Extension of frequency range of the eight-microphone method in normal-incidence sound absorption coefficient measurement by eliminating influence of the first radial mode

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Abstract: The authors have previously proposed the normal-incidence absorption coefficient measurement method using multiple microphones which enables a measurement in the frequency range beyond the cut-on frequencies of the higher order acoustic modes in an impedance measurement tube. The maximum measurement frequency of the method is determined by the cut-on frequency of the first radial mode, \((0, 1)\) mode, in the cylindrical tube. In this study, in order to extend the measurement frequency range more, the method which can reduce the factor of the \((0, 1)\) mode in the sensing signals is investigated. The method consists of two techniques. The first one is a microphone placement way to make the \((0, 1)\) mode signal undetectable by placing the acoustic centers of the microphones at a node of the mode. The second one is related to a sound source structure not to excite the \((0, 1)\) mode. Using these techniques, the normal-incidence absorption coefficient can be evaluated under the cut-on frequency of the \((4, 0)\) mode, which is about 2.89 times higher than that of the conventional two-microphone transfer function method. The results of numerical simulations and experiments are shown to prove the validity and the feasibility of the proposed method.

Keywords: Normal-incidence sound absorption coefficient, Acoustic impedance tube, Transfer-function method, Higher-order acoustic mode, Radial mode

1. INTRODUCTION

The frequency range to measure the normal-incidence sound absorption coefficient of materials using an acoustic impedance tube [1] is limited by the diameter of the tube because the measurement must satisfy the condition that only a normal-propagating wave exists in the tube. Above the cut-on frequencies of the higher order acoustic modes, not only the normal-propagating wave but also obliquely-propagating waves can exist, so that the sound field is too complicated to measure the normal-incidence absorption coefficient by the two-microphone transfer function method, which is commonly used in evaluation of the sound absorption property of materials. Hence, the authors proposed a simple practical method to measure the normal-incidence sound absorption coefficient in the frequency range where the waves of the \((0, 0)\), \((1, 0)\), and \((2, 0)\) modes can propagate using eight microphones under an assumption that a large amount of scattering on the specimen surface does not exist [2–4]. This method is called the eight-microphone method in this study. In order to extract the normal propagating wave factor through a cross section of a circular cylindrical tube, four microphones are located with one in each quarter of the circumference and their signals are summed. Then, the normal-incidence absorption coefficient is calculated from the frequency response function between the normal-propagating factors of two cross sections, which are placed at a prescribed distance. Figure 1 shows the experimental results of normal-incidence absorption coefficients of a 25-mm-thick melamine resin foam by the two-microphone transfer function method and the previously proposed eight-microphone method [2,3]. It is seen that by using the eight-microphone method, the measured normal-incidence absorption coefficient agrees well with the result measured with the small tube \((D = 29\text{ mm})\), whereas the two-microphone method cannot measure correctly above the cut-on frequency of the \((1, 0)\) mode.
mode \( f_{c1,0} \). The maximum measurement frequency of the eight-microphone method is determined by the cut-on frequency of the first radial mode, \((0, 1)\) mode, because the \((0, 1)\) mode has the circular nodal line in the radial direction and summing the signals of the microphones cannot eliminate the signal of the \((0, 1)\) mode.

In this study, in order to extend the measurement frequency range more, the method which the factor of the \((0, 1)\) mode in the sensing signals is reduced is proposed. The method consists of two ideas. The first one is related to a placement way of microphones, the acoustic centers of which are matched to the node of the \((0, 1)\) mode. This can make the \((0, 1)\) mode signals undetected. The second one is that the structure of the sound source is designed not to generate the \((0, 1)\) mode. This can improve the robustness of the proposed method. Using these techniques, the normal-incidence absorption coefficient can be measured in the frequency range where the waves of the \((0, 0), (1, 0), (2, 0), (0, 1),\) and \((3, 0)\) modes can propagate. The maximum measurement frequency increases up to the cut-on frequency of the \((4, 0)\) mode \( f_{c4,0} \), which is about 2.89 times higher than that of the conventional two-microphone transfer function method. The results of numerical simulations and the experiments are shown to prove the validity and the feasibility of the proposed method.

