Sound focusing in rooms: The time-reversal approach

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New perspectives in audible range acoustics, such as virtual sound space creation and active noise control, rely on the ability of the rendering system to recreate precisely a desired sound field. This ability to control sound in a given volume of a room is directly linked to the capacity to focus acoustical energy both in space and time. However, sound focusing in rooms remains a complicated problem, essentially because of the multiple reflections on obstacles and walls occurring during propagation. In this paper, the technique of time-reversal focusing, well known in ultrasound, is experimentally applied to audible range acoustics. Compared to classical focusing techniques such as delay law focusing, time reversal appears to considerably improve quality of both temporal and spatial focusing. This so-called *super-resolution* phenomenon is due to the ability of time reversal to take into account all of the different sound paths between the emitting antenna and the focal point, thus creating an adaptive spatial and temporal matched filter for the considered propagation medium. Experiments emphasize the strong robustness of time-reversal focusing towards small modifications in the medium, such as people in motion or temperature variations. Sound focusing through walls using the time-reversal approach is also experimentally demonstrated. © *2003 Acoustical Society of America.* [DOI: 10.1121/1.1543587]

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I. INTRODUCTION

Most challenging applications in audible range acoustics, such as virtual sound space creation and active noise control, rely on the ability of the rendering system to recreate precisely a desired sound field.[1] This accuracy in rendering has to be achieved both temporally and spatially in a finite volume around the desired listener. From a theoretical point of view, the ability to recreate such a desired sound field is directly linked to the capabilities of the system in terms of spatial and temporal focusing. The propagation medium in practical situations is *a priori* unknown and complex (e.g., multiple reverberations, absorption effects, etc.) thus requiring focusing to be adaptive. Current approaches to achieve this adaptive sound control consider modeling of the propagation medium. Other signal-processing approaches based on the solution of the inverse problem have also been proposed to achieve this focusing objective.[2]

Adaptive focusing of acoustic waves is also an important research topic in domains such as medical imaging and therapy, underwater acoustics, and nondestructive evaluation. The time-reversal method for such applications has been proven a very efficient and easy technique to achieve adaptive focusing regardless of the complexity complex the propagation medium.[3] The aim of this paper is to study both theoretically and experimentally the potential of this technique for audible range acoustics.

The time-reversal focusing technique is based on the time-reversal invariance of the wave equation. In the case of room acoustics, the wave equation can be written in a fairly simple form: Propagation occurs in a lossless fluid with a constant sound velocity c in a medium bounded with nondissipative boundaries (in an ideal case). Thus, for the acoustic pressure field $p(\mathbf{r},t)$ in a transient regime, the propagation of sound obeys the condition[4]

$$\nabla^2 p - \frac{1}{c^2}\frac{\partial^2 p}{\partial t^2} = 0. \tag{1}$$

This equation is time-reversal invariant because it contains only second-order derivatives with respect to time. Consequently, for each burst of sound $p(\mathbf{r},t)$ diverging from a source which can then possibly be reflected, refracted, or scattered within the propagation medium, there exists in principle a set of waves $p(\mathbf{r},-t)$ that retraces precisely all of these complex paths and converges simultaneously at the original source site as if time were running backwards.

By taking advantage of the Huygens principle, this time-reversal wave can be obtained by measuring the field $p(\mathbf{r}_S,t)$ on a surface enclosing the experimental volume. This surface is covered with microphones that detect the field $p(\mathbf{r}_S,t)$ during a time T sufficiently long for the wave to vanish. Once this field is stored in memories, the microphones are then replaced by loudspeakers and the surface of loudspeakers reemits the time-reversed signals $p(\mathbf{r}_S,T-t)$. The resulting wave field converges optimally towards the initial source exactly as if the scene was played backwards.

In nondissipative media, such a time-reversal cavity acts as an inverse filter of the diffraction transfer function that relates the wave-field propagation from the source to the closed surface: The movie of the diverging wave propagation is played backwards resulting in an optimal focusing at the initial source location. However, this filter is not perfect because evanescent waves[5] emitted by the source (high spatial frequency content of the wave field) cannot be recorded on

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the surface and thus cannot be time reversed. Diffraction acts as a low-pass filter of the spatial frequencies, and it leads to the classical diffraction limit that prevents the size of the time-reversed focal spot from being smaller than half the wavelength. Thus, details of the source smaller than the shortest wavelength are lost during a time-reversal experiment. So, for an ideal point source emitting a wideband pulse, the ''returning'' field refocuses on it with a spot whose dimensions is on the order of the smallest wavelength. For the same reasons, the transducers covering the cavity surface do not need to be uniformly distributed over the whole surface. Because the smallest details are filtered by diffraction, the surface may be sampled by a finite number of transducers distributed on a two-dimensional array with spacing equal to half the smallest wavelength.

