Scientia Iranica A (2012) 19 (4), 1005–1012

Sharif University of Technology
Scientia Iranica
Transactions A: Civil Engineering
www.sciencedirect.com

Assessment of a practical technique for active control of sound using microphone and speaker

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Received 15 October 2011; revised 25 February 2012; accepted 15 May 2012

KEYWORDS
Active control;
Sound;
Loudspeaker;
Microphone;
Pure tone.

Abstract In this analytical study, it has been desired to develop a practical and simple control mechanism to control, at a given point and its neighborhood, the sound arriving from a distant source, assuming that a primary pure-tone sound pressure is propagated from a relatively far distance. The control model consists of a microphone as a sensor for measuring the sound pressure and a loud speaker for applying the control force. Corresponding equations have been developed to determine an optimum control force, and afterwards a parametric study on the factors affecting the control results has been performed. The results show that the control system can significantly reduce low frequency sound pressure in the region near the target point. The results also demonstrate less effectiveness in controlling high frequency sound pressures. Moreover, the larger the distance of the loudspeaker to the given controlled point is, the wider the controlled area will be. Also, it was found that though the distance between the sensor and the actuator does not have any effect on the size of the area which can be controlled, such distance is of greatest importance in that the available time to control increases directly by increasing the distance.

1. Introduction

Unwanted sound from distant sources, such as airplanes, cars, construction activities and crowds in public places, is becoming more and more a noticeable problem in society, and has already been considered an environmental problem in industrial society. Hence, it is desired to attenuate such unwanted sounds.

Sound is a mechanical wave that is an oscillation of pressure transmitted through solid, liquid, or gas. In the area of sound control, only sonic waves, composed of frequencies within the range of hearing and of a level sufficiently strong to be heard, are considered as sound. When the amplitude of sound is high, it becomes annoying and, should it pass higher limits, it can even damage the hearing system. Low amplitude sound is found nearly everywhere, such as in conference halls, schools, streets, train stations and airports, while higher amplitude sound may be experienced in factories and industrial sites.

Methods of sound control have a long history, which consists of, but is not limited to, the use of sound absorbing materials, sound insulating elements, vibration insulators, acoustic walls and ceiling panels. As most of these passive control methods are inadequate, the active control of sound was suggested before the mid-20th century. However, the lack of micro-processors delayed its implementation [1]. In 1988, the possibility of active cancellation of sound in a pure tone diffuse sound field was investigated by Elliott et al. [2]. Two years later, some experiments on active control of the transmission of sound through a clamped rectangular plate, using point actuators, were performed by Thomas et al. [3]. The results showed the feasibility of utilizing point actuators for controlling transmitted sound through a plate, even in cases where the plate is in resonance mode. Afterwards, the implementation of piezoelectric actuators in the active control of sound transmission through rectangular single-panel and double-panel plates was theoretically studied by Fuller et al. [4,5]. In 1995, Elliot et al. [6,7] worked on the cancellation of pressure in a diffuse sound field. Arrays of discrete actuators were used by Johnson and Elliot [8] with the aim of controlling sound radiation from vibrating surfaces. In recent years, more

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Peer review under responsibility of Sharif University of Technology.

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doi:10.1016/j.scient.2012.06.011

Nomenclature

a	sound source location
b	microphone location
c	loudspeaker location
d	control point
e	a point near control point
C	sound speed
P_s	sound source power
I	sound intensity
t	time
Δt	time of processing
L_1	distance from microphone to loudspeaker
L_2	distance from loudspeaker to control point
R	radial distance from sound source
R_{ab}	distance of b to original sound source
R_{ad}	distance of d to original sound source
R_{cd}	distance of d to control sound source
R_{ce}	distance of e to control sound source
r	distance of point e from point d
x	distance between two points
f	frequency of original sound
f'	frequency of control sound
w	angular frequency of sound source
w'	angular frequency of control sound
λ	wavelength of original sound
λ'	wavelength of control sound
ΔP	peak amplitude of original sound pressure at point d
$\Delta P'$	peak amplitude of control sound pressure at point d
$\Delta P_{a,b}$	original sound pressure at point b
$\Delta P_{a,d}$	original sound pressure at point d
$\Delta P_{a,e}$	original sound pressure at point e
$\Delta P_{c,d}$	control sound pressure at point d
$\Delta P_{c,e}$	control sound pressure at point e
ΔP_d	sound pressure at point d after control
ΔP_e	sound pressure at point e after control
ϕ'	phase difference
ϕ	phase difference between control sound and original sound at point d
$\Delta\phi$	phase difference between control sound at point e to point d
$\|\|$	absolute

research on smart materials and structures for active control of sound has been reported. In 2004, Gardonio and Elliott applied techniques for active structural control of sound on smart panels [9]. Alujevic et al., in 2008, proposed smart double panels for sound control by implementing active damping units [10].