2. THEORY

2.1. Placement of the Microphones

In this study, a cylindrical tube with a radius \( R \) (Fig. 2) is considered. The sound pressure field in the tube is expressed as \([2]\)

\[
p(r_T, \theta, z, t) = \sum_{m=0}^{\infty} \sum_{n=0}^{\infty} \left( \Psi_{mn}^\sigma(r_T, \theta)(A_{mn}^\sigma e^{-jk_{c(mn)}z} + B_{mn}^\sigma e^{+jk_{c(mn)}z}) + \Psi_{mn}^\tau(r_T, \theta)(A_{mn}^\tau e^{-jk_{c(mn)}z} + B_{mn}^\tau e^{+jk_{c(mn)}z}) \right)e^{int} ,
\]

where \( r_T \) denotes the distance from the center of the circular cross section and \( \theta \) is the angle from the \( x \)-axis. \( \Psi_{mn}^\sigma, \Psi_{mn}^\tau \) are acoustic modal functions as follows:

\[
\Psi_{mn}^\sigma = C_{mn}J_m(k_{c(mn)}r_T)e^{-jnt} ,
\]

\[
\Psi_{mn}^\tau = C_{mn}J_m(k_{c(mn)}r_T)e^{jnt} .
\]

\( m \) and \( n \) denote the acoustic modal order in the circumferential direction and in the radial direction, respectively. \( J_m \) is the Bessel function of the first kind and \( C_{mn} \) is normalization constant as following equations is satisfied due to the orthogonality among the acoustical modes.

\[
\int_S \Psi_{mn}^\sigma \Psi_{pq}^\tau * dS = S \quad (\sigma = \tau, m = p, n = q) ,
\]

and

\[
\int_S \Psi_{mn}^\sigma \Psi_{pq}^\tau * dS = 0 \quad (\sigma \neq \tau \text{ or } m \neq p \text{ or } n \neq q) .
\]

where \( S \) is a cross-sectional area of the tube. \( A_{mn}^\sigma, B_{mn}^\sigma, A_{mn}^\tau, \) and \( B_{mn}^\tau \) are the amplitudes of the waves. \( k_{c(mn)} \) is the wave number of the \((m, n)\) mode in the \( z \)-direction:

\[
k_{c(mn)} = (k_0^2 - k_{c(mn)}^2)^{1/2} ,
\]

where \( k_0 = \omega/c \) (\( c \): sound speed), and \( k_{c(mn)} \) is the wave number in the cross section of the tube and satisfies the boundary condition at the inner surface of the tube. \( k_{c(mn)}R \) that satisfies the boundary condition is expressed as \( \lambda_{mn} \). Figure 3 shows each acoustic mode and the corresponding value of \( \lambda_{mn} \). The minimum frequency (cut-on frequency) at which the \((m, n)\) mode can propagate in a tube can be expressed by the following equation.
Since from Eq. (1) as follows: in the authors’ previous papers [2,3]. That can be derived signals at these four measurement points is considered as the normal-propagating factor, the sum of the pressure factors. Then, a frequency response function for the normal-propagating factor. The normal-propagating factors are extracted on two cross sections, A and B in Fig. 2. The proposed method considers the frequency range below the cut-on frequency of the (4,0) mode. However, in practice, it is hard to have the acoustic centers of the microphones exactly located on the nodal line of the (0,1) mode because the actual impedance tube can have slight errors in size and uncertainty of the boundary condition. Furthermore, a microphone cannot measure sound pressure at a point in a precise sense, because it senses the average pressure on the surface area of the diaphragm and in general, the acoustic center of a microphone is not identical with the surface of the microphone diaphragm [5].