This time-reversal cavity remains an idealized concept which is difficult to implement in practice. The strongest limitation is linked to the difficulty of surrounding the focal region by a huge set of transducers. In ultrasound-based medical or nondestructive evaluation applications, a time-reversal mirror (TRM) consisting of a simple linear array is generally used. However, in most cases, its limited aperture reduces the focusing capabilities of the technique.[3,6]

In room acoustics, propagation occurs in a closed cavity with reverberating boundary conditions. This specific configuration leads to interesting properties that can be anticipated from previous results obtained in an ultrasonic frequency range. Indeed, in lossless ultrasonic waveguides,[7] Roux *et al.* have shown that time reversal benefits from the multiple reflections that occur during propagation, permitting the creation of a virtually infinite transducers array. Consequently, the focal resolution of the array is improved in comparison with the focal resolution of the system in free space. For the case of a closed and chaotic 2D cavity,[8] Draeger *et al.* have even shown that the information contained in the reflections of the wave field coming from a single source is sufficient to recreate a complete virtual time-reversal cavity. Thus, a single loudspeaker time-reversal mirror is sufficient to achieve focusing in such a closed and chaotic cavity. Our configuration in room acoustics is slightly different because the studied cavity does not have a chaotic shape and is not as reverberant as the one studied in Ref. 8. However, similarities with the work done by Draeger *et al.*[9] and De Rosny *et al.*[10] can be established. See Fig. 1.

Our goal is to compare the performances of classical time-delay-law focusing and time-reversal focusing in room acoustics. This study is divided into three sections. In Sec. II, time-reversal processing is studied and experimentally achieved using a loudspeaker array inside a reverberating room. The quality of focusing is studied both spatially and temporally and the phenomenon of super resolution achieved by time reversal in the room is clearly demonstrated. In Sec. III, the loudspeaker array is placed outside of the room where focusing is to be achieved. This second experiment allows study of the robustness of the time-reversal focusing process in nonideal conditions where the attenuation due to the walls causes one to reconsider the initial assessment of time-reversal invariance. Finally, the digital communications experiment described in the last section will allow us to

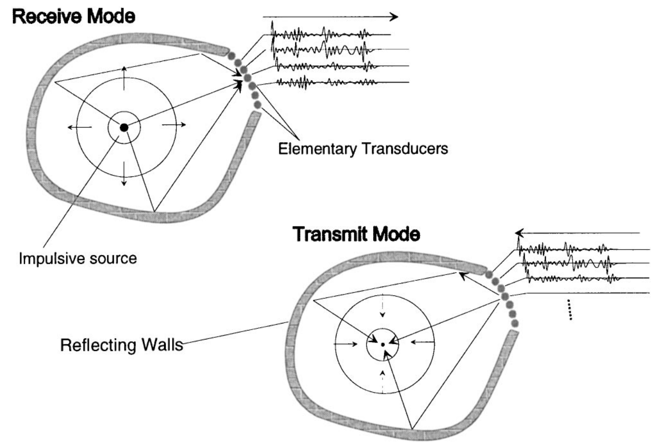

FIG. 1. Time-reversal principle in a reverberating cavity.

quantify the capacity of time reversed to improve the quantity of information that can be transmitted in a reverberating medium.

II. TIME REVERSAL IN A REVERBERATING ROOM

A. Experimental setup

Experiments have been carried out in a typical room. The room is neither a white reverberating room nor an anechoic room. By choice, we use a room where many people work on a variety of experiments and, consequently, with many obstacles to sound propagation such as tables or computers. This will prove helpful to study robustness of the different tested focusing methods. The typical applications of interest concern speech; for this reason, our study is limited to the bandwidth in the audible frequency range [300–4000 Hz].

The room in which experiments are undertaken is rectangular, with dimensions 4.7×5.9×3.1 m. The focusing system consists of a 20-loudspeaker linear array (Fig. 2), parallel to the measurement bench. The receiving system consists of an electret microphone that scans the pressure field gen-