The present paper studies the active control of sound, transmitted from a distant source through a free field, using a microphone as the sensor and a loudspeaker as the source for controlling the sound. As the source of the original sound is considered as a point which generates periodic sound waves through a free field, the sound intensity is proportional to the square of the sound pressure; also it falls inversely proportional to the square of the distance [11]. The main objective has been to alleviate the sound propagated from airplanes, trains, vehicle engines, or other sources in an ambient environment to be controlled, such as a private area, where a sensitive measurement device may be placed, such as an emergency outdoor telephone, etc. This method seems to be both technologically and economically feasible.

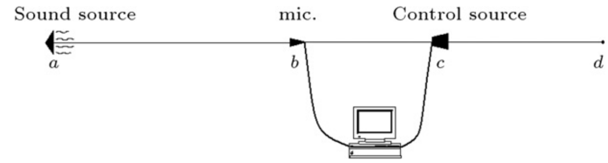

Figure 1: Active control system.

2. Active control system

Details of the problem and the active control system used are schematically shown in Figure 1. The sound source is located at point a. It is desired to control and reduce the amplitude of the sound wave pressure which arrives at point d. A sensor, which is capable of measuring the intensity of the sound wave pressure, is placed at point b. In this study, a precise microphone with a nearly uniform frequency response has been considered as the sensor. The data provided by the sensor is then delivered to the controller, which is a computer software designed to calculate the control excitation. The controller should be designed to:

1. Identify, based on the measurement by the sensor, and in order to calculate the control excitation, the characteristics of the sound wave generated at the source point, a, as it arrives at point d.
2. Determine the amplitude and frequency of the controlling sound pressure needed to mitigate the sound at point d; and
3. Determine the best time for implementation of the control sound based on the distances between points b, c and d.

After the characteristics of the controlling sound have been identified, the controller determines the controlling sound, which should be generated at point c. Determination of the amplitude and frequency of the sound pressure, as well as the best time to issue the control wave, depend on the control criteria. It has been desired in this research to study the effect of different parameters that can affect control results, including distances, sound properties and loud speaker properties, as will be explained later. In this paper, a loudspeaker has been considered as the controlling actuator. The interference of original and controlling sound pressures provides the final sound pressure at d and its surroundings.

3. Feasibility of the system

There is a time delay between the instance when the sound wave arrives at point b, and is sensed by the microphone, until the controlling sound wave is issued at point c by the loudspeaker. One of the conditions for the control system to be feasible and capable of using the data collected by the sensor at b, to determine the control sound at d, is:

$$L_1 \geq C \cdot \Delta t \tag{1}$$

where Δt = time delay, L_1 = distance between points b and c and C = speed of sound in the medium. It may be possible to design predictive controllers that can determine control excitation based on predicting future sound wave pressure that is expected to arrive. Such predictive control strategies have been tested successfully in other disciplines, including structural control engineering, where the objective has been to mitigate the unwanted response of structures subjected to different dynamic loadings, like earthquakes and wind.

However, predictive control requires a thorough understanding and formulation of the system to be controlled as well as the control system itself. The control algorithm used in this study has not been predictive, in that, no prediction of sound induced excitation has been used in estimation of the control force.

The time delay, Δt, comes from the main following sources: (1) the time required for sensing, collecting and processing of the sound wave characteristics, and (2), the time needed for the computation and determination of the controlling sound by the controller.

The microphone is a transducer which senses sound pressure and changes mechanical vibrations into electrical voltage signals. The electrical voltage issued by the microphone throughout the time is transmitted to a processor, which collects and processes the voltage signals in order to identify the characteristics of the sound. The control software then uses the processed data to determine the required control electrical voltage to be sent to the loudspeaker, which generates the controlling sound. The controlling sound interacts with external sound and alleviates it to a desired level.

The speed of sound in the air depends on air density and temperature, which is assumed to be 340 m/s under normal conditions. Also, noticing that digital information travels at the order of light speed in copper wires and in fiber optic cables, and since the speed of light is about 300,000 km/s, the time delay, because of sensing and transferring data to the computer, is so short, it is practically negligible. However, the time required for processing signals and calculating the control signal is much larger and should not be ignored. It has been assumed that a time delay of $\Delta t = 0.01$ s in Eq. (1) is large enough to include this delay effect. Then, the minimum length, L_1, for the control system to be feasible can be calculated from:

$$(L_1)_{\min} = C\Delta t = 340 \text{ m/s} \times 0.01 \text{ s} = 3.4 \text{ m}. \quad (2)$$

Hence, in this study, a minimum distance of 3.4 m has been considered between the two points where the microphone and the loud speaker are placed.

