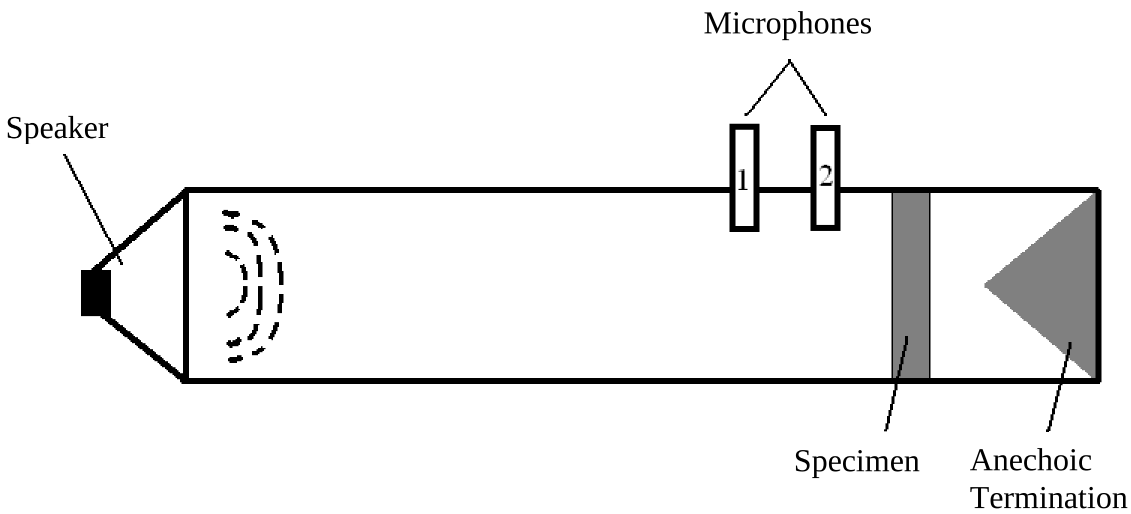

Figure 10 Impedance Tube method for determining Sound Absorption

The pressure amplitudes for the incident and reflected waves at any position x is given by:

$P_+ = A \times e^{jk(L-x)}$ and $P_- = B \times e^{-jk(L-x)}$

Where the subscripts '+' and '-' indicate the forward and backward travelling wave component.

The relationship between the ratios of incident and reflected waves measured at the first microphone, the second microphone is located at a distance 'l' from the end and's' is the spacing between the two microphones then:

$$\frac{P_-}{P_+} = \frac{B}{A} e^{-2jk(l+s)} = Re^{-2jk(l+s)} \quad (25)$$

where, L-x = l + s

Now, the transfer function measure between microphones 1 and 2 is defined as:

$$H_{12} = \frac{P_2 P_1^*}{P_1 P_1^*} = \frac{\text{cross power spectrum}}{\text{auto power spectrum}}$$

$$\rightarrow \ H_{12} = \frac{P_2}{P_1} = \frac{Ae^{jk(l+s)} + Be^{-jk(l+s)}}{Ae^{jkl} + Be^{-jkl}} = \frac{P_+ e^{-jks} + P_- e^{-jks}}{P_+ + P_-}$$

Therefore, $\dfrac{P_-}{P_+} = \dfrac{e^{-jKs} - H_{12}}{H_{12} - e^{jKs}}$

Then the reflection coefficient is given by:

$$R = \frac{P_-}{P_+} e^{2jK(l+s)} = \frac{e^{-jKs} - H_{12}}{H_{12} - e^{jKs}} \times e^{2jK(l+s)} \tag{26}$$

And the sound absorption coefficient is given by:

$$\alpha = 1 - R \times R^* \tag{27}$$

where: R^* is the complex conjugate.

Since the transfer function is a complex ratio of the acoustic pressure responses, any mismatch in the amplitude or phase responses of the two microphone systems will affect the accuracy of the transfer function measurement. The following sequence of measurements and computations are carried out for correcting the measured transfer function data in both measurement channels.

A highly absorptive specimen is used to ensure the calibration procedure and to prevent strong acoustic reflections and to obtain most accurate correction factor possible. Considering the initial microphone position to be the standard configuration, the transfer function is measured to be H^{I}_{12} and now interchanging the microphone positions and the transfer function H^{II}_{12} is measured in the switched configuration. Now the calibration factor H_c is calculated as: $H_c = (H^{I}_{12} \times H^{II}_{12})^{1/2}$ [13, 14].

Procedure

Fig.11 shows the experimental setup used for the measurement of the sound absorption coefficient of materials using the two microphone method. A plane wave tube in this experiment is made up of copper with smooth inner walls to provide sound hard boundary conditions. A speaker attached at one end produces the constant broadband sound using the white noise generator. The tube has a cut-off frequency of 4000 Hz in plane wave mode. All contact surfaces are to be smooth to prevent loss of acoustical energy. Two omni directional electret condesor microphones are powered by the Phantom power available from the microphone pre-amplifier.

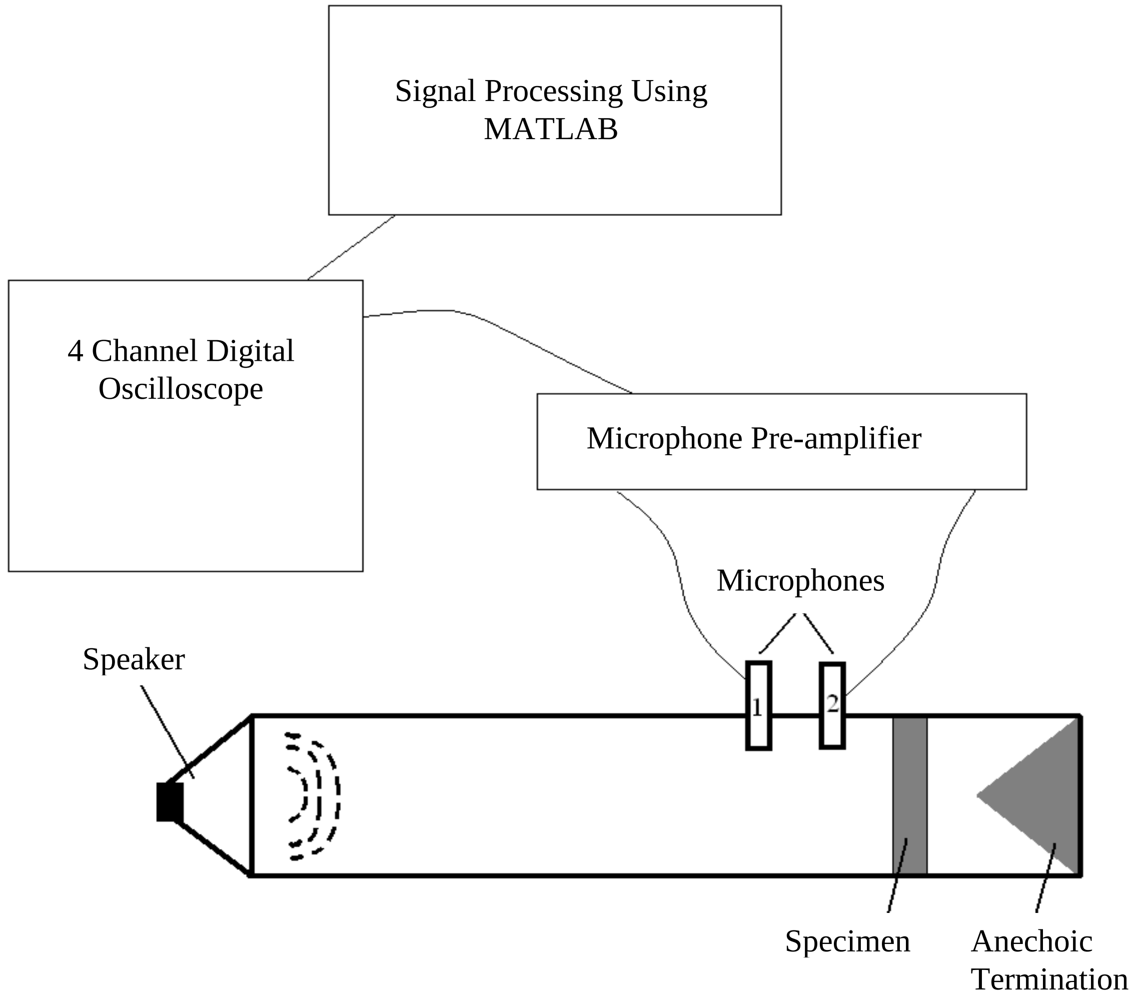

Figure 11 Schematic for the setup for determining Sound Absorption Coefficient

The microphones are flush mounted with the interior wall through holes on the tube. The microphones spacing was adjusted and fixed to obtain a smooth frequency response. A white noise generator is connected to the speaker and the frequency response of the two microphones is determined by using the data recorded from the digital oscilloscope. The material whose properties are to be determined is placed at the other end of the tube, as shown in Fig.11.