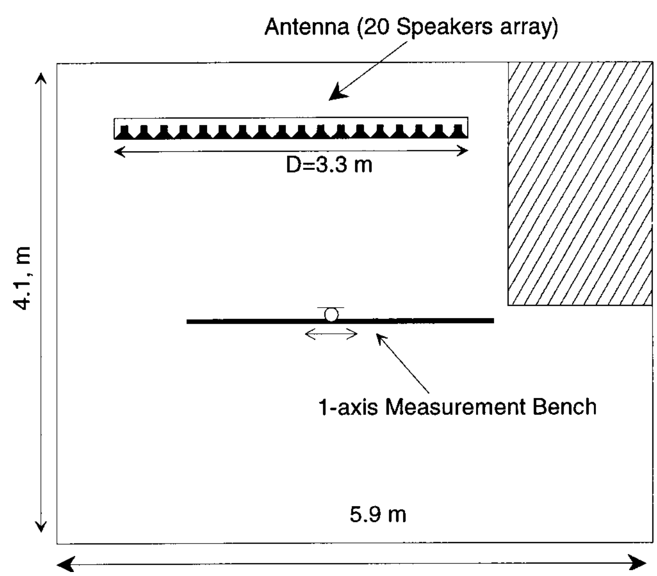

FIG. 2. Setup for experiments in reverberating room.

erated by the antenna along its measurement axis (see Fig. 2). This simple configuration has been deliberately chosen to avoid problems due to modification of the sound field by the measurement system. Emitting and receiving systems are controlled from an electronic device including 12-bit A/D converters for microphones, 256 kBytes memory for each of the 50 time-reversal channels, and 12-bit D/A converters and amplifiers for each loudspeaker. Sampling frequency is set to 20 kHz. This sampling frequency limits us to work in the 0–5-kHz frequency range because focusing quality is linked to the achievable precision in phase between each channel.[11]

Our goal in these experiments is to compare the performances of classical time-delay-law focusing and time-reversal focusing. For practical reasons, there are no active sources at the desired focal point contrary to the usual time-reversal experiment. Indeed, the first step of the time-reversal experiment is conducted using the spatial reciprocity property of the medium: instead of acquiring Green functions $G(\mathbf{Y}_i,\mathbf{r}_{fp},t)$ by emitting from the desired focal point $\mathbf{r}_{fp}$ and receiving at location $\mathbf{Y}_i$ of each transducer of the array, we acquire Green functions by letting each loudspeaker emit individually assuming that $G(\mathbf{Y}_i,\mathbf{r}_{fp},t)=G(\mathbf{r}_{fp},\mathbf{Y}_i,t)$. This method avoids having to use a complex configuration of transducers consisting of a couple of coaxially mounted loudspeaker and microphone for each element of the array.

In the audible range, electro-acoustical response of transducers cannot be neglected, especially for loudspeakers. Impulse responses from an element i of the array to the focal spot are measured by emitting a chirp of bandwidth 100–4100 Hz with the loudspeaker i and acquiring the corresponding signal with a microphone located at the focal point $\mathbf{r}_{fp}$ (the use of a chirp signal excitation instead of a delta function is only due to sensitivity considerations). Impulse responses are then obtained by correlating the signal with the initial chirp, and is given by

$$h_{i,fp}(t)=h^{\mu}_{\mathrm{AE}}(t)\overset{*}{_t}G(Y_i,r_{fp},t)\overset{*}{_t}h_{\mathrm{EA}}(t), \tag{2}$$

where $h_{\mathrm{EA}}(t)$ is the electroacoustic impulse response of loudspeaker i (it will be considered identical for each loudspeaker in the present paper), $\mathbf{Y}_i$ is the position of the ith loudspeaker, $h^{\mu}_{\mathrm{AE}}(t)$ is the acousto-electric impulse response for the microphone, and $\overset{*}{_t}$ denotes convolution with respect to time. Due to the linearity of the problem, impulse response of the electronic system and transducers will be considered as a whole, denoted $h^i_E(t)$. Thus

$$h_{i,fp}(t)=G(Y_i,r_{fp},t)\overset{*}{_t}h^i_E(t). \tag{3}$$

These impulse responses are time reversed and reemitted in the medium by the set of loudspeakers, using the same electronic system. The signal received at location $\mathbf{x}$ in the medium can be written

$$h_{RT}(x,t)=\sum_{i=1,\ldots,N}G(Y_i,x,t)\overset{*}{_t}G(r_{fp},Y_i,-t)\overset{*}{_t}\times h^i_E(t)\overset{*}{_t}h^i_E(-t), \tag{4}$$

where $\mathbf{x}$ is a point in the medium. In the case $\mathbf{x}=\mathbf{r}_{fp}$ (i.e., at the focal point) and $t=0$, $h_{RT}(x,t)$ is maximized: Eq. (3) illustrates the fact that time reversal acts as a spatial and temporal matched filter, both for electronic response $h_E(t)$ and for the propagation operator $G(\mathbf{Y}_i,\mathbf{r}_{fp},t)$. This property is emphasized in the next two parts.