4. Analysis

Following the Fourier theorem, any sound wave can be decomposed into a number of pure tones (sine waves) of different periods, and a sound wave can be obtained by the superposition of pure tones. The controlling sound wave, as a result, can also be constructed from the superposition of sound waves required to control each of the pure tones [12]. Hence, the basic step to find the required controlling sound would be to determine the general equation for the individual controlling sound, which can be used to control a pure tone of any given period, w.

4.1. Sound pressure at control point due to a pure tone and control sound

Assuming the sound source generates a pure tone, the sound pressure received at point b by the microphone is as follows:

$$\Delta P_{a,b} = \Delta P \ \sin(\omega t) \quad (3)$$

where t represents time, $\Delta P_{a,b}$ is the sound pressure measured at point b produced by the sound source located at point a, and ΔP denotes the peak amplitude of the original sound pressure at point b, recalling that the Δ character represents a change in the quantities.

When a point source generates periodic sound waves through a free field, the following equations explain the relation

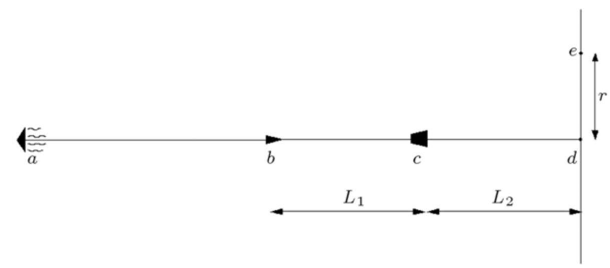

Figure 2: System dimensions.

between I = sound intensity, defined as the sound power per unit area, R = distance from the sound source and P_s = sound power [11]:

$$I = \frac{P_s}{4\pi \cdot R^2} \quad (4)$$

where:

$$I \sim \Delta P^2. \quad (5)$$

From Eqs. (4) and (5), it can be concluded that:

$$\Delta P \sim 1/R. \quad (6)$$

Applying Eq. (6) to determine a relationship between sound pressures at points b and d, leads to:

$$\frac{|\Delta P_{a,b}|}{|\Delta P_{a,d}|} = \frac{R_{ad}}{R_{ab}} = 1 + \frac{L_1 + L_2}{R_{ab}} \quad (7)$$

where $\Delta P_{a,b}$, $\Delta P_{a,d}$, R_{ab}, R_{ad} and L_1 and L_2, which are also shown in Figure 2, are original sound pressure at point b, original sound pressure at point d, distance of point b to original sound source, distance of point d to original sound source, distance from microphone to loudspeaker and distance from loudspeaker to control point, respectively.

For cases where the sound source is far from points b, c and d, $L_1 + L_2 \ll R_{ab}$, and the right term in Eq. (7) is approximately equal to 0, from which it can be concluded that the pressure amplitude of the external sound is almost the same at both points, b and d:

$$|\Delta P_{a,b}| = |\Delta P_{a,d}| = \Delta P. \quad (8)$$

Furthermore, denoting the wavelength of sound by λ, when the sound wave travels distance x between any two points in space, the phase difference between the two points is as follows [11]:

$$\phi' = -2\pi \frac{x}{\lambda}. \quad (9)$$

4.1.1. Pressure at point d because of sound generated at source a

From Eqs. (2), (8) and (9), the sound pressure induced at point d, because of the source at point a, is denoted by $\Delta P_{a,b}$:

$$\Delta P_{a,d} = \Delta P \ \sin\left(wt - 2\pi \frac{(L_1 + L_2)}{\lambda}\right). \quad (10)$$

4.1.2. Pressure at point d because of controlling sound at c

For the most general type of controlling sound, the amplitude of the controlling sound pressure at point d can be calculated from:

$$\Delta P_{c,d} = \Delta P' \ \sin\left(w't - 2\pi \frac{(L_1 + L_2)}{\lambda} + \phi\right). \quad (11)$$

where $\Delta P'$ is the peak amplitude of control sound pressure at point d, and ϕ is the phase difference between the control sound and the original sound at point d.

