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ScienceDirect
Journal of Sound and Vibration 320 (2009) 726–743
JOURNAL OF SOUND AND VIBRATION
www.elsevier.com/locate/jsvi

Active control of noise on the source side of a partition to increase its sound isolation

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Received 29 November 2007; received in revised form 2 September 2008; accepted 3 September 2008
Handling Editor: C.L. Morfey
Available online 8 October 2008

Abstract

This paper describes a local active noise control system that virtually increases the sound isolation of a dividing wall by means of a secondary source array. With the proposed method, sound pressure on the source side of the partition is reduced using an array of loudspeakers that generates destructive interference on the wall surface, where an array of error microphones is placed. The reduction of sound pressure on the incident side of the wall is expected to decrease the sound radiated into the contiguous room. The method efficiency was experimentally verified by checking the insertion loss of the active noise control system; in order to investigate the possibility of using a large number of actuators, a decentralized FXLMS control algorithm was used. Active control performances and stability were tested with different array configurations, loudspeaker directivities and enclosure characteristics (sound source position and absorption coefficient). The influence of all these parameters was investigated with the factorial design of experiments. The main outcome of the experimental campaign was that the insertion loss produced by the secondary source array, in the 50–300 Hz frequency range, was close to 10 dB. In addition, the analysis of variance showed that the active noise control performance can be optimized with a proper choice of the directional characteristics of the secondary source and the distance between loudspeakers and error microphones.

1. Introduction

A common noise control problem in enclosed spaces is the reduction of sound radiated into rooms adjacent to the one containing a noise source. Passive partitions are often used to cut down the transmitted acoustic energy. Several studies [1–5] have focused on the behaviour of passive barriers, with the aim of predicting their performances and improving their efficiency. In particular, the insertion loss (IL) of partitions in the low frequency range depends on several parameters, such as the room dimensions, the reverberation time and the noise source position [5]. A study by Sas et al. [6] on double layer partition efficiency showed that the performances of these structures decrease sharply at low frequencies and their transmission loss was enhanced

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0022-460X/$ - see front matter
doi:10.1016/j.jsv.2008.09.004

by inserting small secondary acoustical sources into the air gap between two panels. The study provided the groundwork for the method described in this paper: the main idea underlying our approach is to reduce the sound pressure on the primary source side of a dividing wall by means of an array of secondary acoustic sources. The array is used to cancel the sound pressure at the location of an array of error microphones (placed close to the partition surface) and, consequently, a zone of quiet is generated on the incident partition side. In other words, this setup is expected to reduce the potential energy on the partition and therefore to reduce also the sound radiated into a contiguous room. The main advantage of this method in comparison with that described in Ref. [6] is that sound isolation can be enhanced without mechanically affecting the original structure. This method can therefore be applied *a posteriori* to an existing wall that cannot be re-built (for structural reasons, for instance).

An important potential field of application of this method is the reduction of machinery noise in warships and submarines, since the sound pressure reduction inside the keel is expected to cut down the noise transmitted to the water. The effectiveness of this method was confirmed in a recent study [7], where it was established that the active noise control (ANC) system can be used to reduce the sound pressure level on the source side of a partition by more than 10 dB in a frequency range from 25 to 200 Hz. Simulations showed that, in small rooms, the sound pressure surface attenuation and the feedforward noise control stability mainly depend on the distance between the loudspeakers and the error microphones and on the directivity of the loudspeakers. Other minor factors are the distance between the error microphones and the wall, the wall absorption coefficient and the grid meshing of the active barrier. Room dimensions and sound source position have negligible effects on system performances and noise control stability.

The paper is presented as follows. Section 2 introduces the proposed method. Section 3 describes the experimental results that are discussed in Section 4. The paper is then concluded in Section 5.

2. Description of the method

The proposed approach for reducing the noise transmitted between adjacent enclosures is illustrated in Fig. 1. Let us consider two contiguous rooms, one of which contains a sound source (Fig. 1a). The noise transmitted to the neighbouring enclosure is the sum of the structure borne sound and the sound transmitted through the partition (STP). The former has been widely analyzed and discussed in the literature [8] and will not be dealt with here. In cases where the noise transmitted through the partition is much greater than the structure borne noise, noise control procedures usually focus on reducing the STP, which depends on the sound isolation of the division wall and on the sound incident on the partition (SIP). Unlike traditional noise reduction approaches, which involve designing more effective partitions, the method proposed here is designed to attenuate the noise incident on the dividing wall (Fig. 1b). Sound pressure is reduced using a particular ANC [9] technique, where an array of loudspeakers creates a quiet zone by generating destructive interferences on the surface where an array of error microphones is positioned. Henceforth, the arrangement of secondary sources and error microphones arrays will be referred to as ‘‘diagonal active barrier’’ (DAB). The terms ‘‘active barrier’’ recall the earliest application of this setup that was proposed more than 15 years ago to substitute the traditional noise barriers used outdoor [10]. The word ‘‘diagonal’’ refers to the decentralized

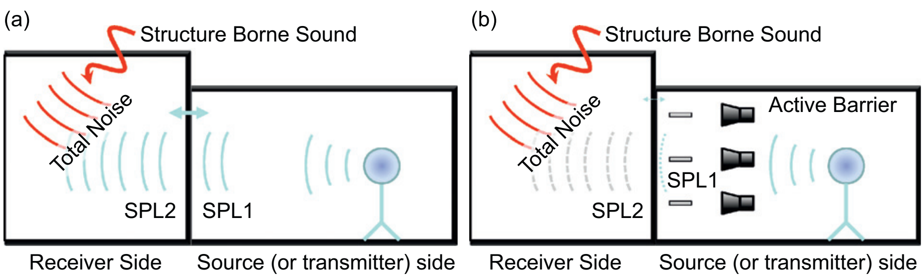

Fig. 1. Diagram of the approach presented in this paper. Comparison between the standard condition (a) and condition (b) with the active noise control system designed to minimize the sound pressure on the right (source) side of the partition.

control algorithm adopted that allows one to simplify the control system in N single-channel control units instead of an $N \times N$ global control system.

The efficiency of this approach has been tested in the facility whose scheme is shown in Fig. 2. The source room is a rectangular enclosure measuring $3.95 \times 2.95 \times 2.05$ m. All the enclosure walls are made of concrete, except the experimental partition, which consists of a plate of steel 3 mm thick with three vertical stiffeners. The ceiling is lined with expanded polystyrene and the floor is made of concrete. The average room absorption coefficient can be adjusted by adding sound absorbers to all the enclosure walls—except for the one with the ANC. The primary source is an omnidirectional cluster of loudspeakers in a dodecahedral configuration that radiates white noise with a spherical distribution in the 50–300 Hz frequency range. This choice of broadband noise makes it possible to measure in a single experiment the ANC performances of each set of variables tested in the entire frequency band. The use of white noise gives ANC performances that are usually lower than those obtained with pure tones, which means that the active barrier has been tested here under relatively challenging conditions. The primary sound source can be moved to various positions (Fig. 2).