The two-microphone method was applied to measure the sound absorption coefficient of various materials as preliminary testing. This test method covers the use of an impedance tube, digital oscilloscope and signal analysis and use of microphones for determination of absorption coefficient of different materials. The sample which is to be tested is placed at one end of the tube and the speaker at the opposite end. The two microphones are flush mounted with the inner walls near the sample end of the tube and the procedure described above was followed. The data acquired from the oscilloscope was fed to the MATLAB program, which would plot the sound absorption coefficient over the frequency range of interest: after performing the required necessary calculations.

Results

Before starting to test the actual samples for the determining the sound absorption coefficient, it must be verified that the setup and tube is yielding the true or at least a very good acceptable results. So to check the working of the setup, and also to get a good calibration factor, a highly absorptive material has to be tested. Therefore, a thick layer of cotton like sponge was used which is considered to absorb a lot of sound. As described earlier, a material is highly sound absorbing if it has a high sound absorption coefficient. Therefore, if a material has a value close to 1 then it is said to be a very good sound absorbing material. Experimental result as expected for this sponge was very high and close to 1 as expected. Fig. 9 shows the plot for the sound absorption coefficient for the sponge over the entire working frequency range. The following figure 10 shows the plot of the measured transfer function of the input white noise, as a measure from the two microphones. From Fig.12 and Fig. 13 the results clearly show an excellent working of the experimental setup with a very good result of almost equivalent to 1, which is the ideal case.

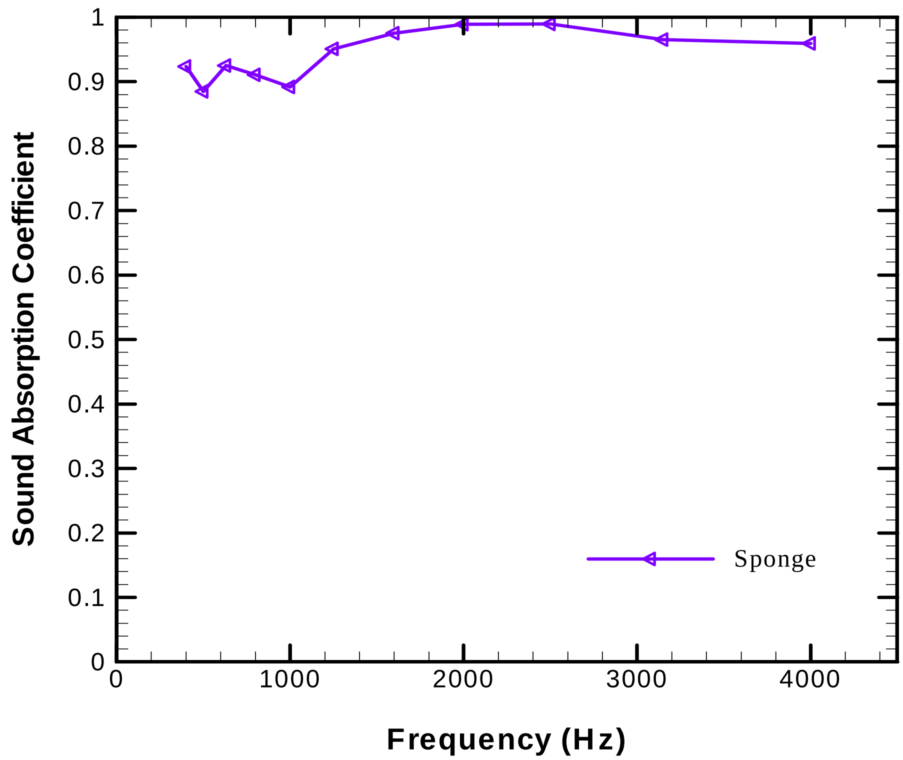

Figure 12 Experimental result for 18mm sponge.

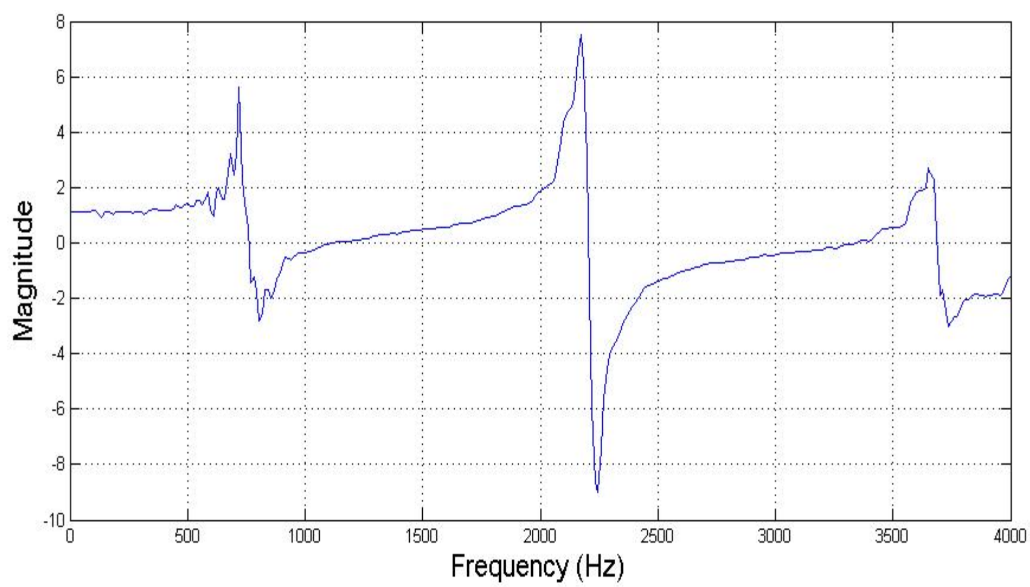

Figure 13 Plot of the measured Transfer function.

With a satisfying result observed for an almost ideal case, the experimental setup is considered to yield good results and experiments were performed on different other materials and then on the aerogel composites. Fig. 14 shows the result for different composites and Fig.15 shows the plot of all the materials that have been tested including the ideal case.