4.1.3. Total sound pressure at point d

The total sound pressure at point d can be obtained by adding the two pressures from the source and controller:

$$\Delta P_d = \Delta P_{a,d} + \Delta P_{c,d}. \tag{12}$$

Using Eqs. (9) and (10), the detailed form of Eq. (12) is:

$$\Delta P_d = \Delta P \sin\left(wt - 2\pi\frac{(L_1+L_2)}{\lambda}\right) + \Delta P' \sin\left(w't - 2\pi\frac{(L_1+L_2)}{\lambda} + \phi\right). \tag{13}$$

It can be concluded from Eq. (13) that an anti-phase sound with the following characteristics is able to control the sound pressure at d completely:

$$\left.\begin{array}{l}\phi = \pi\\ f' = f\\ \Delta P' = \Delta P\end{array}\right\} \Rightarrow \Delta P_d = 0$$

$$\text{or, equivalently: } \left.\begin{array}{l}\phi = 0\\ f' = f\\ \Delta P' = -\Delta P\end{array}\right\} \Rightarrow \Delta P_d = 0. \tag{14}$$

The above control of sound pressure at a certain point (here, point d), can have application in some very precise engineering problems, such as the control of sound induced vibration in places where a sensitive measurement device is located, i.e., an emergency outdoor telephone, etc. However, the above formulation can also be used in cases where it is desired to reduce the sound noise in larger environments, such as waiting areas in public places like airports, train stations, and restaurants, or in residential buildings and houses that are close to highways or airports. The following section formulates this problem.

4.2. Controlling sound pressure at points in the vicinity of the control point

Point e in Figure 2 is located at distance r from control point d in a surface perpendicular to line ad. As the sound source is far from both d and e, the phase difference between d and e is negligible. Therefore:

$$\Delta P_{a,e} = \Delta P_{a,d} = \Delta P \sin\left(wt - 2\pi\frac{(L_1+L_2)}{\lambda}\right). \tag{15}$$

However, the loud speaker at point c is close to both points, d and e, and there is both an amplitude and a phase difference in controlling the sound which arrives at these points. Eq. (7), in this case, reduces to:

$$\frac{|\Delta P_{c,e}|}{|\Delta P_{c,d}|} = \frac{R_{cd}}{R_{ce}} \tag{16}$$

where R_{cd} and R_{ce} denote the distances between d and e to c, respectively. Hence:

$$|\Delta P_{c,e}| = \frac{L_2}{\sqrt{L_2^2 + r^2}}\Delta P'. \tag{17a}$$

Noticing that:

$$R_{ce} - R_{cd} = \sqrt{(L_2^2 + r^2)} - L_2 \tag{17b}$$

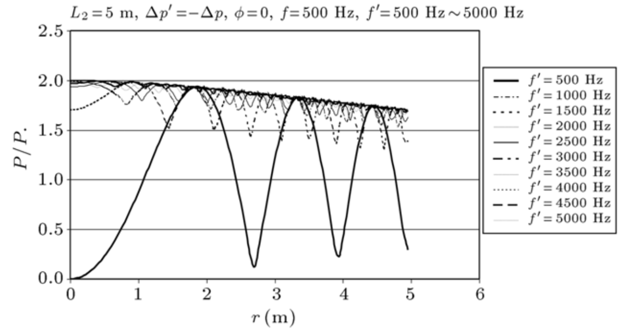

Figure 3: Effect of different control frequencies.

and using Eq. (9), the phase difference of the control sound between d and e is:

$$\Delta\phi = 2\pi\frac{\sqrt{L_2^2 + r^2} - L_2}{\lambda'}. \tag{18}$$

The control sound pressure at e can then be determined using Eqs. (11), (17) and (18), as follows:

$$\Delta P_{c,e} = \frac{L_2}{\sqrt{L_2^2 + r^2}}\Delta P' \sin \times \left(w't - 2\pi\frac{(L_1+L_2)}{\lambda} - 2\pi\frac{\sqrt{L_2^2 + r^2} - L_2}{\lambda'} + \phi\right). \tag{19}$$

Since:

$$\Delta P_e = \Delta P_{a,e} + \Delta P_{c,e} \tag{20}$$

and substituting from Eqs. (15) and (19) into (20), the sound pressure at point e is determined as:

$$\Delta P_e = \Delta P \sin\left(wt - 2\pi\frac{(L_1+L_2)}{\lambda}\right) + \frac{L_2}{\sqrt{L_2^2 + r^2}}\Delta P' \sin \times \left(w't - 2\pi\frac{(L_1+L_2)}{\lambda} - 2\pi\frac{\sqrt{L_2^2 + r^2} - L_2}{\lambda'} + \phi\right). \tag{21}$$

5. Design of optimum controller

Eq. (13) for point d and Eq. (21) for points around d are now used to study the effect of the different parameters of the control problem on the controlling of sound noise in the vicinity of d. The effects of independent parameters of the problem, including the amplitude and frequency of the source and control sound, and the distance between the source, control and target points on the control results, have been studied, and the sensitivity of the control results to variations in each parameter has been investigated. The independent parameters selected have included: $f, f', \Delta P'/\Delta P, L_1, L_2, \phi$ and r, where f is the frequency of original sound and f' is the frequency of control sound (where the angular frequency can be determined from $\omega = 2\pi f$). Some results of this parametric study are shown in Figures 3–7.