The DAB was set up with 12 secondary sources; tests were carried out with both monopole and dipole radiation characteristics (with the dipole axis perpendicular to the sound-controlled partition). A photograph of the dipole-like loudspeakers is shown in Fig. 3. The monopole-like radiation characteristic was obtained by closing one of the two apertures of the loudspeakers. Based on the results of simulations [7] and previous studies [11], the 12 error microphones were placed near the actively controlled surface (at a distance of 1 cm). The distance between each error microphone and the corresponding loudspeaker could be varied between

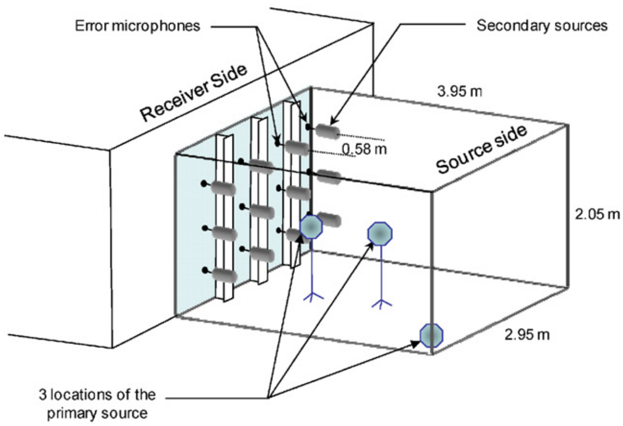

Fig. 2. Scheme of the experimental set-up, giving the room dimensions, the position of the primary noise source and the location of the active noise control system with respect of the stiffened partition.

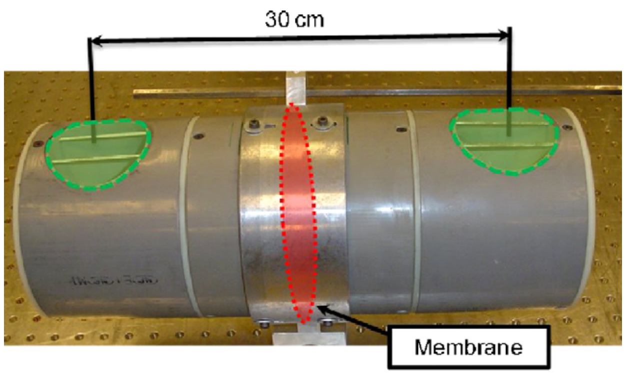

Fig. 3. Picture of one of the 12 loudspeakers (dipole-like configuration). The position of the membrane is evidenced in the centre of the figure. The figure also indicates the distance between the two apertures of the dipole. The monopole-like configuration is obtained by closing one of the two apertures.

15 and 75 cm. Because of the presence of the stiffeners, the distance between the secondary sources was set at 58 cm; this parameter was not varied in our tests, although the results of simulations showed that it is liable to influence the system performance [7]. The effects of possible interactions between factors were analyzed by using a factorial design of experiments (DOEs). This technique is generally used to model the influence of a certain number of independent variables (factors) on one or more dependent variables (response variable). Unlike in the one-factor-at-a-time method, the independent variables are varied together, so as to point out not only the effect of each factor on the response variable (main effect), but also the influence of interactions between the variables.

2.1. The control algorithm

The controller used in our tests was a specialized multichannel, multiprocessor ANC device (COMPARS system) [12], which can be used to perform centralized or decentralized noise control, giving highly efficient real time performances.

The control algorithm used with each of the 12 pairs of loudspeakers and error microphones was a classical time domain single-channel FXLMS algorithm without leakage [13]. The sampling frequency of the controller was 3000 Hz, the antialiasing cut-off frequency was 1400 Hz, the secondary path filter length was 300 points and the control filter length was 200 taps. The convergence coefficient of the algorithm (named β in this paper) was the same with all the modules and was set in each ANC experiment at half the maximum value at which the algorithm was still stable. The electric broadband (50–300 Hz) signal driving the primary source was used as the common reference signal for all the modules. Preliminary exploratory tests showed that using, as a reference signal, the sound pressure measured by a microphone located in an optimized position near the primary source, gave the same sound control performances as those obtained with our set-up. We opted here for the electric signal, however, since it reduces the variability resulting from the microphone position.

2.2. Measurement of the barrier efficiency

The barrier efficiency was assessed on the basis of the IL of the DAB, computed at the limits of decentralized control convergence. The IL was evaluated as the ratio between the spatially averaged sound pressure levels with and without the ANC. Spatial averaging took place on both sides of the partition, as explained later. Let us define the IL at a specific measurement point and frequency f as follows:

$$\mathrm{IL} = -20\log\left|\frac{P(f)}{P_0(f)}\right| \tag{1}$$

where $P(f)$ indicates sound pressure when the DAB is turned on and $P_0(f)$ is the sound pressure measured at the same position when the DAB is turned off. Positive IL values indicate that the ANC system cuts down the noise at the measurement point whereas negative IL values indicate that the DAB increases the primary noise at the measurement point. The spatially averaged IL evaluated in the receiving space (IL_R) reflects the efficiency of the method

$$\mathrm{IL}_R = -10\log\frac{\sum_{i=1}^{8}|P^R(f)|^2}{\sum_{i=1}^{8}|P_0^R(f)|^2} \tag{2}$$

This quantity was measured with eight microphones, the positions of which is shown in Fig. 4b. The transducer locations were chosen to be as far as possible from the error microphones, in order to estimate the barrier efficiency in the most critical conditions. The distance from the plate was selected considering that the optimal solution for minimization of acoustic power measured in the near field of a vibrating plate is very similar to the minimization of far field acoustic power [14]. Owing to the low sound pressure levels to be measured when the DAB is active, microphone distance from the controlled surface was very low (10 cm).

The enclosure in which the observation microphones were placed was an ‘‘ordinary’’ room where some noise sources (such as fans, computers, the ANC controller, etc.) were operating. Although the noise levels generated by these sources were about 30 dB lower than the noise transmitted through the partition, the H_1

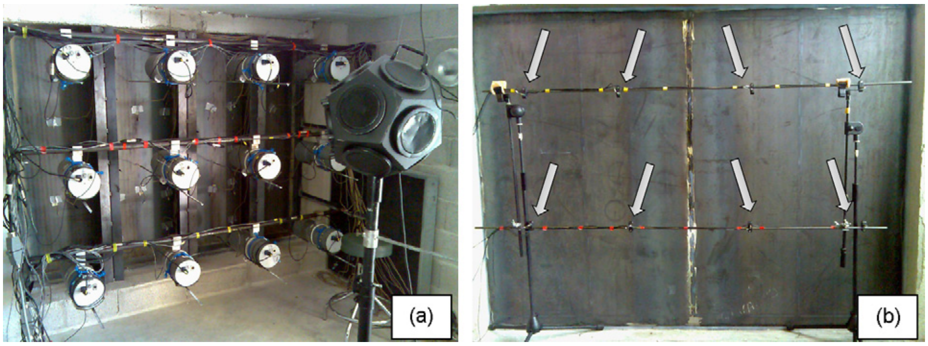

Fig. 4. Photographs of the source side (a) and receiver side (b) of the partition: (a) shows the primary source and the 12 secondary dipole-like loudspeakers and (b) illustrates the positions of observation microphones (marked by arrows) on the receiver side of the partition.

expression [15] of the transfer function between the sound pressure and the electrical input x of the primary source was used rather than the "simple" sound pressure suggested in Eq. (2). If the number of averages in the measurement is sufficiently large, this quantity is not sensitive to the uncorrelated output noise and it therefore provides a more robust index to the DAB performances. Taking G_{xp} and G_{xp_0} to denote the cross-spectra between x and the pressure signals with and without ANC, and G_{xx} to denote the x auto-spectrum, the IL in the receiving space IL_R is given by

$$\mathrm{IL}_R = -10\log\frac{\sum_{i=1}^{8}\left|\dfrac{G_{xp}^{R}(f)}{G_{xp}^{R}(f)}\right|^2}{\sum_{i=1}^{8}\left|\dfrac{G_{xp_0}^{R}(f)}{G_{xx}(f)}\right|^2} \tag{3}$$