Unless the (0,1) mode is excited, of course, the effect of the (0,1) mode is not detected and does not affect the measured absorption coefficient. The primary cause of exciting the (0,1) mode may actually be the sound source to radiate sound into the tube. If the sound source does not excite the (0,1) mode, it is considered that the effect of the (0,1) mode can substantially be reduced. Hence, this study considers the source structure not to excite the (0,1) mode. In Fig. 4, the circular-shaped sound source which is proposed in this paper is shown. This circular-shaped sound source can be realized by placing a partition plate with a circular-shaped slit just in front of a loudspeaker unit. The air in the slit vibrates uniformly when the sound source structure has perfect symmetry and the width of the slit is sufficiently smaller than the wavelength. It is

\[ f_{cm,n} = \frac{\lambda_{mn} c}{2\pi R} \]  

(6)

In the previously proposed method [2-4], four measurement points, indicated as \( r_1, r_2, r_3, \) and \( r_4 \) in Fig. 2, are considered at one measurement cross section. These points are located on the inner surface of the tube at every quarter of the perimeter. On the other hand, in the method proposed in this paper, measurement points are indicated as \( r_1', r_2', r_3', \) and \( r_4' \) which are on the nodal line of the (0,1) mode. When the radius of the nodal line of the (0,1) mode is assumed to be \( r_N \), the following equation is satisfied.

\[ J_0(k_{r(0,1)} r_N) = 0 \]  

(7)

Since \( k_{r(0,1)} R = \lambda_{0,1} \approx 3.83 \), \( r_N \) is derived as \( r_N \approx 0.627 R \).

The proposed method considers the frequency range below \( f_{\text{cut},0} \) in which the waves of the (0,0), (1,0), (2,0), (0,1) and (3,0) modes can propagate. In order to extract the normal-propagating factor, the sum of the pressure signals at these four measurement points is considered as in the authors’ previous papers [2,3]. That can be derived from Eq. (1) as follows:

\[ p_s = p(r_1) + p(r_2') + p(r_3') + p(r_4') = 4(A_00 e^{-j k_{00} z} + B_00 e^{+j k_{00} z}) e^{j k_{00} R}. \]  

(8)

Summing the signals cancels the signals due to the circumferential modes, the (1,0), (2,0) and (3,0) modes. And the signal due to the (0,1) mode cannot be sensed because of the placement of the measurement points at the node of the (0,1) mode. Then, these procedures yield the normal-propagating factor. The normal-propagating factors are extracted on two cross sections, A and B in Fig. 2. Then, a frequency response function for the normal-propagating factors between two cross sections can be calculated as \( H_{AB} = P_{AB} / P_{AA} \), where \( P_{AA} \) and \( P_{AB} \) are the frequency spectrums of the sum of the measurement signals at cross sections A and B, respectively. Using this frequency response function, the normal-incidence sound absorption coefficient can be obtained by the same equations as those in the two-microphone transfer function method, like our previous papers [2,3].

2.2. Sound Source Not to Excite the (0,1) Mode

As shown in the previous section, by placing the measurement points on the nodal line of the (0,1) mode and the summing the measured signals, the normal-incidence sound absorption coefficient can be obtained in the frequency range below the cut-on frequency of the (4,0) mode. However, in practice, it is hard to have the acoustic centers of the microphones exactly located on the nodal line of the (0,1) mode because the actual impedance tube can have slight errors in size and uncertainty of the boundary condition. Furthermore, a microphone cannot measure sound pressure at a point in a precise sense, because it senses the average pressure on the surface area of the diaphragm and in general, the acoustic center of a microphone is not identical with the surface of the microphone diaphragm [5].

**Fig. 3** Acoustic modes in the tube.

**Fig. 4** Proposed circular-shaped sound source.
the source is assumed to be $u_z = \frac{1}{\rho \omega} \frac{\partial p}{\partial z}$, derived as

$$u_z = \frac{1}{\rho \omega} \sum_{m=0}^{\infty} \sum_{n=0}^{\infty} k_{m,n} \left[ B_{m,n}^* \Psi_n^* (r_T, \theta) + B_{m,n} \Psi_n (r_T, \theta) \right] e^{j k_{m,n} z + j \omega t}.$$  

Now the velocity distribution on the surface of the source is assumed to be

$$u_z = U_s (r_T, \theta) e^{j \omega t}.$$  

Since the vibration velocity at the surface of the sound source is the same as the particle velocity of the air at the tube end ($z = L_c$), the following equation is obtained from Eqs. (11) and (12).