Figure 3 represents the relationship between r and $P/P.$, where r = distance to point d, and P and $P.$ = sound pressures at control point, after controlling and before controlling,

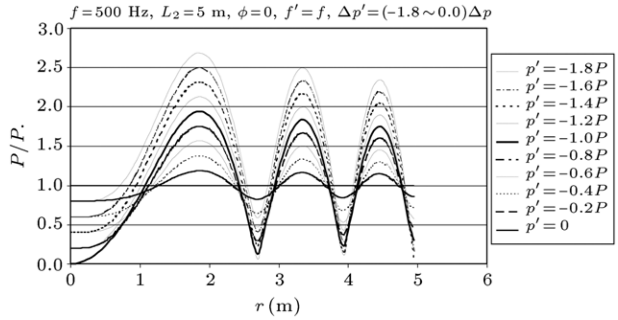

Figure 4: Effect of different control amplitudes.

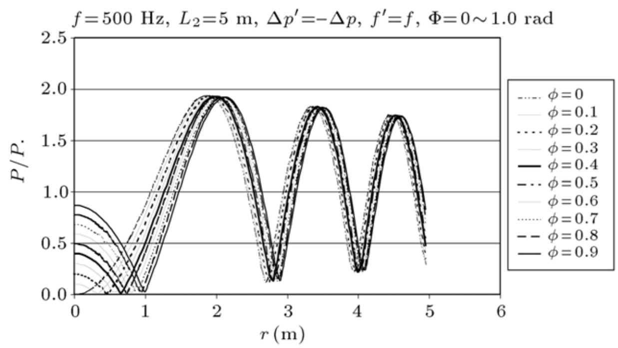

Figure 5: Effect of different phases.

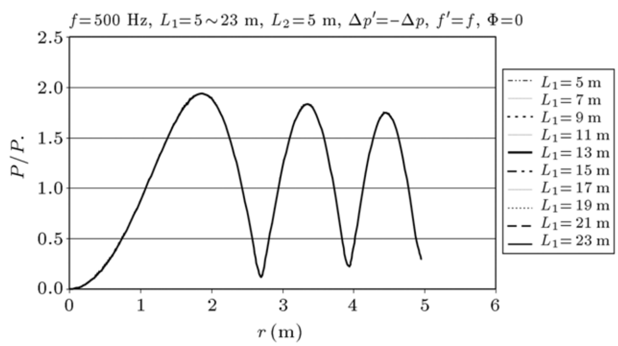

Figure 6: Effect of different L_1 values.

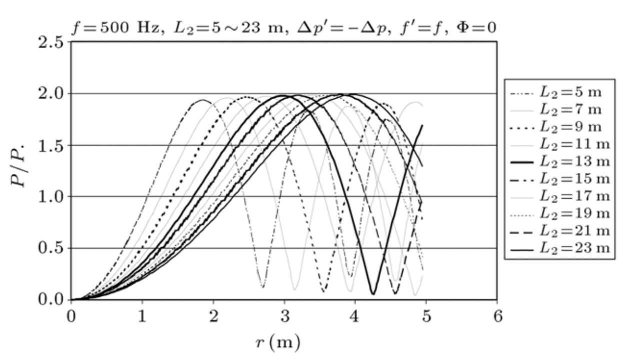

Figure 7: Effect of different L_2 values.

respectively, for $f = 500$ Hz, $L_2 = 5$ m, $\phi = 0$ and $\Delta P' = -\Delta P$, where different graphs have been drawn for different f' values.

The first conclusion from this part of the parametric study has been that the frequency of controlling sound must be the same as that of the sound source for the largest area to be controlled (Figure 3). Similar results have been obtained for other frequencies of the controlling sound, f', which are not shown in Figure 3 because of space limitation. Furthermore, the same calculation and diagrams are provided for other frequencies of sound source f, and also different amounts of L_2, all of which lead to the same conclusion.

Figure 4 represents the relationship between r on the horizontal axis and $P/P.$ on the vertical axis, for $f = 500$ Hz, $L_2 = 5$ m, $\phi = 0$ and $f' = -f$, where different graphs have been drawn for different $\Delta P'/\Delta P$ values. As shown in the graphs, $\Delta P' = -\Delta P$ can result in the best controlled area. It can be concluded that the optimum amplitude of control force at control point d is the amplitude with the same absolute value of original sound pressure that exists at the control point. Similar results have been obtained for other proportions of $\frac{\Delta P'}{\Delta P}$, which are not shown in Figure 4 due to space limitation. Moreover, the same calculation and diagrams are provided for other frequencies of sound source (f) and also different values of L_2 all of which have lead to the same conclusion.