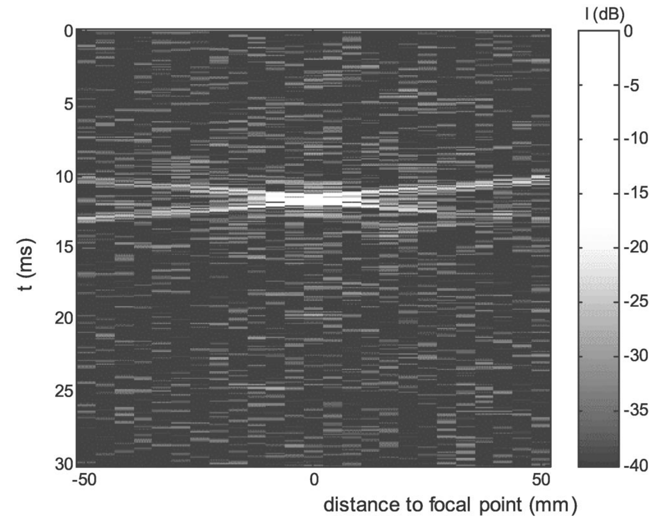

FIG. 3. Intensity of the spatio-temporal focusing pattern achieved by time-reversal processing.

B. Spatial focusing

The performances of the two focusing techniques can be compared using spatial domain point spread functions (PSF), defined as $d(\mathbf{x})=\max_t\{h(\mathbf{x},t)\}$, where $h(\mathbf{x},t)$ is the impulse response obtained at location $\mathbf{x}$ after the focusing process. To obtain the PSFs for the different cases, initially, the set of impulse responses $h_{i,fp}(t)$ relating the loudspeakers noted i and the focal point f_p is acquired. First, the classical focusing technique (time-delay law) is realized by deducing from the experiments the travel time between each loudspeaker and the microphone located at the focal point. Focusing is then achieved by emitting the same signal on each loudspeaker with different delays corresponding to inverse of the previous travel time-delay law. Thus, the signal emitted by each loudspeaker arrives at the same time at the focus [in practice, this time-delay law is computed using the position of the maximum of correlation between $h_{i,fp}(t)$ and $h_E(t)$]. Second, signals to be emitted for time reversal are easily deduced by reversing the set of the $h_{i,fp}(t)$ impulse responses with respect to time.

The set of impulse responses $h(\mathbf{x},t)$ obtained in the focal plane after the focusing process is measured by translating the microphone along the $\mathbf{x}$ axis (see Fig. 2). A typical spatio-temporal response obtained along the focal axis $\mathbf{x}$ by time-reversal processing is presented in Fig. 3.

1. Focal spot width

Figure 4 presents the PSF around the desired focal point obtained with the partial antenna. By comparing PSFs obtained by time reversal with that obtained by the classical delay-law focusing technique, the improvement introduced by time reversal is evident: the main lobe is sharper, and sidelobes are 7 to 10 dB lower.

The dashed intermediate curve corresponds to the simulation of time-reversal focusing when propagation occurs in