$$U_s (r_T, \theta) = \frac{1}{\rho \omega} \sum_{m=0}^{\infty} \sum_{n=0}^{\infty} k_{m,n} \left[ B_{m,n}^* \Psi_n^* (r_T, \theta) + B_{m,n} \Psi_n (r_T, \theta) \right] e^{j k_{m,n} z + j \omega t}.$$  

Multiplying $\Psi_{p,q}^*$ at both sides of this equation and integrating over the cross-section $S$, the amplitude of the $(p,q)$ mode is obtained as follows:

$$B_{p,q}^* = \frac{\rho c}{S \kappa_{p,q} e_{L_c}} \frac{1}{\left[ 1 - \left( \frac{k_{p,q}}{k_0} \right)^2 \right]} \int_S U_s \Psi_{p,q}^* dS.$$  

From this equation, the ratio of the amplitude of $(0,1)$ mode to that of $(0,0)$ mode is

$$\frac{B_{01}}{B_{00}} = \frac{C_{01}}{\sqrt{1 - \left( \frac{k_{01}}{k_0} \right)^2}} \int_0^1 U_s J_0 (\lambda_{01} r') r' dr'$$  

where $r' = r_T / R$.

As an index indicating the degree of excitation in the $(0,1)$ mode with respect to the normal incidence component $(0,0)$ mode, the following $R_{ex01}$ is defined, excluding the frequency-dependent component.

$$R_{ex01} = \left| \int_0^1 U_s J_0 (\lambda_{01} r') r' dr' \right|$$  

$R_{ex01}$ is called a $(0,1)$ mode excitation factor in this study.

In the case of the circular-shaped slit, $R_{ex01}$ can be analytically obtained as follows:

$$R_{ex01} = \frac{2C_{01}}{\lambda_{01}} - \frac{r_2}{R} \left( \frac{\lambda_{01}}{R} \right)^2 - \frac{r_1}{R} \left( \frac{\lambda_{01}}{R} \right)^2.$$  

where $r_1$ and $r_2$ are the radii of the inner and outer edges of the circular-shaped slit, respectively.

$r_A$ is the radius of the center line of the slit width, defined as $r_A = (r_1 + r_2)/2$. $R_{ex01}$ was calculated while changing the value of $r_A/R$. The calculations were performed when the slit widths ($w = r_2 - r_1)$ were 2 mm, 4 mm, 8 mm, and 16 mm. The results for the case of 4 mm and 16 mm are shown in Fig. 5. For comparison, the results in the case when sound is directly radiated from the piston plate having the radius $r_A/R$ are also shown. According to the results, in the case of the piston plate, the value of $R_{ex01}$ is generally large, and for the smaller value of $r_A/R$, the $(0,1)$ mode is more excited. On the other hand, in the case of the 4 mm-width circular-shaped slit, it can be seen that the value of $R_{ex01}$ is less than 0.2 around 0.59–0.67, and $R_{ex01}$ has a dip value when $r_A/R$ is around 0.63. The node position of the $(0,1)$ mode is equivalent to $r_A/R = 0.6276$. Hence, it is confirmed that the $(0,1)$ mode is not excited and only the $(0,0)$ mode is excited when the center line of the circular-shaped slit is identical to the nodal line of the $(0,1)$ mode. In the case that the slit width is 16 mm, the

![Fig. 5 (0,1) mode excitation factor of the circular-shaped sound source.](image-url)
value of $r_A/R$ where $R_{ex01}$ has a dip becomes lower than that in the case of 4 mm-width slit. But the amount of shift is small. It is found from Eq. (14) that the width of the slit affects the amplitude of the (0, 0) mode. Hence, the width is needed to be determined to have enough sound in the tube.

### 2.3. Proposed Structure of the Impedance Tube

Figure 6 shows an outline of an impedance tube employing the above two ideas to eliminate the influence of the (0, 1) mode. A total of eight microphones are inserted through the tube wall so that the acoustic centers of the microphones can match the nodal line of the (0, 1) mode. The partition plate with a circular-shaped slit is placed in front of the loudspeaker. In order to suppress the resonance in the tube, sound absorbing material should be installed at the source side in the tube [1]. However, the sound absorbing material changes the acoustic modes in the tube due to changes in boundary conditions at the tube wall and wave attenuation in the material. This can ruin the function of the circular-shaped slit. Hence, the circular-shaped slit is extended into the tube using two pipes of different sizes, which is called a double-pipe-slit in this study. The sound generated by the loudspeaker goes through the gap of the double-pipe-slit into the tube. The space around the double-pipe are filled with sound absorbing materials. This source structure does not affect the acoustic modes in the tube, because the sound wave going through the slit does not directly enter the absorbing materials. As a result, sound is radiated from the nodal line of the (0, 1) mode into the inside of the tube, and sound reflected from the test specimen is absorbed by the sound absorbing materials.