Figure 5 shows one of the many diagrams in which the effect of different phase angles, ϕ, have been studied. An interesting result of this parametric study has been that, although, for controlling the sound at a given point d, an anti-phase sound can be used (Eq. (14)), it will not create the largest controlled area around the point. As can be seen in Figure 5, where an anti-phase controlling sound has been used, the pressure at point d ($r = 0$) has reduced to zero, but the area in which the sound pressure has reduced by 50%, has been inside a circle with a radius of about 0.8 m, while, for $\phi = 0.5$ rad, the area corresponding to more than 50% reduction has been a circle with a radius of more than one meter.

Referring to Figure 6, it is obvious that the distance between the microphone and the control source (L_1) has no effect on the control results. However, as mentioned in Section 3, this parameter determines the feasibility of the control system. This is also correct for other sound source parameters and other distances of L_1 and L_2 not shown in Figure 6.

Figure 7 explains the relationship between r on the horizontal axis, and $P/P.$ on the vertical axis, for $f = f' = 500$ Hz, $\phi = 0$ and $\Delta P' = -\Delta P$, where different graphs have been drawn for different L_2 values. The study has shown that the longer the distance is between the control source and the control point (L_2), the more successful the control results will be. In addition to Figure 7, the same calculation and diagrams are provided for other frequencies of sound source f, and also different values of L_2, which have provided the same result.

6. Closed form equations

Using the results obtained from this parametric study, it is possible to simplify Eqs. (13) and (21). One conclusion has been that the frequency of the controlling sound, f', should be the same as the frequency of the source sound, f, and the peak amplitude of the control pressure is equal to the sound pressure at the control point, i.e. $\Delta P' = -\Delta P$. Therefore, Eqs. (13) and (21), respectively, become:

$$\Delta P_d = \Delta P \sin\left(wt - 2\pi \frac{(L_1 + L_2)}{\lambda}\right) - \Delta P \sin\left(wt - 2\pi \frac{(L_1 + L_2)}{\lambda} + \phi\right) \quad (22)$$

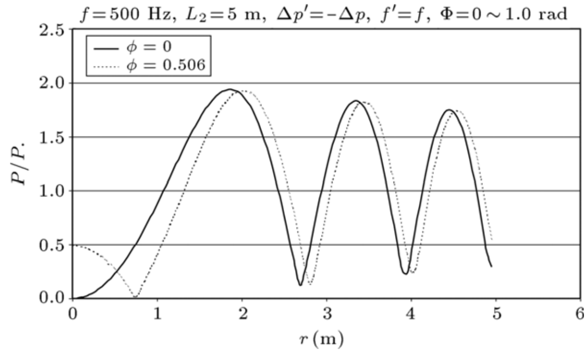

Figure 8: Comparison of results for $\phi = 0$ and $\phi = 0.506$ rad.

and:

$$\Delta P_e = \Delta P \sin\left(wt - 2\pi\frac{(L_1+L_2)}{\lambda}\right) - \frac{L_2}{\sqrt{L_2^2+r^2}}\Delta P \sin \times \left(wt - 2\pi\frac{(L_1+L_2)}{\lambda} - 2\pi\frac{\sqrt{L_2^2+r^2}-L_2}{\lambda} + \phi\right). \quad (23)$$

6.1. Closed form equation for sound pressure at control point

Using the following equation from basic trigonometry:

$$\sin a - \sin b = 2\sin\left(\frac{a-b}{2}\right)\cos\left(\frac{a+b}{2}\right) \quad (24)$$

by implementing Eq. (24) into Eq. (22),

$$\Delta P_d = 2(\Delta P)\sin\left(\frac{-\phi}{2}\right)\cos\left(wt - \frac{2\pi(L_1+L_2)}{\lambda} + \frac{\phi}{2}\right). \quad (25)$$

So, the sound at d after controlling is a pure tone, where the amplitude is:

$$|\Delta P_d| = 2(\Delta P)\sin\left(\frac{\phi}{2}\right). \quad (26)$$

Eq. (26) provides a clear explanation for what has been mentioned in the last paragraph of Section 5. In other words, to reduce the sound pressure at point d and its neighboring area by 50%, the phase difference is calculated from Eq. (26), as follows:

$$|\Delta P_d| = \frac{1}{2}\Delta P \to 2\sin\left(\frac{\phi}{2}\right) = \frac{1}{2} \to \phi = 0.506 \text{ rad}. \quad (27)$$

Figure 8 provides additional explanation for this result.