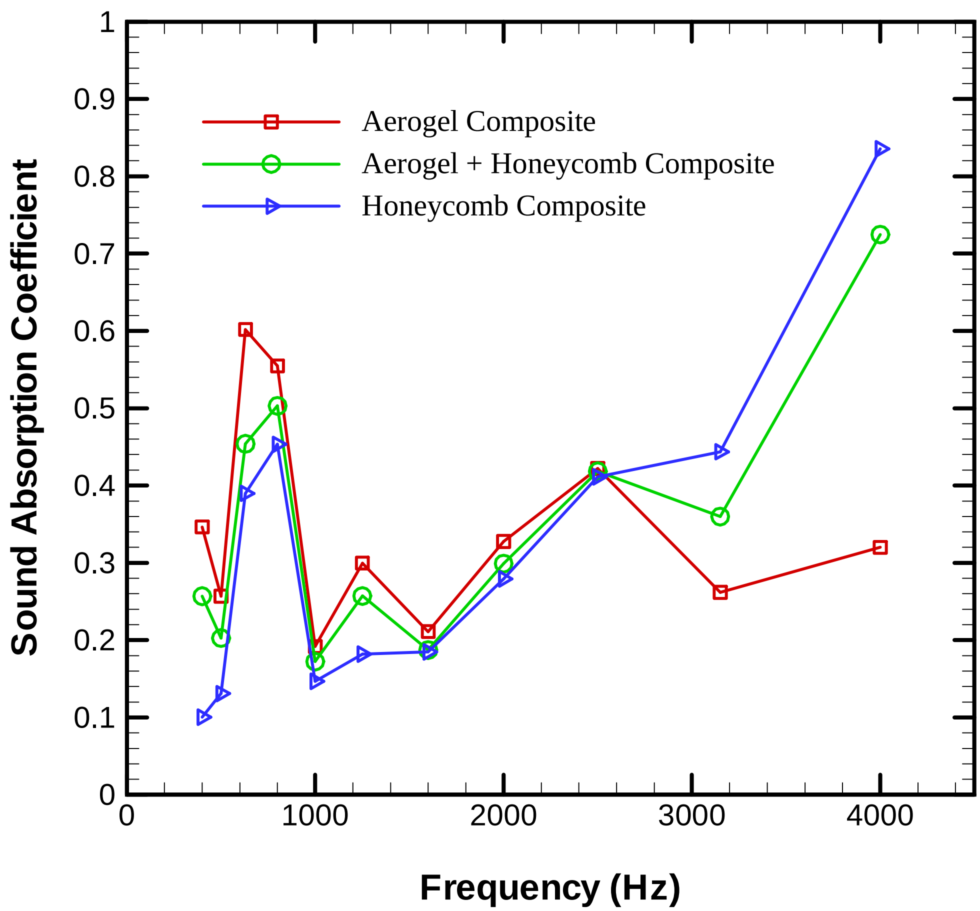

Figure 14 Sound Absorption Coefficient for Aerogel composites.

From Fig. 14 it is clear that the composite with aerogel core has the highest sound absorption coefficient at low frequencies while at higher frequencies i.e., from 3 KHz – 4

KHz, the composite with a core of combined aerogel and honeycomb has a higher sound absorption coefficient than the composite with aerogel core.

Figure 15 below shows the sound absorption coefficient for all the composites and sample of plywood and polyurethane foam with glass fibers. The trend is clearly seen from the figure and can be compared with the ideal case.

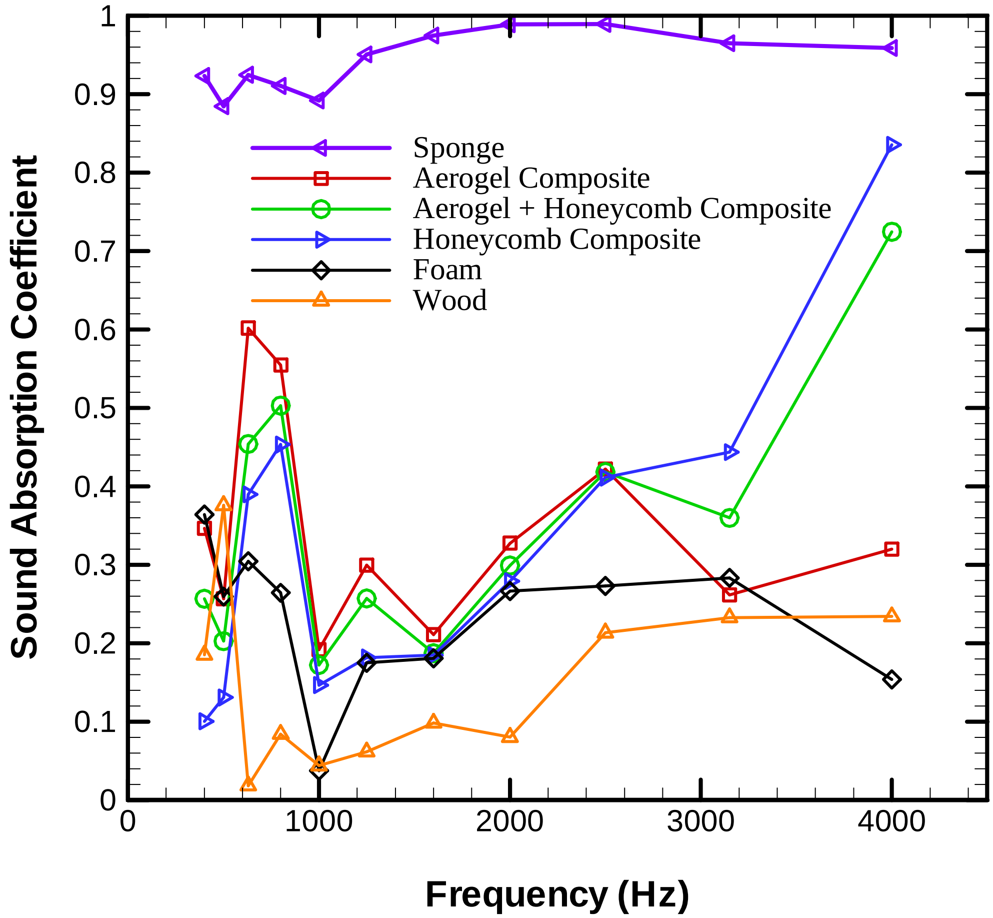

Figure 15 Plot of Sound Absorption Coefficients of the materials tested.

CHAPTER IV

TESTING FOR SOUND TRANSMISSION LOSS: FOUR-MICROPHONE METHOD

As discussed earlier, the four-microphone method involves more complex equations and calculations and to understand the concept of working of the four-microphone method and for easier understanding of the complex equations and calculations, the two-microphone method was performed. With successful experimental setup and results, the working theory was clearly understood and much of the theory involved in the calculations of transmission loss using the four-microphone method was simplified to understand. Also, much of the queries involved in the measurements and usage of the setup were cleared by doing experiments in the simpler two-microphone setup.

Scope and Significance

This test method was developed with ASTM E-2249, SAE J1400, ASTM E-336 and ASTM E-90 as major references which describe the methods for measuring the sound transmission loss of acoustical materials. This test method can be applied to measure sound transmission loss of absorptive materials at normal incidence, i.e., 0°. Normal incidence sound transmission loss is useful in basic research and product development of sound insulating materials like aerogels. This method is quite useful in situations where the material being tested in placed in cavities just like the present case of impedance tube. This method with its highly accurate results clearly provides itself to be replaced with the currently being used reverberation room method which involves in big rooms/chambers and also bigger samples to partition the rooms. By the use of this method, the acoustical properties of the materials can be measured in a smaller area with just a small specimen unlike a big sheet of sample used in the conventional reverberation room [15]. This method involves the use of impedance tube, a digital oscilloscope and signal processing, and measurement of complex pressures and transfer function using the microphones, from the source speaker which has an input from white noise generator.

Equipment and Requirements

While different acoustical properties are determined by these different methods, viz., sound absorption coefficient and sound transmission loss using two-microphone method and four-microphone method respectively, much of the requirements and the experimental setup remains the same for both methods. The four microphone method makes use of another two microphones at the other end of the sample with the tube extending after the sample. Fig. 16 shows the schematic of the tube setup for the four-microphone method.

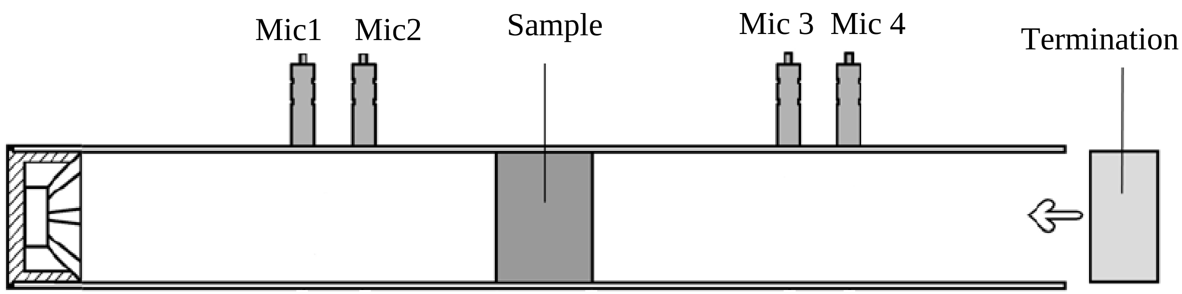

Figure 16 Impedance tube: Four-Microphone Method.