Averaging was carried out on the magnitude of the transfer function rather than taking the complex values, in order to obtain a representative picture of the overall potential energy reduction. The control stability was assessed indirectly from the attenuation measured at the 12 error microphones, whose positions are shown in Fig. 4a. If the algorithm is stable, it is possible to use large β values and thus to achieve high levels of attenuation at the error microphones. Conversely, if the algorithm is not stable, the pressure reduction recorded at the error microphones will be small. The IL evaluated on the source side of the partition (IL_S) has been evaluated starting from measurements of the 12 error microphones as follows:

$$\mathrm{IL}_S = -10\ \log\frac{\sum_{i=1}^{12}\left|\dfrac{G_{xp}^{S}(f)}{G_{xx}(f)}\right|^2}{\sum_{i=1}^{12}\left|\dfrac{G_{xp_0}^{S}(f)}{G_{xx}(f)}\right|^2} \tag{4}$$

These independent variables were evaluated with a measurement chain set-up including:

- 12 electret error microphones with conditioning amplifier; uncertainty smaller than 0.2 dB;
- eight ICP measurement microphones located at 10 cm from the receiver side of the partition (uncertainty in the investigated frequency range: 0.1 dB);
- an OROS 38 data acquisition system with 24-bit A/D converters; and
- a PC for data acquisition and signal analysis.

The unconventional metrics IL_R and IL_S were chosen since this part of the paper aims to identify the factors that influence the DAB efficiency, and not to identify the accurate value of the transmission loss of the partition. However, both IL_R and IL_S, are robust indicators of the DAB performances since, thanks to the use of averages, their sensitivity to disturbances is low. Additionally, the time required for a single measurement is shorter than the one required for spatial sound power average on both sides of the partition or for the standard transmission loss measurement as per ISO specifications. Due to the large number of tests compulsory for the statistical analyses, our choice allows reducing the time required for each test. In the final part of this paper, a sound power analysis for a specific DAB configuration will be presented.

2.3. Independent variables

In order to check the validity of the method in a wide range of configurations, the barrier performances were investigated with different primary source positions and changes in the room reverberation time. Geometric and acoustic parameters of the active barrier were also varied, with the aim of understanding their effects on the IL. Independent variables describing the test setup and their values are listed in Table 1.

The results of both recent studies [16] and simulations [7] showed that the use of a dipole-like source can enhance ANC performances: two types of secondary sources (with monopole- and dipole-like directivity patterns) were therefore tested. The bipolar loudspeakers used (with distance between acoustic centres of 30 cm) were identical to those described in Ref. [11]; as previously mentioned, monopole directivity was obtained by closing one of the two apertures of the dipole.

The averaged absorption coefficient of the room could be increased by covering the room walls (except the one with the ANC system) with glass wool layers. The following three absorption conditions were tested:

- no absorption, i.e., no glass wool covering the room surfaces;
- intermediate absorption: the wall opposite the one where the active barrier was set up was lined with absorbent material; and
- full absorption: in addition to the measures used to obtain ‘‘intermediate absorption’’, the other room walls were also covered with glass wool.

The glass wool lining used was 20 cm thick. Thus, the room volume and the surface area of the walls were slightly reduced in the presence of the absorbent. The average absorption coefficients in the 50–300 Hz range (calculated using the Sabine equation) were 0.07 (no absorption), 0.13 (intermediate absorption) and 0.18 (full absorption). The absorption coefficient was found to be reasonably constant in the frequency range between 50 and 300 Hz. The absorption coefficient uncertainty, determined as the standard deviation between theoretically identical tests, was 0.02.

Since ANC performances depend on the distances between the secondary sources and the error microphones, the three distances 20, 35 and 50 cm were tested. With dipoles, these values are the distance between the pole nearer to the partition and the error microphone. The aim of this choice was to compare the two solutions in terms of the free space between the partition and the loudspeaker array. The distance between the acoustic centre of a dipole-like source and its error microphone can be derived by adding 15 cm to the three above listed values.

The primary source was placed in three different positions (see Fig. 2). In the position labelled ‘‘near’’, the primary source was 2 m far from the partition. In the ‘‘mean’’ position, the primary source was 3 m from the partition. In these two positions, the source was placed at a height of 1.5 m, at a horizontal distance of 1 m

Table 1
ANOVA factors and their levels

Factor	Description	Levels investigated in ANOVA		
–	Loudspeaker directivity	Monopole		Dipole
A	Room absorption	No	Intermediate	Maximum
B	Distance error microphones–loudspeakers	20 cm	35 cm	50 cm
C	Source position	Near	Centre	Corner

Factors are the independent variables describing the test configuration and A, B and C are the letters with which factors are referred to in the next tables.

from the side wall. In the "corner" position, the primary source was located in a corner of the room so as to excite all the acoustic modes of the enclosure.

2.4. Factorial DOEs

The effects of the independent variables on the active barrier performances were assessed by using a factorial DOEs [17]. Similarly to what was done in Ref. [7], all the factors were varied together to show any cross-interaction occurring between independent variables. With the DOE technique each factor can assume two or more values; in the simplest case of two levels for each factor, if one wants to investigate the effect of K factors with all the possible interactions, 2^K tests are required. When all data have been collected, the behaviour of dependent variables is described with a regression analysis. The second order regression model used in our tests is

$$\xi = \gamma_0 + \gamma_1 x_1 + \gamma_2 x_2 + \cdots + \gamma_{11} x_1^2 + \gamma_{22} x_2^2 + \cdots + \gamma_{12} x_1 x_2 + \cdots + \gamma_{112} x_1^2 x_2 + \gamma_{122} x_1 x_2^2 + \cdots + \varepsilon \quad (5)$$

where:

- ξ is the dependent variable (IL_R or IL_S);
- x_i are the independent variables expressed in terms of coded units (the variables can take values of 1, 2 or 3 corresponding to low, mean and high levels, respectively);
- γ_0 is the overall test average;
- γ_1 describes the effects of the independent variables, of interactions and of the second order effects; and
- ε is the residual, namely the difference between the actual behaviour reflected in the data and the predictions of the model.

ANOVA checks the hypothesis that the variance introduced by an independent variable is significant in comparison with the variance of the residuals. In other words, a factor influences the barrier IL if the variance it introduces is larger than the variance of the residuals. In a full factorial design, residuals are given by test replications: preliminary tests showed that tests repeated in nominally identical conditions had a IL_R standard deviation of 0.2 dB (mean 7.9 dB), while the standard deviation of IL_S was 2.4 dB (mean 23.5 dB). The standard deviation observed in one-third octave bands was usually slightly larger than the above values. Since the threshold of 0.2 dB for the significance of investigated factors was too restrictive, high-order interactions were neglected and then a simplified second order regression model was used. This increases the standard deviations of residuals, and thus the thresholds of meaningfulness (Table 2) for the independent variables. With the adopted regression model, a factor was considered influencing if its effect on IL_R is close to 1 dB, and thus if its contribution was much larger than the threshold of repeatability.

Since all the tests were performed with the same instrumentation, the repeatability values do not include the contribution of systematic errors due to nonlinearity and those due to the metrological calibration of the transducers.