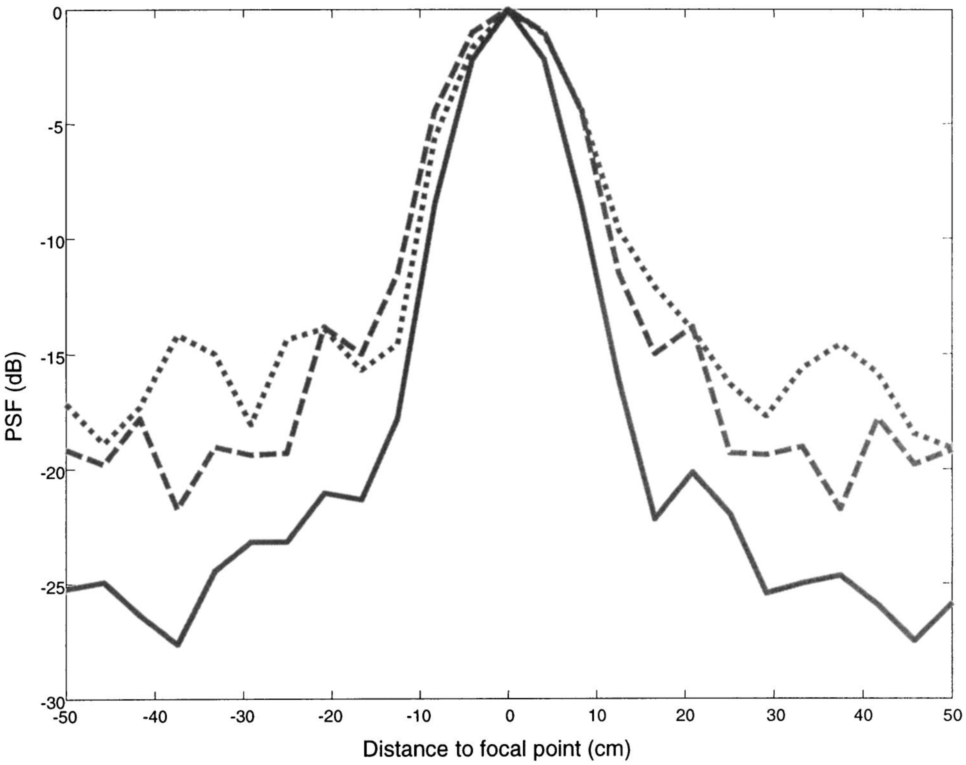

FIG. 4. Point spread functions obtained with the partial antenna: Delay law (dotted line), TR in free space simulation (dashed line), and experimental TR in the room (solid).

free space. This result is obtained by assuming that transducers are point-like. In this situation, Green's function relating to point **x** and **y** can be written as

$$G(\mathbf{x},\mathbf{y},t)=\frac{1}{\|\mathbf{xy}\|}\delta\left(t-\frac{\|\mathbf{xy}\|}{c}\right),$$

where $\|\mathbf{xy}\|$ is the distance between points **x** and **y**, and $\delta(t)$ is the delta function. Substitution of this formula in Eq. (3) leads to

$$h_{RT}(x,t)=\sum_{i=1}^{N}\frac{1}{\|xY_i\|\|r_{fp}Y_i\|}\times\delta\left(t-\frac{\|xY_i\|-\|r_{fp}Y_i\|}{c}\right)\overset{*}{_t}h_E^i(-t)\overset{*}{_t}h_E^i(t). \tag{5}$$

A good approximation of the impulse response of electronic system and transducers $h_E(t)$ is measured in an anechoic room using the actual system; calculation of $h_{RT}(\mathbf{x},t)$ is then straightforward. The corresponding dashed curve in Fig. 4 demonstrates the advantages of time-reversal process. Time reversal in free space provides results that are very similar to those obtained with the classical delay-law focusing. The slight improvement provided by time reversal in such a situation is due to the temporal matched filtering of the impulse response of the electronic system $h_E(t)$. In the case of propagation in a reverberating medium, such as our room, time reversal gives results that appear to be even better than those obtained in free space.

This phenomenon, called *super-resolution*, is also observed in waveguides[7] and in closed cavities.[8] After time reversal, the rays that arrive at the focal point appear to be emitted from multiple sets of virtual sources. These sources are the images of the actual emitting antenna with respect to the different interfaces in the rooms (walls or furniture). The superposition of the signals coming from each virtual antenna at the focal point leads to a virtually infinite antenna. Thus, focusing by time reversal in a highly reverberant room can be assimilated to near-field focusing, and the focal spot width does not depend on the size of the antenna, but is always equal to $\lambda_0/2$, where λ_0 is the central wavelength of the used signal. In the present experimental case, we can clearly check that the focal spot has a lateral width of $\lambda_0/2 = 17$ cm, corresponding to the central frequency of the loudspeakers, 975 Hz.

In Fig. 5 the size of the focal spot is plotted versus the

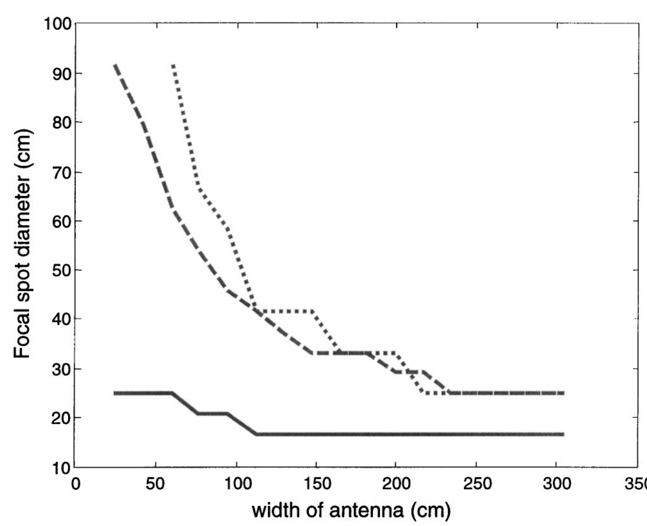

FIG. 5. Focal spot width vs antenna diameter for the delay law (dotted line), TR in free space simulation (dashed line), and TR in the room (solid).