While much of the requirements for the hardware setup remain same, there are a few important changes. The four microphone method uses a longer tube, in fact two tubes of the same size, shape, material and finish, with sample at the center of the tube and two microphones on either side of the sample. A mechanical coupling has been used to maintain the tube-sample-tube order in-line. It was ensured that there was minimal sound energy loss through this coupling by using four air-tight o-rings on the inner side of the coupling to make an air-tight seal with the outside of the tube A sound generating speaker is fitted at one end and two terminations are used at the other end. All other requirements regarding the sound source has be maintained and has to be ensured that the signal strength is 10-20 dB higher than the background noise at all the times. The test signal in this case shall be the broadband noise and a white noise generator has been used to generate the required test signal. Four microphones of same size and shape have been flush mounted at same distance on either ends of the two tubes. A four channel digital oscilloscope has been utilized to record the signal from the four microphones which was connected to a computer for signal processing and other calculations. A temperature and relative humidity indicators were also installed near the impedance tube, as the temperature, pressure and humidity would cause changes in the sound speed and air density.

Theory

Fig. 17 shows the experimental setup along with the forward and backwards travelling sound wave intensities, which are to calculate using the data obtained from the four microphones.

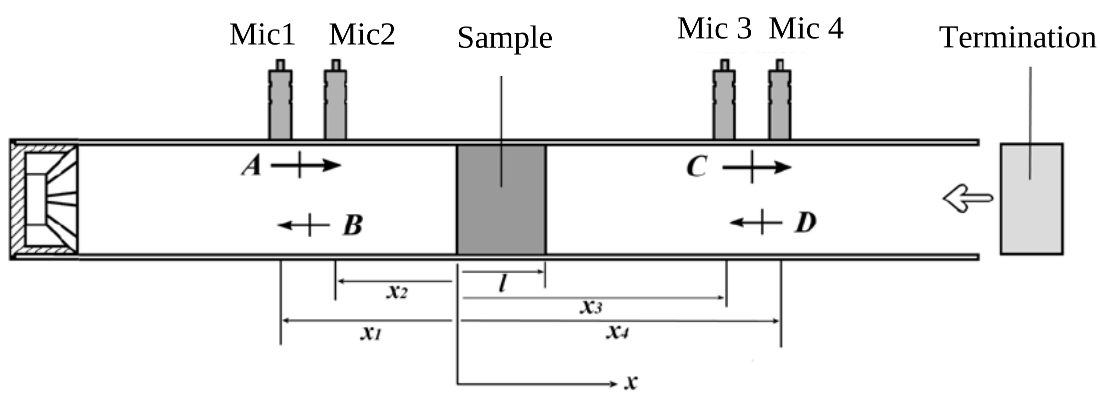

Figure 17 Schematic explanation for four-microphone impedance tube.

While the distances of all the four microphones form a reference plane which in our case is the surface of the specimen, are determined which are to be utilized for determining the pressure and velocities at given distance. For determining the elements of the transfer matrix and the sound transmission loss, the complex sound pressure and velocities at both the surfaces of the sample are to be determined. These are calculated using the following equations:

$$P(x,t) = \begin{cases} \left(A(\omega)e^{-jkx} + B(\omega)e^{jkx}\right)e^{j\omega t}, \ x \leq 0 \\ \left(C(\omega)e^{-jkx} + D(\omega)e^{jkx}\right)e^{j\omega t}, \ x \geq l \end{cases} \tag{28}$$

$$v(x,t) = \begin{cases} \dfrac{A(\omega)e^{-jkx} - B(\omega)e^{jkx}}{\rho \times c} \; e^{j\omega t}, & x \leq 0 \\ \dfrac{C(\omega)e^{-jkx} - D(\omega)e^{jkx}}{\rho \times c} \; e^{j\omega t}, & x \geq l \end{cases} \tag{29}$$

As derived in earlier chapters, the unknowns A, B, C and D are determined using the following equations:

$$A = \frac{j\left(P_1 e^{jkx_2} - P_2\, e^{jkx_1}\right)}{2\, Sin\, k\, (x_1 - x_2)}\,, \quad B = \frac{j\left(P_2 e^{-jkx_1} - P_1\, e^{-jkx_2}\right)}{2\, Sin\, k\, (x_1 - x_2)}$$

$$C = \frac{j\left(P_3 e^{jkx_4} - P_4\, e^{jkx_3}\right)}{2\, Sin\, k\, (x_3 - x_4)}\,, \quad B = \frac{j\left(P_4 e^{-jkx_3} - P_3\, e^{-jkx_4}\right)}{2\, Sin\, k\, (x_3 - x_4)}$$

Now to determine the transfer matrix elements two different termination conditions are used, open ended and anechoic termination. The two conditions are schematically shown in Fig.18 and Fig. 19 respectively.

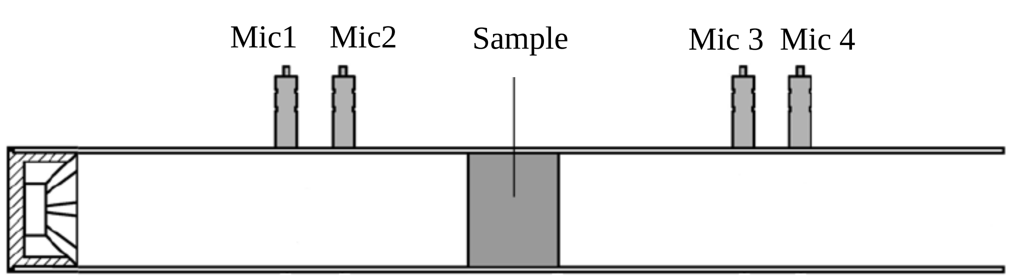

Figure 181 Open Termination.

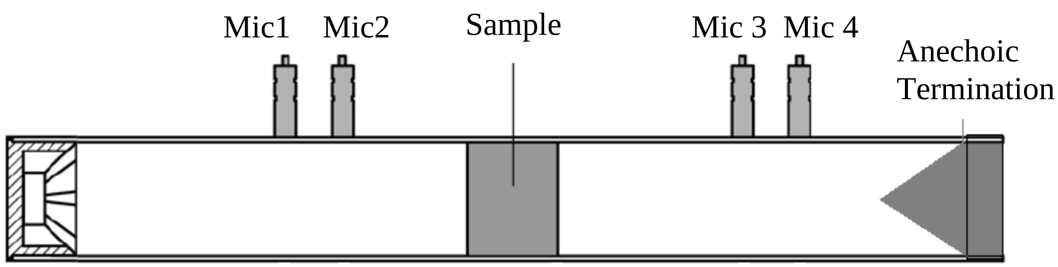

Figure 19 Anechoic Termination.

With the two different conditions all the unknowns are determined and the sound transmission loss is calculated using the following equation:

$$\mathrm{STL} = 10\log\left(\frac{1}{4}\left|\mathrm{T}_{11} + \frac{\mathrm{T}_{12}}{\rho c} + \rho c\mathrm{T}_{21} + \mathrm{T}_{22}\right|^2\right) \tag{30}$$

where:

$$T_{11} = \frac{1}{P_{x=L}^{a} \times v_{x=L}^{b} - P_{x=L}^{b} \times v_{x=L}^{a}} \times \left\{P_{x=0}^{a} \times v_{x=L}^{b} - P_{x=0}^{b} \times v_{x=L}^{a}\right\} \tag{31}$$

$$T_{12} = \frac{1}{P_{x=L}^{a} \times v_{x=L}^{b} - P_{x=L}^{b} \times v_{x=L}^{a}} \times \left\{P_{x=0}^{a} \times P_{x=L}^{b} - P_{x=0}^{b} \times P_{x=L}^{a}\right\} \tag{32}$$

$$T_{21} = \frac{1}{P_{x=L}^{a} \times v_{x=L}^{b} - P_{x=L}^{b} \times v_{x=L}^{a}} \times \left\{v_{x=0}^{a} \times v_{x=L}^{b} - v_{x=0}^{b} \times v_{x=L}^{a}\right\} \tag{33}$$

$$T_{22} = \frac{1}{P_{x=L}^{a} \times v_{x=L}^{b} - P_{x=L}^{b} \times v_{x=L}^{a}} \times \left\{P_{x=L}^{b} \times v_{x=0}^{a} - P_{x=L}^{a} \times v_{x=0}^{b}\right\} \tag{34}$$

Where the superscripts ‘a’ and ‘b’ denote the different termination conditions and subscripts ‘x = 0’ and ‘x = L’ are the locations at either faces of the sample with ‘x = 0’ being the sample surface facing the incoming sound source.