Table 2
Standard deviation of the residuals in the ANOVA analysis: these values represent the thresholds for the multiple hypothesis testing, i.e. factors of Table 1 affect the IL if the variances they introduce are larger than the variances of the residuals

	Frequency (Hz)									
	50	63	80	100	125	160	200	250	315	50–300
Standard dev IL_S monopole	0.92	0.93	1.37	2.09	1.58	2.00	1.48	0.85	1.00	0.79
Standard dev IL_R monopole	0.71	0.76	0.94	1.61	1.70	1.99	1.21	1.49	1.98	1.08
Standard dev IL_S dipole	1.93	1.47	1.51	1.02	0.76	0.82	0.86	1.38	0.86	0.76
Standard dev IL_R dipoles	1.44	1.53	1.10	1.44	1.45	2.12	2.29	1.47	1.91	1.53

3. Experimental results: optimization of barrier performances

Due to the completely different effect of the distances between error microphones and secondary sources on the monopole and dipole loudspeakers, two separate analyses of variance were performed.

3.1. Monopole sources

The influence of independent variables is summarized for both IL_R and IL_S in the plots of Fig. 5. The figure shows that the source position and absorption coefficient effects are negligible in comparison with the distance between secondary sources and error microphones. The largest attenuations in both error and observation microphones were obtained with loudspeakers located near the error microphones: in practice, with a stable control algorithm, it is possible to use large β coefficients and thus to obtain large values of IL_R and IL_S, regardless of the room configuration.

Since the residuals were normally distributed in the case of both IL_R and IL_S, the second order model is suitable for describing data behaviour [17]. In our tests, the null hypothesis H_0 (i.e., the investigated variable effect is below the threshold value given in Table 2) was rejected if the *p*-value (the significance level of the statistical test) was lower than the type I error risk α, which was set at 5%. In other words, the influence of a factor—for instance of the source position—is considered important if the *p*-value is smaller than 5%.

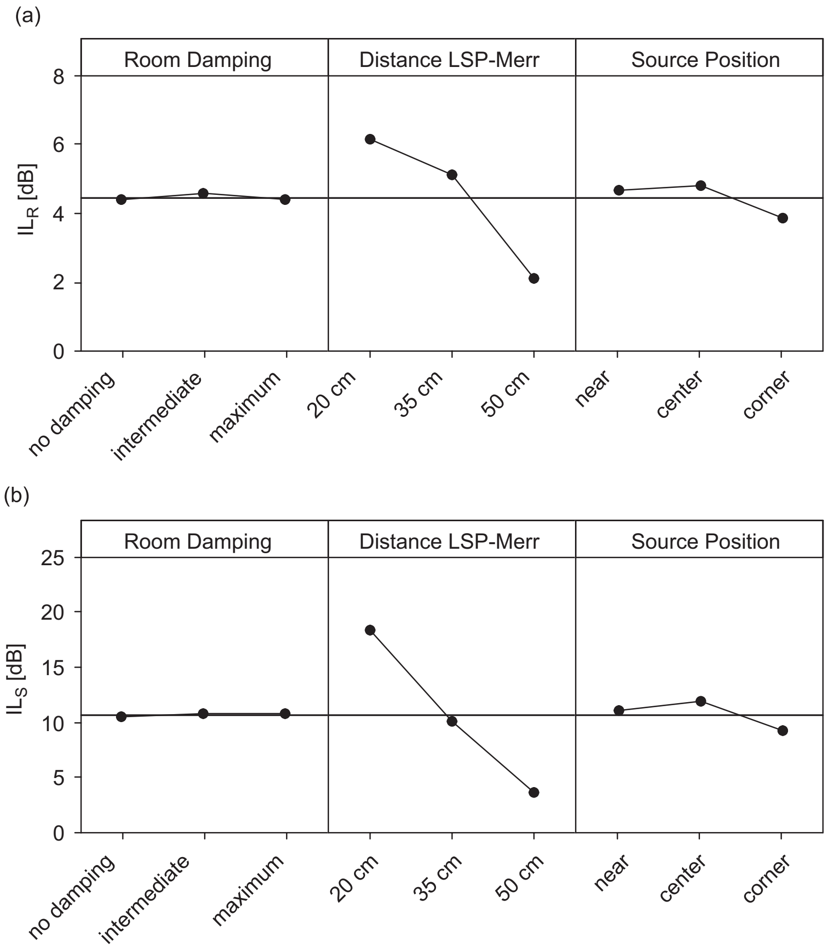

Fig. 5. Insertion losses measured on the receiver side (IL_R) and on the source side (IL_S) of the partition in the 50–300 Hz frequency range for the monopole sources. Each point represents the mean of all the tests characterized by the specific value of the independent variable indicated on the *x*-axis.

Table 3
p Values of the ANOVA for IL_S (related to the stability of the algorithm) on monopoles

Factor	Frequency (Hz)									
	50	63	80	100	125	160	200	250	315	50–300
A	0.09	0.75	0.46	0.56	**0.03**	0.13	0.06	0.14	0.53	0.85
B	**0.00**	**0.00**	**0.00**	**0.00**	**0.00**	**0.00**	**0.00**	**0.00**	**0.00**	**0.00**
C	**0.00**	**0.01**	**0.00**	**0.01**	0.23	**0.01**	0.62	0.80	0.06	**0.00**
A*B	**0.04**	0.15	0.41	0.83	0.49	0.45	0.19	0.06	0.52	0.53
A*C	0.14	0.25	0.42	0.40	0.33	0.34	0.51	0.44	0.49	0.10
B*C	**0.00**	0.15	0.54	0.53	0.52	0.80	0.50	0.34	0.10	0.21

Bold numbers indicate that the factor affects IL_S.

Table 4
p Values of the ANOVA for IL_R (attenuation recorded at the observation microphones) on monopoles

Factor	Frequency (Hz)									
	50	63	80	100	125	160	200	250	315	50–300
A	0.12	0.16	0.84	0.68	0.06	0.06	0.34	0.11	0.22	0.83
B	**0.00**	**0.00**	**0.00**	**0.00**	**0.00**	**0.00**	**0.00**	**0.00**	**0.03**	**0.00**
C	**0.00**	**0.02**	0.27	0.75	**0.04**	0.67	0.34	0.47	**0.02**	0.08
A*B	**0.00**	0.17	0.86	0.81	0.48	0.35	0.25	**0.03**	**0.05**	0.73
A*C	**0.05**	**0.02**	0.30	0.83	0.43	0.60	0.58	0.66	0.79	0.19
B*C	0.06	**0.01**	0.96	0.27	0.09	0.53	0.88	0.09	**0.00**	0.05

Bold numbers indicate that the factor affects IL_R.

The p-values obtained in the ANOVA test are summarized in Tables 3 and 4. ANOVA results show that, especially at low frequencies, IL_S (and thus the stability of the feedforward control) depended on the primary source position. In particular, the ANC was less stable when the source was placed in the corner of the room. This aspect is in accordance with what was found by Elliot et al. in Ref. [18]: their main conclusion was that ANC in lightly damped enclosures is more efficient when there are only a few acoustical modes in the room. Since the corner position excites all the room modes, the above result appears to confirm their findings. In addition, inside an enclosure, it seems to be better to position the secondary sources as close as possible to the primary excitation source, as indicated in Ref. [19] by Baek. Thus the corner position is particularly restrictive, since the distance from the primary and secondary sources is the largest one among the tested conditions. A similar behaviour was also found in the observation microphones, but the effect of the source position amounted to less than 1 dB in this case. Such a small value points out that this method can be applied independently from the source position. The results obtained also show that the second-order interactions did not significantly affect the performances of the system; in other words, interactions between factors were negligible.