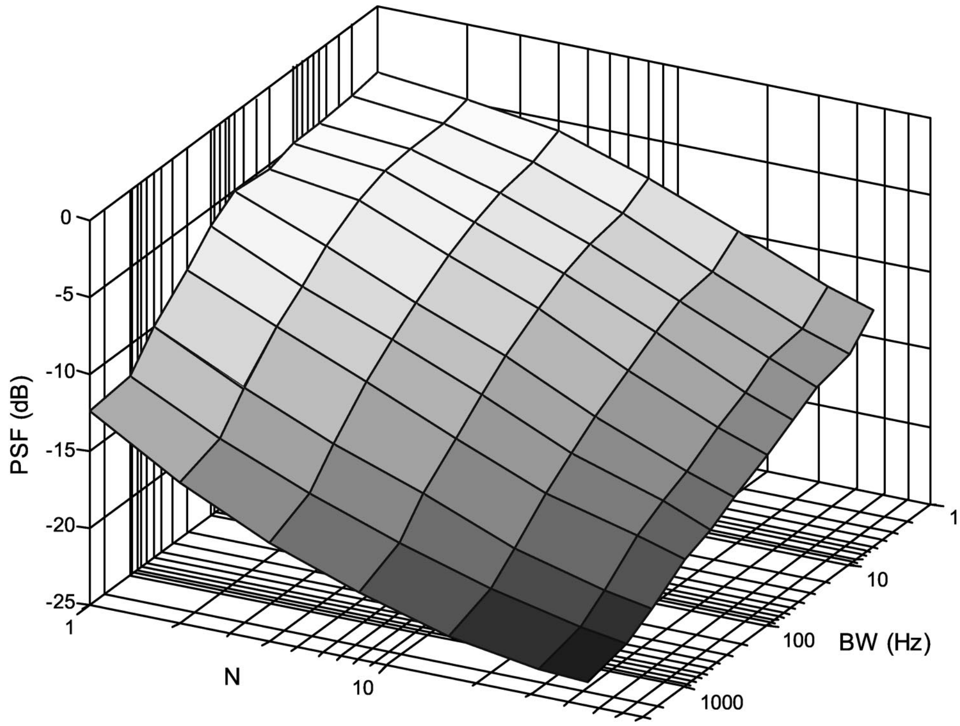

FIG. 6. Sidelobe level $SL_{(dB)}$ as a function of the number of used loudspeakers N and of the bandwidth BW. Value is the average of 32 repetitions.

size of the antenna for the different focusing techniques. As stated above, the size of the focal spot remains constant for time-reversal focusing in a reverberating room corresponding to the optimal focus size $\lambda_0/2$. In the case of focusing in free space, the focal spot width follows the classical diffraction relation $d=\lambda_0(F/D)$, where F is the distance between antenna and focal point, and D the diameter of the antenna. One can clearly notice that the super-resolution phenomenon is experimentally checked as soon as the antenna is larger than $3\lambda_0$ (i.e., 100 cm). The slight variations observed for smaller antenna diameters on the focal spot width obtained by time-reversal processing show the limits of the virtual antenna approach: the virtual antenna is infinite if the loudspeakers are omnidirectional and if no attenuation occurs during reflections, which is not the case in reality. After each reflection on the wall, the amplitude of the reverberated signal is smaller and it results in a corresponding virtual source of lower efficiency.

2. Secondary lobes level

Beside the importance of obtaining a narrow focal spot, sidelobe level is the second parameter to take into account when evaluating the performance of a time-reversal focusing system. This parameter has been studied by Roux *et al.*[7] for the case of a waveguide, where a simple interpretation of the use of reflections by time reversal in terms of virtual images is introduced. In this article, sidelobe levels are shown to be dependent on the number of transmitters and on the dimensions of the waveguide. More complete analysis is available in Ref. 5. In the present case, other parameters require consideration, such as the available bandwidth or the size of the temporal window used for time-reversal processing. The subjective effect of sound focusing for the human ear is mainly linked to two parameters: quality of sound at the focal point (which depends on the quality of the impulse response after focusing), and the relative level of the first sidelobe (dynamic of the focusing). Thus, we introduce the sidelobe level as $SL_{(dB)}=\langle 20\log_{10}[d(y)/d(0)]\rangle$, where $d(y)$ is the PSF and y varies between $\lambda_0/2$ and $2\lambda_0$.

Figure 6 represents the evolution of $SL_{(dB)}$ with the number of loudspeakers and the bandwidth used during the time-reversal experiment. In order to understand this evolution, we consider four aspects of the function SL (N,BW).