Experimental Setup and Procedure

Fig. 20 shows the complete experimental setup for determining the sound transmission loss using the four-microphone method.

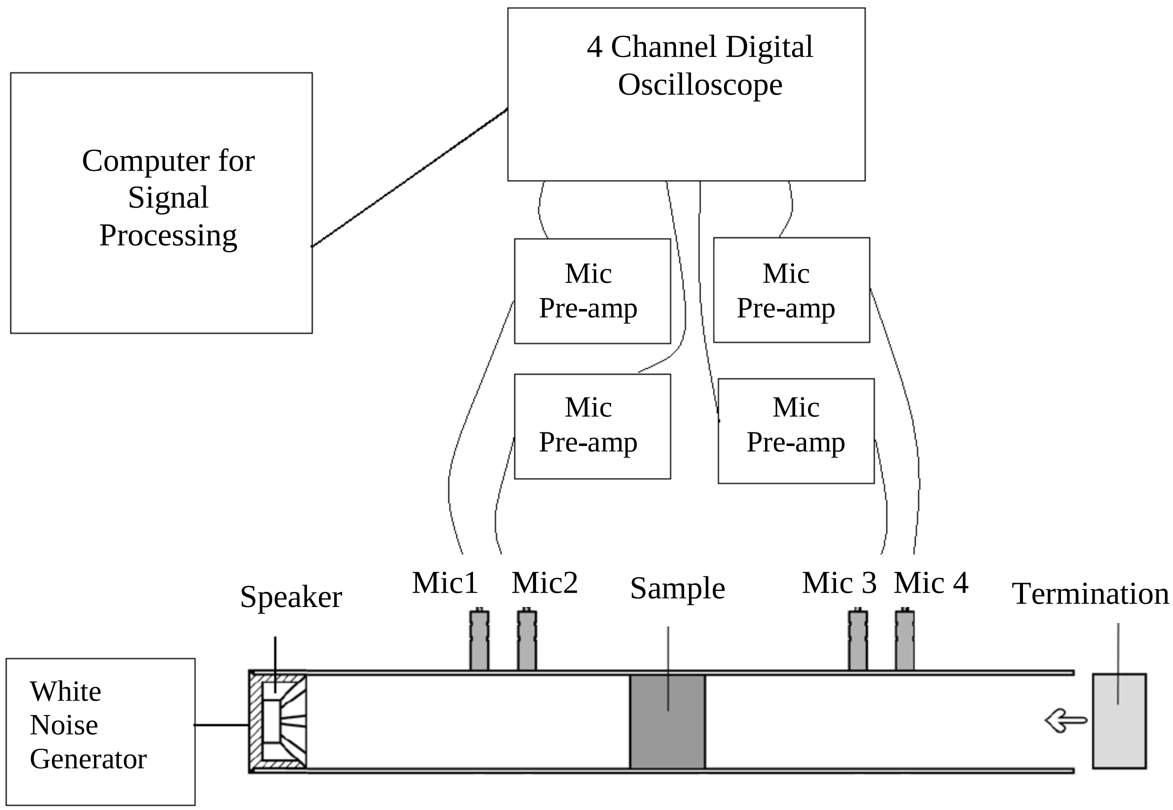

Figure 20 Experimental setup for Four-Microphone Method.

The experimental setup for determining the sound transmission loss using the four-microphone method consists of broadband white noise generator connected to a loudspeaker for the generation of plane waves. The tube is fitted with four microphones, which are flush mounted on the tube along its length. The material whose acoustical properties are to be determined is placed at the center of the tube such that there are two microphones at either side of the sample. The four microphones are individually connected to four identical microphone pre-amplifiers which are in-built with Phantom power to supply the necessary voltage for the working of the microphones. The output of the microphone pre-amplifiers is connected to the four-channel digital oscilloscope to record the signals. The digital oscilloscope is connected to a computer capable of performing signal analysis, calculation of transfer function and other calculations to yield the required sound transmission loss.

The white noise generator is switched on and all microphone amplifiers and the four-channel digital oscilloscope are also turned on. The gain of all amplifiers was adjusted to the same level. The sample was placed inside one-half of the tube and the other half was fitted with the help of the coupling as described in the earlier section. The tube now has a speaker at one end and the other end is left open as the first termination condition. The setup was allowed to stay in this condition for a few seconds before taking the readings; since the white noise generator being used has thirty seconds delayed start. Now the open end of the tube was closed using a cap and an absorbing material to make the anechoic termination. This is the required second termination condition and the readings were taken. The computer which is connected to the oscilloscope now performs the necessary calculations using the MATLAB code to yield the sound transmission loss for the entire working range.

Results

With the successful working of the impedance tube as observed from the results of sound absorption coefficient, the experiments for sound transmission loss were conducted and the following results were obtained. The sound transmission loss (STL) was calculated using the eq.30. The consideration of attenuation constant (α) was of concern here and an experiment was conducted to study how much is the effect of 'α' on the transmission loss

(α – is the attenuation constant and is calculated using eq. (25)). Fig. 21 shows the measured spectrum of the white noise generator and Fig. 22 shows the effect of attenuation constant for experiment conducted on aerogel composite of 0.964 cm thickness.