3.2. Dipole sources

The influence of independent variables on IL_R and IL_S is shown in Fig. 6, which highlights some important issues. First of all, there exists an optimum configuration, in terms of the distance between the error microphones and secondary sources that in the investigated conditions was the "mean" distance, regardless of the frequency (Fig. 7). The second point is that the performances of the dipoles also depend on the source position and on the room reflectivity. The IL measured at the error microphones IL_S shows that better performances can be achieved in heavily damped rooms. A similar conclusion has also been found by Guo and Pan in Ref. [20]. As previously outlined, ANC efficiency is larger when there are only a few acoustical modes in

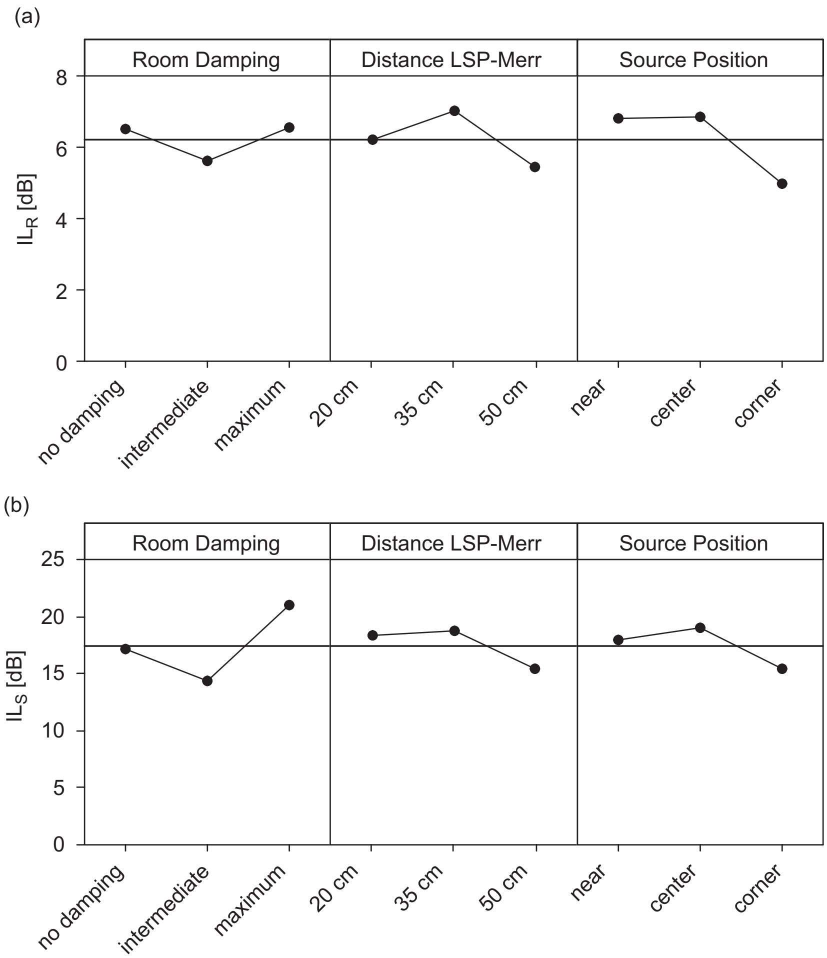

Fig. 6. Insertion losses measured on the receiver side (IL_R) and on the source side (IL_S) of the partition in the 50–300 Hz frequency range for the dipole sources. Each point represents the mean of all the tests characterized by the specific value of the independent variable indicated on the x-axis.

the room. A large damping reduces the effect of acoustic modes and consequently allows having a more efficient control. The plot also shows that better performances were achieved under "no absorption" conditions than under "intermediate absorption" conditions. Although such a difference is close to the test repeatability (less than 1 dB for IL_R and close to 3 dB for IL_S) the conclusion is that, with dipole-like sources, the DAB performances are very sensitive to the room damping and to other environmental factors involved. The DAB IL was also found to decrease when the source was placed in the corner of the room. This aspect has the same reasons already outlined for monopoles and will not be dealt with here.

Let us now analyze the results of the statistical inference procedure. The analysis of the residuals of both IL_R and IL_S showed that the regression model used was appropriate for this purpose. The p-values of the ANOVA are listed in Tables 5 and 6. The only factor that consistently influenced IL_S regardless of the frequency was the room damping. The distance between loudspeakers and error microphones is also important, while the source position does not systematically influence the barrier IL. Since the effects of the factors on IL_R depend on frequency, the DAB cannot be easily summarized. In any case, the conclusion that can be drawn is that dipoles give good results in terms of the attenuation, but that barrier performances are particularly sensitive to the DAB configuration and to the room acoustic parameters. This aspect is also evidenced by of the meaningfulness of second-order interaction in the ANOVA analysis; the barrier optimization, consequently, is more complex.

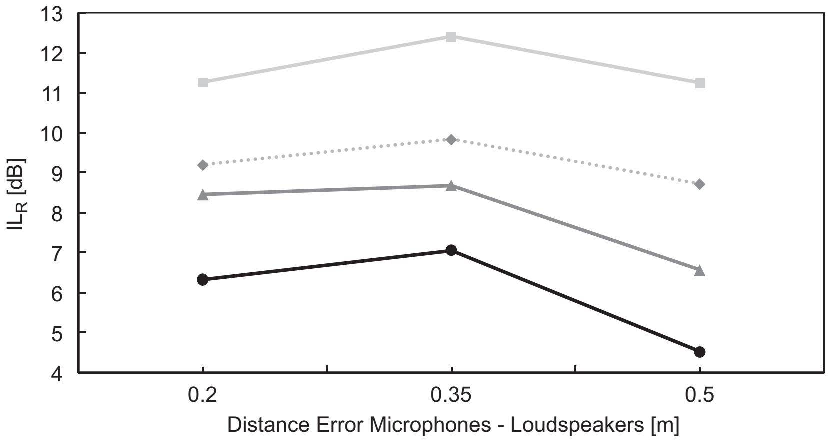

Fig. 7. Spatially averaged insertion loss evaluated on the source side (IL_S) of the partition for the dipole sources as a function of the distance between error microphones and secondary sources: 80 Hz; 125 Hz; 160 Hz; and 200 Hz.

Table 5
p Values of the ANOVA for IL_S (related to the stability of the algorithm) on dipoles

Factor	Frequency (Hz)									
	50	63	80	100	125	160	200	250	315	50–300
A	**0.00**	**0.00**	**0.00**	**0.00**	**0.00**	**0.00**	**0.00**	**0.00**	**0.00**	**0.00**
B	**0.02**	0.07	**0.02**	**0.01**	**0.00**	**0.01**	**0.02**	**0.00**	**0.04**	**0.00**
C	**0.00**	0.05	**0.00**	0.17	0.06	0.06	0.22	0.09	**0.00**	**0.02**
A*B	0.05	**0.04**	0.05	**0.00**	**0.00**	**0.00**	**0.00**	**0.00**	0.96	**0.01**
A*C	**0.04**	0.47	**0.04**	0.53	0.39	0.49	0.45	**0.00**	**0.02**	0.38
B*C	0.62	0.47	0.62	0.90	0.45	0.69	0.75	0.12	**0.00**	0.27

Bold numbers indicate that the factor affects IL_S.

Table 6
p Values of the ANOVA for IL_R (attenuation recorded at the observation microphones) on dipoles

Factor	Frequency (Hz)									
	50	63	80	100	125	160	200	250	315	50–300
A	0.39	**0.00**	**0.04**	**0.02**	**0.00**	**0.00**	0.05	0.05	**0.00**	0.05
B	0.28	0.20	0.06	**0.00**	**0.00**	**0.01**	**0.00**	0.33	**0.04**	**0.01**
C	0.72	0.12	**0.02**	0.13	**0.00**	**0.03**	0.06	0.07	**0.00**	**0.00**
A*B	0.62	**0.02**	0.59	**0.01**	0.09	**0.00**	**0.00**	0.48	0.96	**0.02**
A*C	0.09	**0.02**	0.91	**0.00**	**0.02**	**0.02**	**0.03**	0.63	**0.02**	**0.01**
B*C	0.62	0.34	0.14	0.42	0.09	0.12	0.20	0.52	**0.00**	0.31

Bold numbers indicate that the factor affects IL_R.