6.2. Closed form equation for sound pressure in vicinity of controlled point

Eq. (23) can be modified as follows:

$$\Delta P_e = a_1\sin(wt+\phi_1) + a_2\sin(wt+\phi_2) \quad (28)$$

where:

$$\frac{a_2}{a_1} = -\frac{L_2}{\sqrt{L_2^2+r^2}} \quad (29)$$

$$\phi_1 = -2\pi\frac{L_1+L_2}{\lambda} \quad (30)$$

$$\phi_2 = -2\pi\frac{L_1+L_2}{\lambda} - 2\pi\frac{\sqrt{L_2^2+r^2}-L_2}{\lambda} + \phi. \quad (31)$$

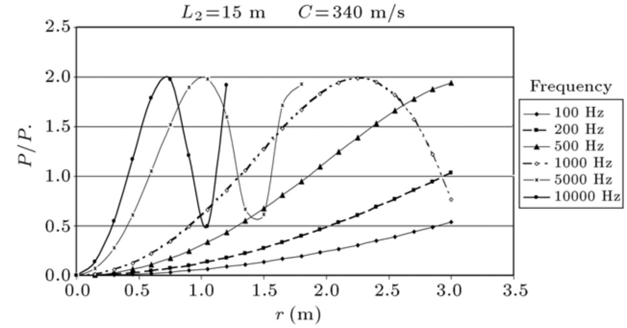

Figure 9: Original sounds with different frequencies.

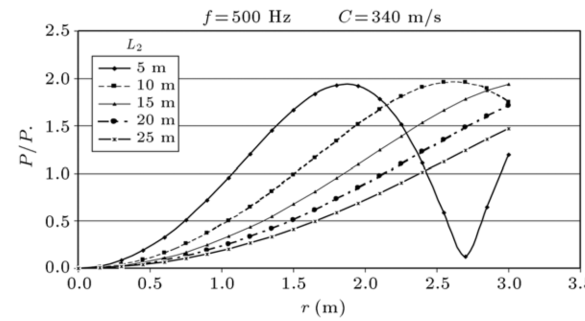

Figure 10: Different positions of control source.

The maximum amplitude at any point can be determined by putting equal to zero the first derivative of ΔP_e in Eq. (28), with respect to time, as follows:

$$\frac{d(\Delta P_e)}{dt} = 0 \to a_1 w\cos(wt+\phi_1) + a_2 w\cos(wt+\phi_2) = 0 \quad (32)$$

where after simplification:

$$\tan(wt) = \frac{\cos\phi_1 + \frac{a_2}{a_1}\cos\phi_2}{\sin\phi_1 + \frac{a_2}{a_1}\sin\phi_2}. \quad (33a)$$

From which the time at which the maximum amplitude occurs is determined as follows:

$$t = (1/w)\tan^{-1}\frac{\cos\phi_1 + \frac{a_2}{a_1}\cos\phi_2}{\sin\phi_1 + \frac{a_2}{a_1}\sin\phi_2}. \quad (33b)$$

Finally, the maximum pressure of sound at point e is determined by substituting t from Eq. (33b) in Eq. (28).

7. Results

The sound pressure has been controlled for audible frequencies, i.e. 20 Hz to 20 kHz, and different positions of the control source. Some of the results are shown in Figures 9–12. These graphs are drawn based on Eqs. (25) and (28) or, equivalently, Eqs. (22) and (23).

Figure 9 represents the relationship between r = distance to point d, and $P/P.$ = proportion of sound pressure at the control point after its controlling to the sound pressure before controlling, for L_2 = 15 m, ϕ = 0 and $\Delta P' = -\Delta P$, where different graphs have been drawn for different frequencies, f. As can be seen in these graphs, as the frequencies have decreased from 10,000 to 100 Hz, the graphs have tended to stretch

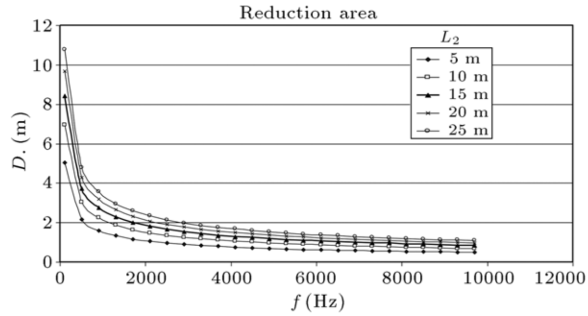

Figure 11: Diameter of reduction area.