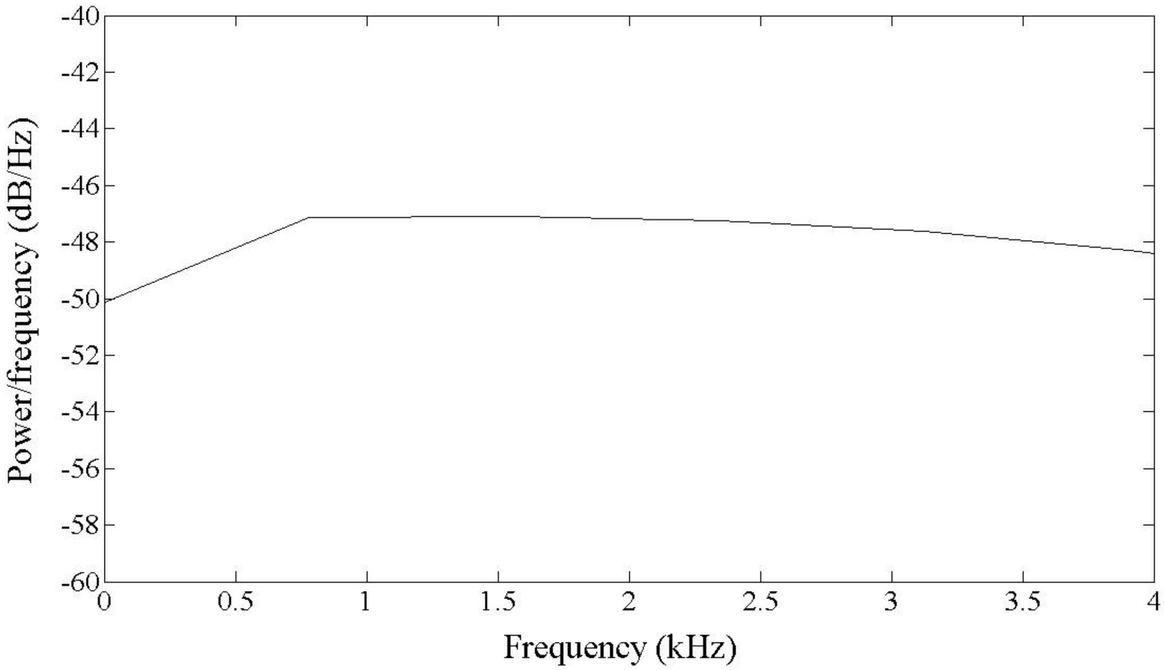

Figure 21 Spectrum of White noise Generator as measured from the microphone

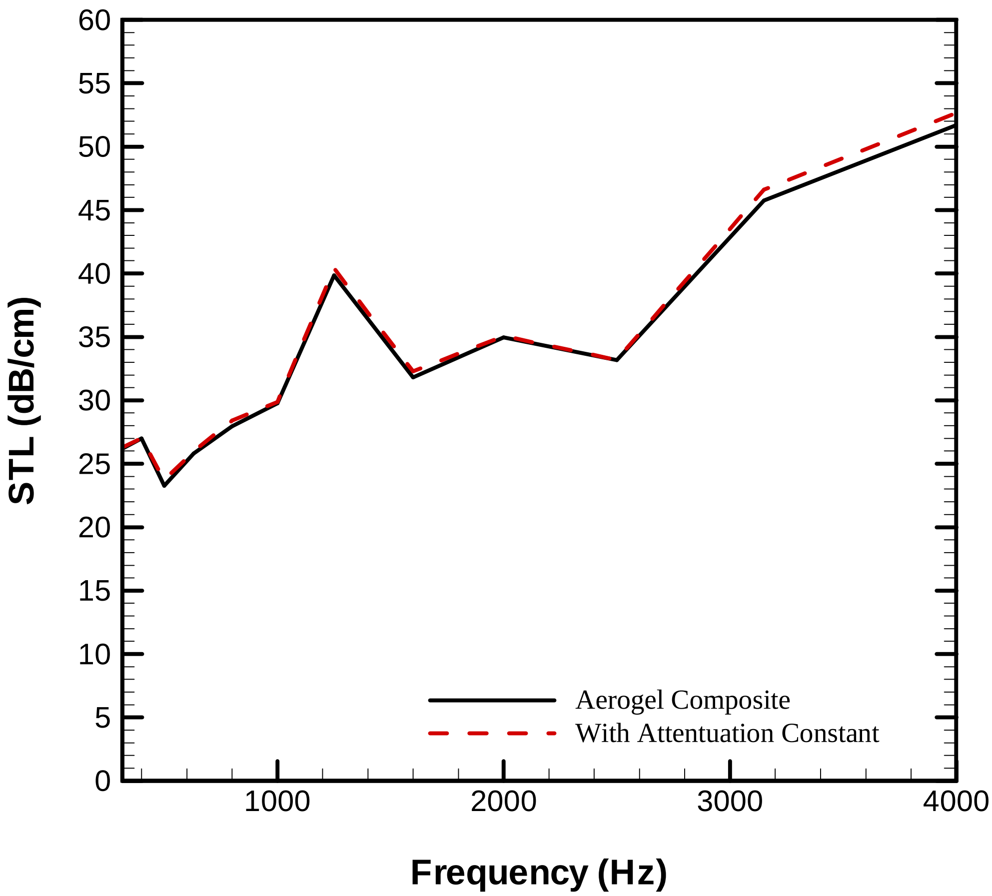

Figure 22 Plot to show the effect of attenuation constant.

Clearly from Fig. 22, the effect of consideration of attenuation constant ‘α’ was not much and was therefore was not considered further. The experiment was conducted on different aerogel samples and a sample of aerogel alone. Fig. 23 shows the results obtained for sound transmission loss of different composites and aerogel.

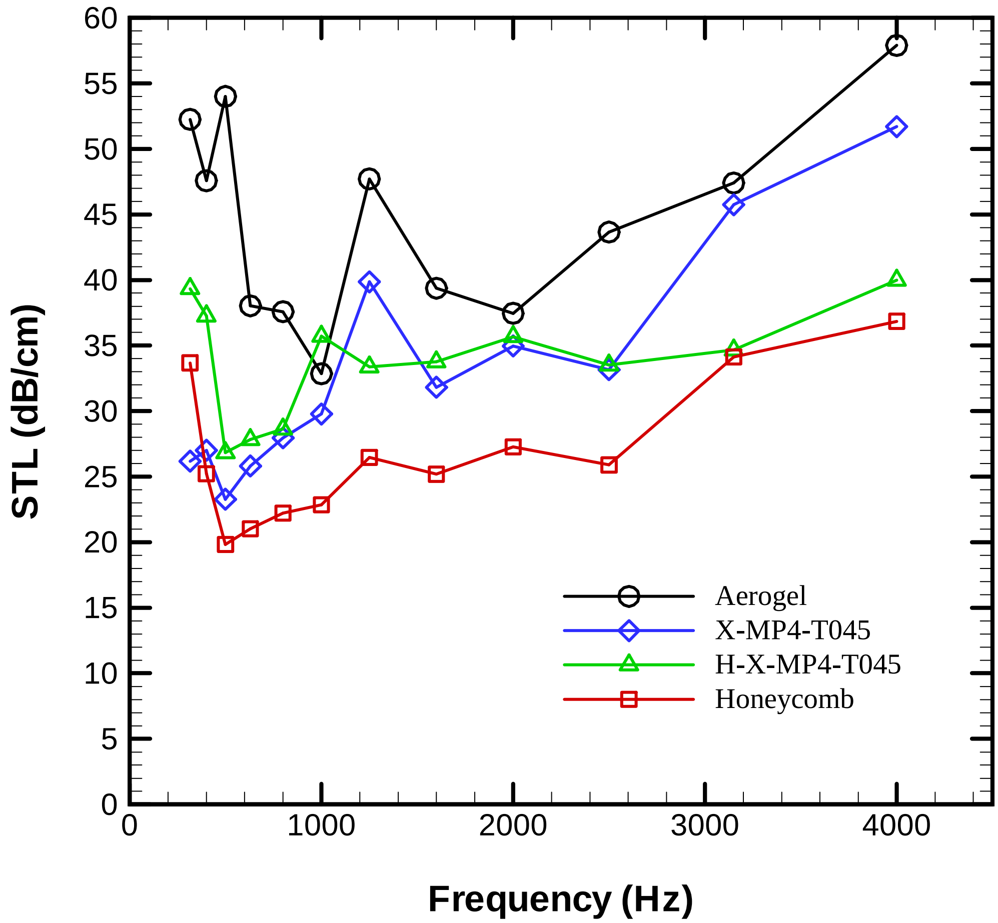

Figure 23 Plot of STL of Aerogel core and Aerogel Composites.

The result has been normalized with the sample thickness for easy comparison of different specimens and plot of STL (dB/cm) against the frequency (Hz) is shown. Tests were conducted on different other samples like wood and polyurethane foam (with glass fibers) to observe and compare the excellent acoustical properties of aerogels.

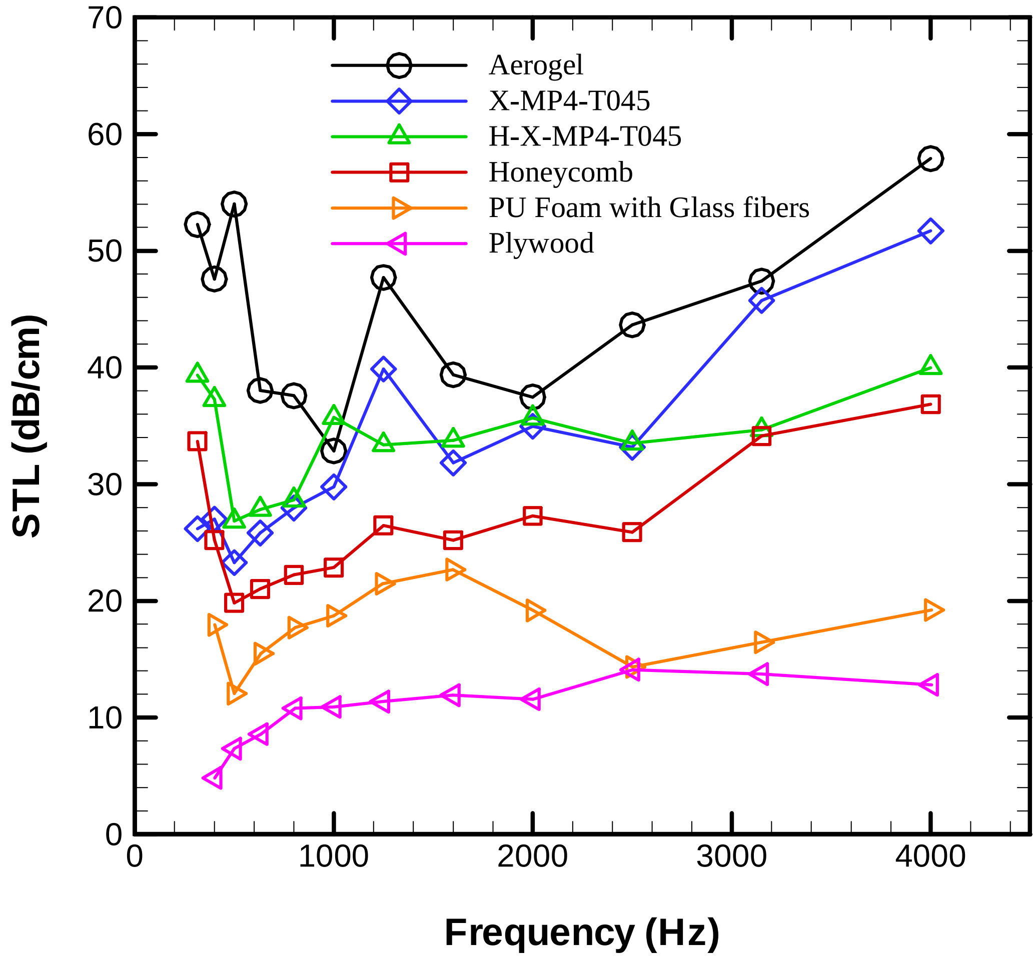

Figure 24 Plot of STL of Aerogel core and other samples.