4. Discussion

The experimental results presented in the previous section show that an active sound barrier can be used inside an enclosure in order to increase the isolation of a partition. Under the conditions tested here, the factors affecting the barrier performances most strongly are the directivity of the secondary sources and the distance between these sources and the error microphones. With monopole loudspeakers, the DAB gave better performances when the secondary sources were placed relatively near the error microphones (at a distance of

Table 7
Summary of the configurations under various room conditions

Room damping	Source position	Best IL_R dipole	Distance dipole (cm)	Best IL_R monopole	Distance monopole (cm)
No	Near	8.2	20	7	20
	Centre	7	35	5.2	20
	Corner	8.4	35	5.7	20
Intermediate	Near	6.9	35	7.6	20
	Centre	8	50	6.6	35
	Corner	4.6	50	4.7	20
Max	Near	7.9	35	7.4	20
	Centre	9.4	35	6.1	20
	Corner	6.9	35	5.2	20

The first two columns describe the room configuration. The two columns named "best IL_R" for monopoles and dipoles indicate the largest spatially averaged insertion loss achieved in our tests (frequency range 50–300 Hz). The column "Distance" indicates the geometric configuration in which the barrier gave the optimized results.

20 cm). Conversely, when using sources in the dipole-like configuration, better results can be obtained by placing the secondary sources relatively far from the partition. In our tests, the largest attenuations were obtained by placing the dipole-like loudspeakers 50 cm from the error microphones.

Configurations that gave better results under different conditions are summarized in Table 7. The ILs measured on the receiving side of the partition with monopole- and dipole-like sources in different room acoustic conditions can be summarized as follows:

- Under heavily reverberating conditions, regardless of the source position, the best results are obtained with dipole sources with the acoustic centre placed 50 cm from the error microphones. With the dipole arrangement, the distance between the source and the pole nearest to the surface of the partition was 35 cm. With monopole sources and a separation distance of 20 cm, the attenuation levels recorded at the observation microphones were 1.2, 1.8 and 2.7 dB smaller when the source was near the barrier, in the middle of the room and in the corner, respectively. The differences between the results obtained with monopoles and dipoles were greater than the repeatability of the test and the measurement set-up uncertainty.
- Under intermediate damping conditions, the results were found to be almost independent of the loudspeaker directivity. The best results achieved with dipoles were obtained with the pole nearest to the surface 50 cm from the error microphones (i.e., 65 cm from the acoustic centre of the dipole). This means that the barrier was very far from the surface of the experimental wall; depending on the application, this can be either an advantage or a disadvantage.
- Under maximum damping conditions, larger attenuations were achieved with dipole sources and sound sources placed at a distance of 35 cm form the error microphones (i.e., 50 cm from the acoustic centre of the dipole). With monopoles, the attenuation values achieved were 0.5, 1.7 and 3.3 dB lower when the source was placed near the surface of the experimental wall or in the centre or corner of the room.

Better results can generally be achieved with loudspeakers having a dipole radiation characteristic. This aspect is also evidenced by the comparison of Figs. 5 and 6, which shows that with a dipole-like configuration, IL_S and IL_R are larger than the ones measured with the monopole-like loudspeakers. As already outlined, the decentralized control is more stable when using dipole-like sources, thanks to the reduced crosstalk between the loudspeaker and the adjacent error microphones. With a more stable control it is possible to use larger convergence coefficients and therefore to obtain larger ILs. Additional tests, not described in this paper, showed that the difference between the optimized diagonal control and the conventional global one is generally limited (typical values are in the range of 1–2 dB): this confirms the validity of the proposed diagonal approach. With the use of dipoles, it is possible to obtain good overall attenuation levels by placing the secondary sources 50 cm from the surface (65 cm from the acoustic centre). Conversely, the use of monopoles is compatible with shorter separation distances and therefore gives a more compact arrangement. On the other hand, if the loudspeaker array is placed far from the partition the latter remains perfectly accessible to users after the installation of the barrier, and this can be an advantage in some situations. The overall reductions of Table 7 are similar to the one measured by Sas et al. in Ref. [6], but are achieved without modifying the existing partition.

Details of system performances in a specific condition are given in Figs. 8 and 9: the plots can be used to compare the sound pressure levels measured on both sides of the dividing wall with monopoles and dipoles, with and without the active control system. The primary source was placed in the centre of the room; all the side walls were lined with glass wool so as to obtain the highest damping. As shown in Fig. 8, the decentralized algorithm works well over the entire 50–300 Hz frequency band (giving attenuation values ranging from 10 to 30 dB at the 12 error microphones). The plot also shows that the IL on the source side of the partition was greater when dipole-like sources were used. This demonstrates that the decentralized control configuration is more stable and thus larger convergence coefficients can be used. Fig. 9 shows the IL of the DAB measured in the receiving space, while Fig. 10 gives IL_R in third of octaves for the optimized monopole- and dipole-based sound barrier. Both Figs. 9 and 10 point out that a consistent control was achieved between 80 and 200 Hz; outside this range the DAB seems to be less efficient. The low frequency limit—that is slightly different for monopole- and dipole-like loudspeakers (85 and 75 Hz, respectively) can be traced to the reduced acoustic efficiency of the adopted secondary sources in the low-frequency region. This ineffectiveness can be easily overcome with a proper choice of loudspeakers (the ones used in our experiments were optimized for a previous experiment [11] and thus not optimized for the frequency range investigated here). The reduced performances at high frequencies are due to the size of the quite zone around the error microphones. In a diffuse sound field and in the proximity of a reflecting surface, the quite zone diameter is a fraction (more than one tenth) of the acoustic wavelength. At low frequencies, the quite zones of adjacent error microphones overlap, and the entire controlled partition takes benefit of the pressure reduction. On the other hand, at high frequencies the quite zones are not large enough to cover all the controlled partition and the efficiency of the barrier rapidly decreases far from error transducers. Owing to the location of the eight observation microphones in the receiving space (i.e., far from error microphone positions), this phenomenon is particularly evident. A more detailed analysis in this direction was performed by adding 28 additional microphones on the source side of the wall; the DAB IL was measured on both the error and additional microphones. The primary

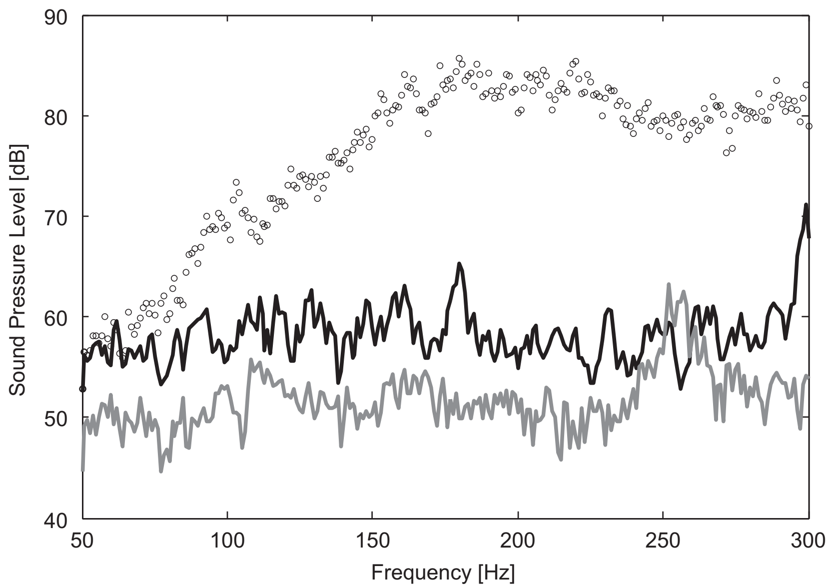

Fig. 8. Sound pressure level measured at the error microphones with and without the DAB: ○ SPL control off; ▬▬ SPL with DAB on secondary monopole-like sources placed at 20 cm from the partition wall; and ▬▬ SPL with DAB on secondary dipole-like sources placed at 35 cm from the partition wall.