By the way, in experiments, the structure which includes the inner pipe, etc. is supported by some bolts to maintain the gap of the double-pipe-slit. Since the diameters of the bolts are small and the number is small, the influence of the bolts to sound waves inside the double-pipe-slit can be assumed to be ignored.

### 3. NUMERICAL SIMULATION

#### 3.1. Numerical Simulation Model

In order to confirm the validity of the proposed method, numerical simulations were conducted. Figure 7 shows the calculation conditions. The calculation model was based on the experimental conditions stated later. The finite element method (FEM) was used with LMS Virtual.Lab Acoustics in the calculation. The diameter and the length of the tube are 100 mm and 410 mm, respectively. It was assumed that a piston of $\phi60$ mm vibrates at the end surface of the tube. The measurement points corresponding to the microphones were defined on two cross-sections of 50 mm and 70 mm from the specimen surface. The partition plate was defined 10 mm away from the sound source, and a double-pipe-slit (slit length: 50 mm) was modeled. In order to prevent the sound reflection on the source end, sound absorbing materials were defined. The sound absorbing materials were modeled by the Johnson-Champoux-Allard model. The vibration of the frame of the sound absorbing materials was assumed to be ignorable. The calculations were performed for the case when the measurement points are at the nodal line of the (0, 1) mode, when they are shifted 1.0 mm outward (+1.0 mm), and when they are shifted

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**Fig. 6** Schematic view of the proposed impedance tube.

**Fig. 7** Calculation conditions of the proposed tube.
1.0 mm inward (−1.0 mm). For comparison, calculations were also performed for the cases without the double-pipe-slit.

3.2. Simulation Results

Figures 8 and 9 show the calculated normal-incidence absorption coefficients of a 25-mm-thick melamine resin foam and a rigid plate, respectively. In each figure, (a) shows the results when the double-pipe-slit is used, and (b) shows the results when the sound is directly radiated from the vibrating piston without using the slit. The theoretical maximum measurable frequencies are $f_{c,1,0}$ (2,029 Hz) for the 2-microphone transfer function method, $f_{c,0,1}$ (4,213 Hz) for the previously proposed 8-microphone method, and $f_{c,4,0}$ (5,864 Hz) for the 8-microphone method which are proposed in this study. From the calculation results, it is found that for both the specimens, when not using the double-pipe-slit, the sound absorption coefficient cannot be calculated correctly over about 4,200 Hz, which is close to $f_{c,0,1}$, whereas smooth sound absorption curves are obtained up to about 5,800 Hz when using the slit. In addition, without the double-pipe-slit, the disturbance of the absorption coefficient is large in the frequency range above about 4,200 Hz when the microphone position is displaced by ±1.0 mm from the nodal line. These indicate that if the double-pipe-slit is not used, the sound absorption coefficient measurement is very sensitive to the microphone position. On the other hand, when the slit is used, it can be seen that the calculation results of the sound absorption coefficient are not significantly affected even when the microphone position is slightly shifted. These results show that it is essential both to match the microphone position with the nodal position of the (0,1) mode and to use the sound source structure which does not generate the (0,1) mode.

3.3. Effect of the Existence of Microphones in the Sound Field

To match the acoustic centers of the microphones and the nodal line of the (0,1) mode, the microphones have to
be inserted into the impedance measurement tube through the tube wall. Then, the sound field in the tube may be disturbed by the existence of the microphones themselves. Hence, the effects of the microphone existence on a measurement result were investigated by the numerical analysis. The shapes of the microphones inserted into the tube were modeled (four in each of two cross-sections of 50 mm and 70 mm from the specimen surface). A measurement point was defined at each microphone tip. Figure 10 shows the calculation results of the normal incidence sound absorption coefficient when a 25-mm-thick melamine resin foam is used as the test specimen.