Fig. 24 shows the plot for aerogel, aerogel composite and other samples as a comparision of aerogel's performance from other samples.

CHAPTER V

CONCLUSION

The main purpose of this thesis is to develop an acoustic impedance tube for easy determination of the sound transmission loss of any material. An impedance tube is developed such that it can evaluate the sound-insulation properties viz., sound transmission loss and absorption coefficient of a given material, upon a slight modification. A conventional impedance tube is constructed to determine the sound absorption coefficient and is then modified by placing two additional microphones at the other end of the sample, the receiving end for the evaluation of the sound transmission loss. This thesis describes in detail the development of the conventional impedance tube and also the modification made to determine the sound transmission loss.

In order to obtain the main purpose of this thesis of evaluating the sound transmission loss, the sound absorption coefficient was determined as preliminary testing to evaluate the proper functioning of the experimental setup. This preliminary testing was done using the two-microphone method, which covered the usage of impedance tube, and all other measurement techniques required for conducting experiments using the four-microphone method. The results of the two-microphone method were satisfactory and encouraging to modify the setup for determining the sound transmission loss. With the proper functioning and validation of the tube using two-microphone method, the tube was modified and experiments were conducted to obtain the results for sound transmission loss of various materials. With the obtained results, it was clearly observed that the aerogel core composite had better sound absorption coefficient that aerogel+honeycomb composite and honeycomb composite up to a frequency of 3 KHz. But in the frequency band of 3000 Hz – 4000 Hz, a reverse patter was observed. While the aerogel core composite was clearly leading the STL curve as compared to other composites. A sample aerogel core alone was also tested and it showed a way better result than the composites. The STL curve of the aerogel core higher than all other materials that were tested, clearly showing high STL of aerogels.

Concepts of engineering acoustics have been used for deriving the equations used for the calculation of the above said acoustic properties. The performance of the tube was determined by performing experiments on materials for which acoustic properties are known and also by using an approximately ideally sound absorbing material. The obtained results were satisfactory. MATLAB programs have been developed for processing of arrays of complex data and subsequent calculation of sound transmission loss and absorption coefficient. The programs involve calculation of complex transfer functions at all frequencies to obtain the desired properties as a function of frequency itself. Although the range of frequencies for human ears is 20 Hz- 20 KHz, the usable range depends upon the diameter of the tube itself, which is 4000 Hz in this case. The tube shows excellent results for higher frequencies i.e., in the range of 400 Hz – 4 KHz.

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APPENDICES

Appendix I: MATLAB program for calculation of Sound Transmission Loss

```
function Output=TB4(Ia,Ib)
% "Ia and Ib are the two input matrices where subscripts ‘a’ and ‘b’ indicate the open end and closed termination conditions, the ith column indicates the input data from the ith number microphone"
% “Xi is the distance of the ith mic from the surface of the sample"
x1=-0.2175;
x2=-0.2065;
x3=0.1775;
x4=0.1885;
% “thickness of sample"
d=0.05334;
"Velocity of the sound"
c=344.989;
% "Density of air"
R=1.21;
“Sampling frequency"
f=1/5e-6;
```

```
% “number of sample points"

L=length(Ia(:,1));

nfft=2^nextpow2(L);

for i=1:4

% “Calculation of transfer function with default Hamming window"

[Ha(:,i),Output(:,1)]=tfestimate(Ia(:,1),Ia(:,i),[],[],nfft/2+1,f,'onesided');

Hb(:,i)=tfestimate(Ib(:,1),Ib(:,i),[],[],nfft/2+1,f,'onesided');

%Ha(:,i)=abs(ha(:,i));

%Hb(:,i)=abs(hb(:,i));

end


% “Calculation Complex amplitudes of the plane wave components”

ap=0.02203*(sqrt(Output(:,1)))/(c*d);

k=2*pi.*Output(:,1)/c-(j*ap);

Aa=j*(Ha(:,1).*exp(j*k*x2)-Ha(:,2).*exp(j*k.*x1))./(2*(sin(k*(x1-x2))));

Ba=j*(Ha(:,2).*exp(-j*k*x1)-Ha(:,1).*exp(-j*k.*x2))./(2*(sin(k*(x1-x2))));

Ca=j*(Ha(:,3).*exp(j*k*x4)-Ha(:,4).*exp(j*k*x3))./(2*(sin(k*(x3-x4))));

Da=j*(Ha(:,4).*exp(-j*k*x3)-Ha(:,3).*exp(-j*k*x4))./(2*(sin(k*(x3-x4))));

Ab=j*(Hb(:,1).*exp(j*k*x2)-Hb(:,2).*exp(j*k*x1))./(2*(sin(k*(x1-x2))));

Bb=j*(Hb(:,2).*exp(-j*k*x1)-Hb(:,1).*exp(-j*k*x2))./(2*(sin(k*(x1-x2))));

Cb=j*(Hb(:,3).*exp(j*k*x4)-Hb(:,4).*exp(j*k*x3))./(2*(sin(k*(x3-x4))));

Db=j*(Hb(:,4).*exp(-j*k*x3)-Hb(:,3).*exp(-j*k*x4))./(2*(sin(k*(x3-x4))));
```

```
%" Calculate the pressures and Particle velocities"
Pa0=(Aa+Ba);
Pad=(Ca.*exp(-j*k*d)+Da.*exp(j*k*d));
Va0=((Aa-Ba)/R/c);
Vad=((Ca.*exp(-j*k*d)-Da.*exp(j*k*d))/R/c);
Pb0=(Ab+Bb);
Pbd=(Cb.*exp(-j*k(:,1).*d)+Db.*exp(j*k(:,1).*d));
Vb0=((Ab-Bb)./R./c);
Vbd=((Cb.*exp(-j*k(:,1)*d)-Db.*exp(j*k(:,1)*d))./R./c);


% "Calculation of elements of transfer matrix"
T11=(Pa0.*Vbd-Pb0.*Vad)./(Pad.*Vbd-Pbd.*Vad);
T12=(-Pa0.*Pbd+Pb0.*Pad)./(Pad.*Vbd-Pbd.*Vad);
T21=(Va0.*Vbd-Vb0.*Vad)./(Pad.*Vbd-Pbd.*Vad);
T22=(-Pbd.*Va0+Pad.*Vb0)./(Pad.*Vbd-Pbd.*Vad);


Output(:,2)=10*log10(0.25*(abs((T11+T12/R/c+R*c*T21+T22)).^2));
plot(Output(:,1),Output(:,2));
title('Output');
xlabel('Frequency (Hz)');
ylabel('Transmission Loss (dB)')
grid;
end
```

Appendix II: Selenium DH 200E driver specifications:

SPECIFICATIONS

Nominal impedance. 8

Minimumimpedance@4,250 H z 6.9

Power handling

Musical Program (w/ xover 2,000 Hz 12 dB / oct) . . . 200 W

Sensitivity

On horn,1W@1m, on axis . . . . 105 dB SPL

Frequency response@-10 dB. . . . . 1,500 to 20,000 Hz

Throat diameter . . . . 25 (1) mm (in)

Diaphragmmaterial . . . . Titanium

Voice coil diameter . . . . 46 (1.8) mm (in)

Re 5.8

Flux density . . . . 1.55 T

Minimumrecommended crossover (12 dB / oct) . . . . 2,000 Hz

Voice coil former material . . . . Polyimide (Kapton)

Voice coil winding length. . . . . 2.8 (9.19) m (ft)

Voice coil winding depth . . . . 2.2 (0.09) mm (in)

Wire temperature coefficient of resistance( ). . . . . 0.00380 1/°C

Specifications to handle normal speech and music program material with 5% maximum acceptable distortion on amplifier, with the recommended passive crossover connected.