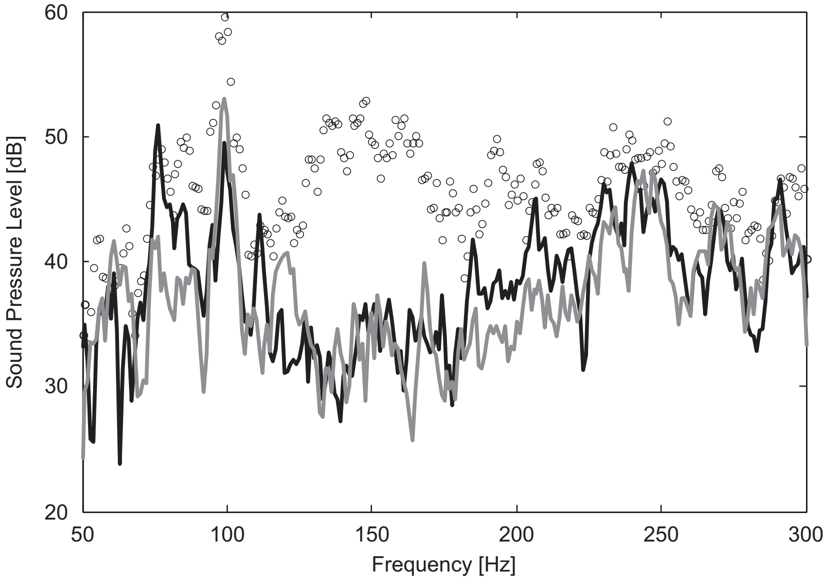

Fig. 9. Sound pressure level measured at the observation microphones with and without the DAB: ○ SPL control off; ▬▬ SPL with DAB on secondary monopole-like sources placed at 20 cm from the partition wall; and ▬▬ SPL with DAB on secondary dipole-like sources placed at 35 cm from the partition wall.

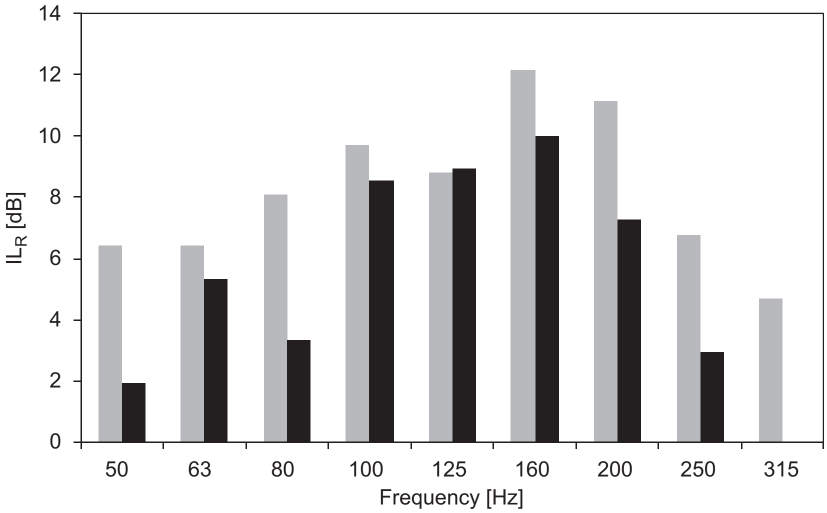

Fig. 10. Comparison between the performances of the monopole-based barrier (black bars) and dipole-based barrier (grey bars) in a specific situation (maximum damping, primary noise source located in the centre of the room).

source was driven by pure tones at frequencies ranging from 100 to 300 Hz. The IL of the DAB is shown in Fig. 11: it can be noticed that the IL on error microphones is always close to 30 dB, but the sound pressure reduction measured on the additional observation microphones decreases with the frequency and DAB performances consequently worsen. From this point of view, results are consistent with the simulations presented in Ref. [7], where it was found that a more dense grid allows larger ILs especially in the high frequency region.

Further experiments were performed with a different attachment method of the secondary loudspeakers that were mechanically uncoupled from the barrier (thus reducing the structure borne sound). The experimental results obtained in this case showed that it is possible to increase the IL up to 3 dB at some

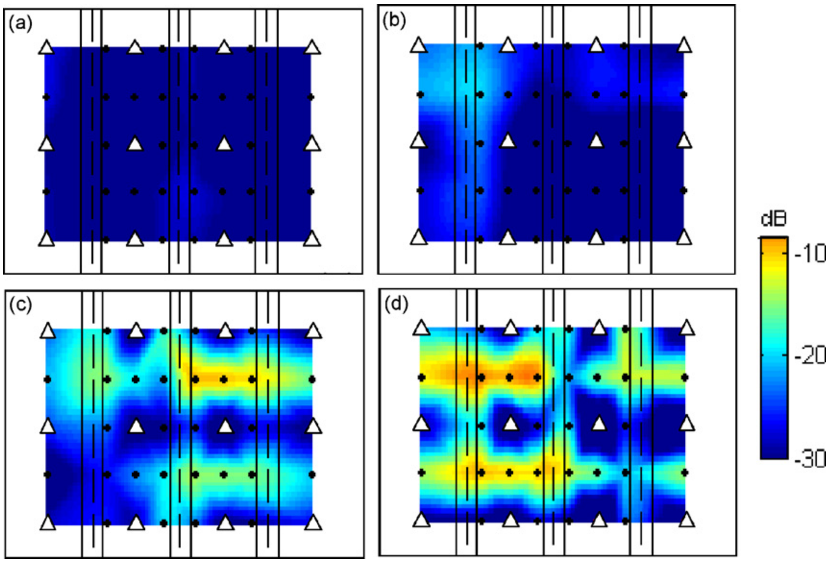

Fig. 11. Sound intensity level of the DAB measured on the source side of the partition at different frequencies: (a) 125 Hz, (b) 200 Hz, (c) 250 Hz and (d) 300 Hz. Triangles indicate the position of error microphones, black dots indicate the position of the 28 additional microphones.

frequencies. It seems reasonable that the improvement of performances should follow by designing specific dipole-like loudspeakers and optimizing the secondary source grid parameters.

4.1. Sound power measurements

As previously mentioned, the choice of unconventional metrics IL_R and IL_S was undertaken for the evaluation of factors that influence the DAB performances. A more generic approach leverages on the sound power analysis that requires the evaluation of the sound intensity on the six surfaces that enclose the active barrier. Owing to the limited space between the secondary sources and the room walls, this kind of analysis is not feasible in our experimental setup. The actual method efficiency was therefore evaluated by means of sound power measurements on the source and on the receiving side of the partition. The DAB was setup with a distance of 35 cm between the nearest pole of a dipole-like loudspeakers and their error microphones; the primary source was placed in the position labelled as centre and the damping condition was the maximum one.