(i) In the center of the figure, $SL_{(dB)}$ evolves in a bilinear way: $SL_{(dB)}=A\log_{10}(N)+B\log_{10}(\mathrm{BW})+C$, where N is the number of loudspeakers, BW the effective used bandwidth, and A, B, C are constants. This point can be explained considering that, due to their different reverberations, the contributing signals coming from each loudspeaker are uncorrelated; thus, they will sum coherently at the focal time, and incoherently elsewhere, leading to a ratio: $d(y)/d(0)\propto\sqrt{N}$. In frequency domain, the number of excited modes is proportional to the bandwidth. Similarly, it may be considered that modes are uncorrelated and contribute incoherently except at focal point. Here again, this leads to a relation: $d(y)/d(0)\propto\sqrt{\mathrm{BW}}$. This bandwidth dependence was addressed by Draeger *et al.* for the case of chaotic cavities.[8,9] Substitution in the above formula gives: $St_{(dB)}=10(\log_{10}(N)+\log_{10}(\mathrm{BW}))+C$. This simple approach provides a coarse approximation of the surface. In fact, relation between bandwidth and number of excited modes appears to be more complicated.

(ii) When bandwidth becomes very small, the excitation is quasimonochromatic. In this case, point spread function in free space is of the form $d(y)=[\sin(2\pi\lambda/Dy)]/[2\pi\lambda/Dy]$. This PSF leads to a

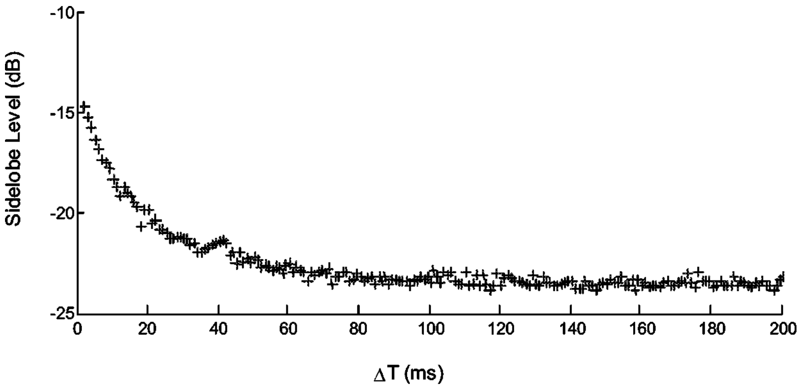

FIG. 7. Dependence of sidelobes level $SL_{(dB)}$ on quantity of data ΔT used for time reversal.

minimum sidelobes level of -13 dB; this agrees with the minimum value of $SL_{(dB)}$ obtained experimentally when N increases for small bandwidths.

(iii) For large bandwidths, $SL_{(dB)}$ seems to reach a limit. This an artifact of the experiment, corresponding to the fact that one bandwidth is limited by the electronic system itself, and not by the applied filter.

(iv) When $N=1$, it can be noticed that $SL_{(dB)}$ reaches a maximum of -4 dB as the bandwidth decreases. Thus, even a single loudspeaker is able to achieve focusing by using the multiple reverberations in the room. This situation is explained with a simple model: the virtual images of the loudspeaker with respect to the different interfaces in the room allows creation of a spatially undersampled antenna. The focusing pattern corresponds to the focusing quality obtained with such an undersampled antenna with monochromatic excitation in free space. A simple model is established considering that time reversal in this situation creates a virtual antenna of transducers with a step equal to the width of the room (5.9 m). The PSF is then calculated using the Huygens–Fresnel principle.[11] In conclusion, even with one loudspeaker and a monochromatic excitation, a minimal focusing does exist as the time-reversal approach benefits from the multiple reflections.

Another interesting point is the quantity of data needed to achieve a good quality of focusing. In Fig. 7, $SL_{(dB)}$ is represented against the size of the temporal window used during the time-reversal process. This figure denotes the evolution from small ΔT (where focusing is very similar to the one obtained with a delay-law technique) to large ΔT for which all available reflections are taken into account in the time-reversal operation. Draeger *et al.*[8] show the existence of a saturation level for sidelobes in regards to ΔT, linked to the fact that not all the modes of the cavity can be equally excited and that this saturation time can be related to the Heisenberg time. However, the saturation time observed in our case cannot be interpreted this way: First, our cavity is