Power is calculated taking into account the true RMS voltage at amplifier output along with transducer nominal impedance.

Musical Program= 2 x W RMS.

2 Measured with HM17-25E horn, 1,500 - 8,000 Hz average.

MOUNTING INFORMATION

Magnet material . . . . Bariumferrite

Magnet weight . . . . 429 (15) g (oz)

Magnet diameter x depth . . . . 102 x 14 (4.02 x 0.55) mm(in)

Magnetic assembly weight . . . . 1,200 (2.65) g (lb)

Housing material . . . . Plastic

Housing finish . . . . Black

Magnetic assembly steel finish . . . . Zinc-plated

Voice coil material. . . . . Copper

Volume displaced by driver . . . . 0.4 (0.014) l (ft)

Net weight (1 piece). . . . . 1,350 (2.98) g (lb)

Gross weight (6 pieces per carton). . . . . 8,400 (18.52) g (lb)

Horn connection . . . . Screw-on1 /” - 18 TPI

Connectors . . . . Push terminals

Polarity . . . . Positive voltage applied to the positive term inal

(Red) gives diaphragm motion toward the throat

Appendix III: Microphone Pre-amplifier Specifications

DESCRIPTION

INPUT: 1/4" and female XLR jack for signal input.

OUTPUT: 1/4" and male XLR jack for signal output. May be connected to stereo headphones for personal monitoring.

POWER: Sleeve-positive barrel jack for connection to the AC adapter.

GAIN: Controls the amount of signal increase from the MP13.

CLIP: LED for indicating circuit clipping. Indicates distortion is possible.

PHANTOM POWER: When pressed in, applies +36 volts of phantom powering to the XLR input.

PWR: LED indicating power is applied to the MP13.

SPECIFICATIONS

Input Impedance: 10K W bal, unbalanced

Max Input Level: +4 dBV XLR bal., +10 dBV unbal.

Max Gain: 50 dB XLR, 44 dB 1/4"

Phase Shift: <10 deg. 20Hz - 20kHz

EIN: 129 dB max.

THD: <.01%

IMD (SMPTE): <.01%

Power: 12 VDC adapter

Weight: 1 lbs (.45 kg)

Size: 3" x 4" x 1.25" (8 x 10 x 3.2 cm)

Appendix IV: Random Noise Generator:

This instrument generates wide-band noise of uniform spectrum level, particularly useful for noise and vibration testing in electrical and mechanical systems. The noise output of a gas-discharged tube is amplified and shaped with low-pass filters to provide ranges to 20 KHz, 500 KHz and 5 MHz.

The output level is controlled by a continous attenuator by a 4-step attenuator of 20 dB per step and is metered from over 3volts to 30 microvolts. When attenuator is used, the output impedance remains essentially constant as the output level is adjusted.

Use the 1390 –B as a broad-band signal source for:

- Frequency response – drive device under test with 1390-B and analyse output with any of several analyzers, manually or with the GR 1521-B Graphic Level Recorder.
- Intermodulation and cross-talk tests.
- Simulation of telephone-line noise.
- Measurements on servo amplifiers.
- Noise interference tests on radar.
- Determining meter response characteristics.
- Setting transmission levels in communication circuits.
- Statistical demonstrations in classroom and lab.

Acoustic Measurements:

- Frequency response.
- Reverberation – use 1390-B with a GR analyzer as a source of narrow-band noise.

- Sound attenuation of ducts, walls, panels or floors.
- Acoustical properties of materials.
- Room acoustics.

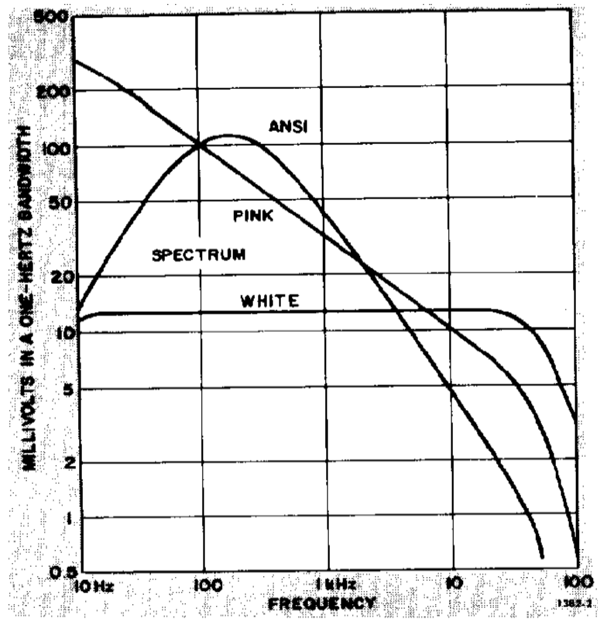
Or use with amplifiers to drive:

- A loudspeaker for structural fatigue tests in high-level acoustic fields.
- A vibration shake-table.

Upper frequency bandwidth (at 3dB point) can be set to 20 KHz, 500 KHz, or 5MHz.

Ideal for random vibration testing, and audio/noise filtering tests.

- Broad band noise signal: 5Hz to 5MHz uniform spectrum level.
- Useful in noise and vibration testing of mechanical and electrical systems.
- Upper frequency cutoff can be set at: 20 kHz, 500 kHz, or 5MHz.
- Output level is controlled by a 4-step attenuator with 20 dB steps.
- Continuously variable control providing a calibrated, metered output from <30 μV to 3 V.
- Audio spectrum level uniformity is rated at ±1 dB.
- The optional 1521B Level Recorder can be attached to give continuous records of level versus frequency.



Spectrum of white Noise

VITA

Kalyan Chakravarthy Vengala

Candidate for the Degree of

Master of Science

Thesis: BUILDING A MODIFIED IMPEDANCE TUBE FOR MEASUREMENT OF SOUND TRANSMISSION LOSS AND ABSORPTION COEFFICIENTS OF POLYMER CROSSLINKED AEROGEL CORE COMPOSITES.

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Title of Study: BUILDING A MODIFIED IMPEDANCE TUBE FOR MEASUREMENT OF SOUND TRANSMISSION LOSS AND ABSORPTION COEFFICIENTS OF POLYMER CROSS-LINKED AEROGEL CORE COMPOSITES

Pages in Study: 63 Candidate for the Degree of Master of Science

Major Field: Mechanical and Aerospace Engineering

Scope and Method of Study:

This work describes in detail the development of the conventional impedance tube and also the modification made to determine the sound transmission loss. Concepts of engineering acoustics have been used for deriving the equations used for the calculation of the above said acoustic properties. The performance of the tube was determined by performing experiments on materials for which acoustic properties are known and also by using an approximately ideally sound absorbing material. An impedance tube is developed such that it can evaluate the sound-insulation properties viz., sound transmission loss and absorption coefficient of a given material, upon a slight modification.

Findings and Conclusions:

A conventional impedance tube is slightly modified by placing two additional microphones at the other end of the sample, the receiving end, for the evaluation of the sound transmission loss whereas the conventional two microphone method is used to determine the absorption coefficient. MATLAB program have developed for processing of arrays of complex data and subsequent calculation of sound transmission loss and absorption coefficient. The programs involve calculation of complex transfer functions at all frequencies to obtain the desired properties as a function of frequency itself. Experiments are conducted with the setup configurations as required for the determination of specific property and the input sound source is switched 'on'. The complex signal data received from the microphones is fed to the computer to process the raw data, yielding the required result as a plot of the property against the working frequency range. Although the range of frequencies for human ears is 20 Hz- 20 KHz, the usable range depends upon the diameter of the tube itself, which is 4000 Hz in this case.

ADVISER'S APPROVAL: Dr. Hongbing Lu