The calculations were performed for 1/2, 1/4, 1/6, and 1/8-inch microphones. In the figure, the result without considering the existence of microphones is also shown (broken line). In the case of the 1/2-inch microphone, the calculated absorption coefficient is greatly disturbed around 4,200 Hz and is slightly lower or higher than the result in the case without considering the microphone existence in the whole frequency region. In the case of the 1/4-inch microphone, there is not much large error. The influence of the existence of the microphone is smaller as the microphone size is smaller, and is considered almost negligible when using 1/6 and 1/8-inch microphones.

4. EXPERIMENT

4.1. Experimental Setup

In order to verify the validity and the feasibility of the proposed method, experiments were conducted. Figure 11 shows photos of the experimental setup. The acoustic impedance tube is made from acrylic. The inner diameter $D$ is 100 mm and the length is 410 mm. The upper frequency limit which is determined by the microphone distance (20 mm) is 7,796 Hz (in the case of $c = 346.5$ m/s). Eight 1/4-inch microphones were used and inserted into the tube so that the acoustic centers could coincide with the $(0, 1)$ mode node. Before measuring the normal-incidence sound absorption coefficient, mismatch corrections among microphones were performed.

4.2. Experimental Results

Figure 12 shows the measurement results of the normal-incidence sound absorption coefficient of a melamine resin foam with a thickness of 25 mm. Figure 13 shows the measurement results of a reflecting plate (corresponding to a rigid wall). In each figure, (a) shows the results when the double-pipe-slit was used, and (b) shows the results when the sound was directly incident from the cone-shaped speaker unit without using the slit. For comparison, Fig. 12 also shows the results of measurement using a 29-mm-diameter acoustic tube (B&K 4206).

As in the calculation results, when the slit is not used, the sound absorption coefficients are significantly disturbed in the frequency range of about 4,200 Hz or more, whereas when the slit is used the sound absorption coefficient curves are smooth up to around 5,600 Hz. The results of the proposed method agree well with the results measured with the small-diameter tube, and it is seen that the reasonable measurement results were obtained.

In addition, when the slit is used, even if the microphone position is shifted by $\pm 1.0$ mm, there is no significant change similarly to the calculation results. In other words, experiments confirmed that the proposed method has robustness against microphone position shift. Since the acoustic center of a microphone does not usually coincide with the diaphragm position, it is practically difficult to completely adjust the acoustic center at the nodal line of the $(0, 1)$ mode. From a practical point of view, it is considered that the robustness to the uncertainty of the microphone position is useful.
Next, in order to compare the results of the proposed method with the conventional methods, the experimental results of the normal incidence sound absorption coefficient of a 25-mm-thick melamine resin foam measured by the different methods are shown in Fig. 14. From these results it is confirmed that the result for each measurement method agrees well with the measurement result by the small-diameter tube in the frequency ranges below $f_{c1,0}$ for the 2-microphone method, $f_{c0,1}$ for the previously proposed 8-microphone method, and $f_{c4,0}$ for the newly proposed 8-microphone method. These results prove the effectiveness of the 8-microphone method proposed in this study.

5. CONCLUSIONS

In order to have the measurement frequency of the eight-microphone method which was previously proposed by the authors higher, the method which the influence of the (0, 1) mode is eliminated has been considered. Firstly,
the microphone placement method to make the \((0, 1)\) mode undetectable was proposed. The acoustic centers of the microphones are placed at the nodal line of the \((0, 1)\) mode. Secondly, the circular-shaped sound source which cannot generate the \((0, 1)\) mode in the tube was proposed. With the numerical simulations, it was confirmed that the proposed method can work as expected, considering disturbance effect of the microphone existence in the tube. Finally, the experiments were conducted to show the feasibility of the proposed method. Numerical simulation results and the experimental results proved that in the frequency range where the \((0, 0)\), \((1, 0)\), \((2, 0)\), \((0, 1)\) and \((3, 0)\) modes propagate, the normal-incidence sound absorption coefficient can be measured using the proposed method.

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