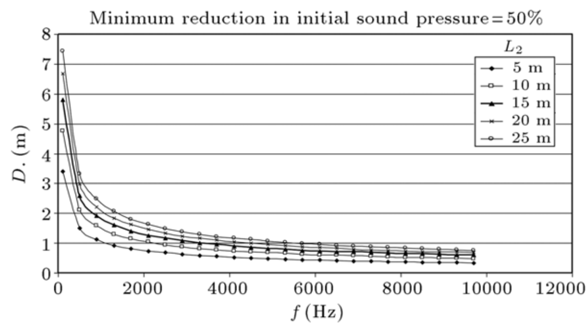

Figure 12: Diameter of area corresponding to at least 50% reduction in original sound.

towards larger r values. In other words, it has been possible to control a larger area for lower frequencies.

Figure 10 has been obtained for cases when the frequency has been fixed to 500 Hz, while L_2 has been changed from 5 to 25 m. From this graph, it can be concluded that by increasing the distance of the loud speaker from the control point, L_2, the controlled area has become larger. So, one way to control a larger area is by placing the controller farther away. However, as the intensity of the sound is directly related to the square of the distance, the farther the control source is placed, the more energy will be needed. For example, by doubling the distance of the loudspeaker from the control point, the intensity should be 4 times more to provide the same control pressure at the control point.

In Figure 11, the effect of both parameters, the frequency and the distance of the control source to the control point, are shown together. The horizontal axis shows the frequency of the sound source and the control source, and the vertical axis shows the diameter of the circle around the control point, where the sound pressure has been reduced because of its controlling. Figure 12 is similar to Figure 11, however, the control criteria has been stronger, where controlling to below 50% of the original sound pressure has been desired. In both figures, the curves corresponding to larger L_2 values have shifted farther from the origin, which, as mentioned before, shows better controlling results as L_2 has increased. Also, for all the curves, the diameter of the controlled area has reduced dramatically as the frequency has increased.

8. Control algorithm

According to the previous sections and results, it is concluded that active control of sound, with the system shown in Figure 1, is a feasible way of sound control. For better emphasis, the algorithm is summarized as follows:

1- A microphone with a nearly uniform frequency response is placed at a point between the sound source and control point, pointing towards the sound source.

2- A speaker is placed at a point between the microphone and the control point, pointing towards the control point.

3- The distance between the microphone and speaker is measured, denoted by L_1.

4- The distance between the speaker and control point is denoted by L_2.

5- Both the microphone and speaker are connected to the controller (a computer system).

6- "The desired percentage of sound attenuation" is input to the controller. This percentage is equal to $= |\Delta P_d|/\Delta P$ in Eq. (26).

7- Using Eq. (26), the desired percentage of sound attenuation $= 2\sin(\frac{\emptyset}{2})$, from which ϕ is determined.

8- The amplitude, frequency and wavelength (ΔP, f and λ) of the original sound pressure are sensed by the microphone and sent to the controller.

9- w is determined as $w = 2\pi f$.

10- The control sound is calculated from $\Delta P \sin(wt - 2\pi \frac{(L_1+L_2)}{\lambda} + \phi)$.

11- The control sound is sent to the speaker to be applied at time $t = L_{1/}f.\,\lambda$, instantly, after the original sound was sensed by the microphone.

The interference of the original and controlling sound pressures provides the final sound pressure at the control point and its surrounding, which is attenuated by the desired percentage in the largest possible area.

9. Conclusions

In this paper, the active control of sound in a free field has been studied. The study includes a check for the feasibility of the system, the formulation of control forces in terms of sound pressure in a general form, a parametric study on each factor of the control sound and the best form of such factors relating to the original sound, using the results of parametric study in order to reach closed form equations for the control sound and reduced sound. Finally, a thorough investigation of the effectiveness of this active control system for a wide range of original sound sources (different original source frequencies) and also changes in the dimensions of the control system (different distances between the control force and control point) is undertaken.

The results show that the frequency and absolute amplitude amount of the control sound must be the same as that of the original sound source.

It was shown that for achieving the largest area of a desired percentage of reduction, not only can an anti-phase sound in the control point be used, but a certain amount of phase difference is available that will provide the largest reduction area.

By investigating different original sound frequencies, it was understood that high frequency sounds will be cancelled at the control point, but they tend to increase significantly at a short distance from the control point. But for low frequency sounds, usually below 1000 Hz, after cancellation of sound at the control point, there is a slight upward increase which is a trend that causes a reasonable area of reduction near the control point. Therefore, the most effective reduction involves low frequency sound.

Moreover, to obtain better control results using this active control system, the distance between the control source and the control point should be increased.

The system described previously for controlling pure tones can also be developed for any periodic sound, as it is known that periodic waveforms can as closely as desired be approximated by the sum of a series of sine waves using the Fourier series.

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