Sound intensity analysis on the receiving space was performed by scanning a surface parallel to the partition (placed at 20 cm from the partition itself to reduce the near field errors of the *p–p* probe) at 110 discrete positions. Similarly to what was done with the spatially averaged sound pressure, data with and without the DAB were compared. The spatially averaged sound intensity component perpendicular to the partition is shown in the plots of Fig. 12. The behaviour is similar to the one evaluated in the same conditions in Fig. 9, thus pointing out the validity of the adopted metrics. The difference between the sound power levels in the range of 50–300 Hz when the DAB is turned on was 9.2 dB: such a value is comparable with the one of IL_R that was pointed out in the previous chapter that was 9.4 dB.

Due to the presence of the loudspeakers, error microphones and cables, sound intensity could not be measured on the source side of the partition; the reduction of the spatially averaged sound pressure level, anyway, has already been confirmed by the analyses performed with the additional microphones (Fig. 11). The energetic behaviour of the active barrier was examined by scanning a surface between the primary source and the secondary source array. Such a surface (parallel to the controlled partition) was located at 40 cm from the acoustic centres of the dipoles. Analyses showed that directional patterns of the sound field inside the enclosure were completely modified by the DAB; an example of the 3D sound intensity map is shown in Fig. 13. The comparison clearly shows changes in the energetic fluxes. Conversely, the spatially averaged sound intensity flowing through the surface in the 50–300 Hz frequency range did not appreciably change with

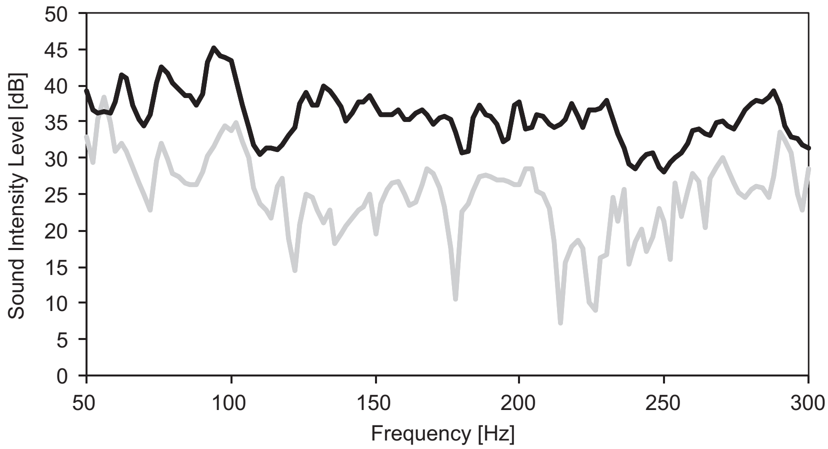

Fig. 12. Comparison of the spatially averaged sound intensity with and without the active barrier: ▬ DAB is turned off and ▬ DAB is turned on.

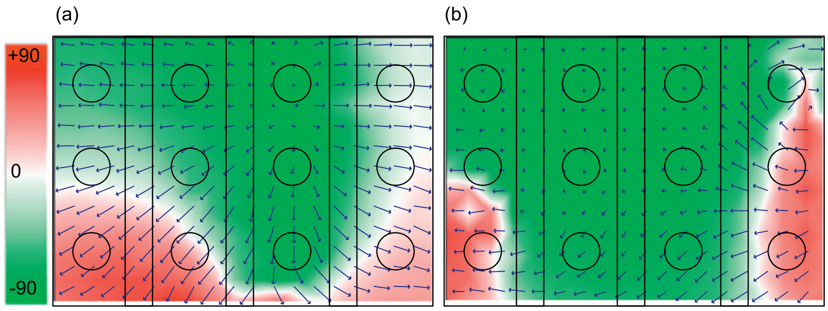

Fig. 13. 3D sound intensity vectors in a plane between the primary source and the secondary source array. The two figures show, respectively, the sound intensity map in the 160 Hz third of octave when the DAB is turned off (a) and on (b). Arrows symbolize the direction of sound intensity, while the background colour indicates the magnitude of the intensity component perpendicular to the scan area.

or without the DAB (with the DAB the sound power flowing toward the partition increases by 1.3 dB). This indicates that the contribution of the ANC on the enclosure acoustic energy is nearly negligible. Such a result was confirmed by an additional sound power analysis performed by scanning the three side walls of the source room. In this case, the contribution of the DAB to the energy incident on the walls was also small (less than 2 dB).

4.2. Comparison with simulation results

Numerical results of regression analyses cannot be directly compared to those of the simulations described in Ref. [7] for different reasons: the two factorial designs of experiments are different and many variables, whose effect has been analyzed in Ref. [7], have been neglected here. In addition, due to practical reasons, factor levels have been changed and some hypotheses undertaken in the formulation of the model (for instance, the uniform reflection coefficient) are not valid for our experimental setup. Nevertheless, a crafted comparison of the main outcomes of the two papers can be useful and will be henceforth presented.

Simulations showed that the barrier performances and the diagonal control stability were mainly driven by the distance between the loudspeakers and the error microphones and by the secondary source directivity; minor effects were due to the wall reflectivity and to the active barrier grid meshing; the influence of the primary source position could be neglected. Experimental results reported here have shown that factors affecting barrier performances are the directivity of the secondary sources and the distance between these sources and the error microphones. The effects of source position and of the reverberation time are less important. From this point of view, the simulations and experiments are consistent with each other, since they both predict the importance of the secondary source radiation characteristics and of the separation distance between secondary sources and error microphones. Both simulations and experiments indicate that the IL of the barrier is usually larger in heavily damped rooms, thus confirming that ANC is more effective when there are fewer acoustic modes excited in the room. In this paper it was shown that the proposed configuration becomes less efficient at frequencies above 200 Hz. This aspect is consistent with simulation results that illustrated that pressure attenuation was decreasing with the frequency. Finally, the sound power analyses presented in the previous section also agrees with the simulations: in Ref. [7] it was stated that the disturbance that the DAB introduces in the noncontrolled part of the enclosure could be considered negligible in most applications (averaged on the whole enclosure volume in the range of -2 to -4 dB); similar results have been found in the present study.

5. Conclusions

In this paper, a method that uses a diagonal active barrier to increase the sound isolation of an enclosure wall was developed and found to be effective. Barrier performances, obtained upon varying several factors describing the enclosure and the acoustic configuration of the barrier, have been statistically analyzed. Tests performed in a small enclosure showed that it was possible to produce a control insertion loss of up to 20 dB at some discrete frequencies. The best attenuation was usually achieved with dipole-like secondary sources, which generally give greater stability in the decentralized control algorithm. The diagonal active barrier seems to provide a promising approach to sound transmission control between two neighbouring rooms or between an enclosed noisy workplace and an open space. The proposed approach is particularly efficient if the sound transmitted through the partition is considerably larger than the structure-borne flanking sound. In this case, the method has several advantages over other active sound control techniques: first of all, as shown by the experimental results, the insertion loss do not depend very strongly on the characteristics of the room or on the position of the primary noise source, especially if the secondary source have the classic monopole-like radiation characteristic. In addition, the number of actuators can be as large as required due to the simplicity of the diagonal control system; this also makes it possible to treat large surfaces. Lastly, with this method, it is not necessary to place any transducers or actuators on the structure. Further studies are now required to enhance the directivity pattern of the secondary sources, and thus to improve the transmission loss augmentation which can be obtained with this method.

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