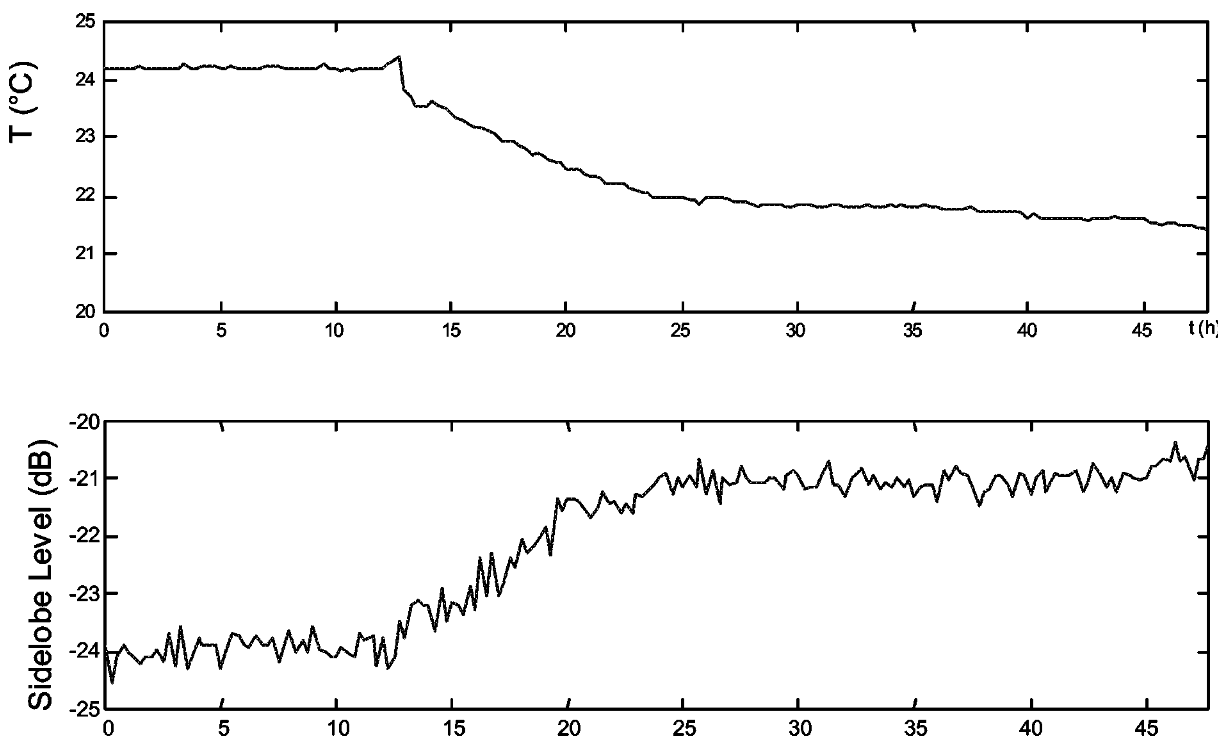

FIG. 8. Evolution of sidelobe level SL(dB) over a 48-h period (bottom) and the corresponding temperature evolution (top).

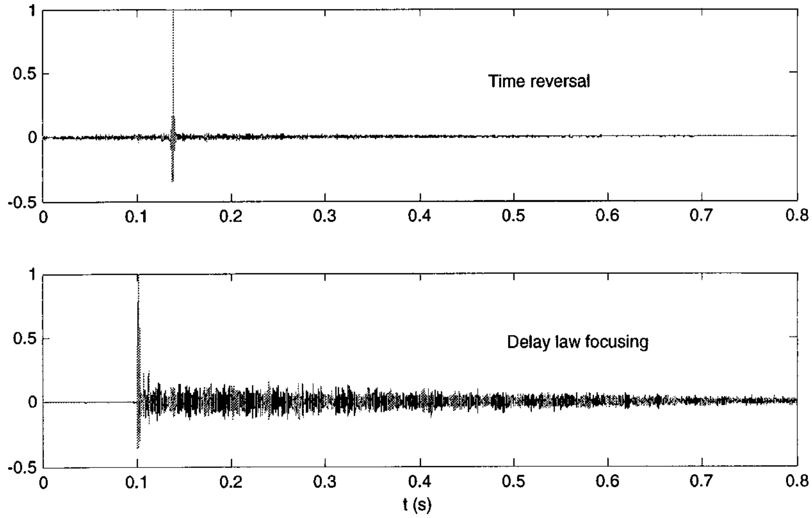

FIG. 9. Temporal signal received at focus for delay-law focusing (bottom) and for time reversal at the focal point $h_{RT}(r_{\mathrm{fp}},t)$.

not strictly speaking chaotic. Moreover, observation of such a saturation phenomenon requires a very low attenuation during reflections and signal-to-noise ratios that are impossible to obtain in a standard room and generally in audible acoustics contrary to the silicon wafers used in Ref. 8. In our case, emitted signals are shorter due to attenuation: thus, the observed saturation is due mainly to the signal falling below the noise level.

3. Influence of changes in the propagation medium

In practical systems, focusing needs to be robust: focusing has to be achieved in media where objects will eventually move, and conditions of the propagation can vary. The influence of such an evolution upon focusing is measured by acquiring an initial set of signals and calculating the corresponding set of time-reversed signals to emit. This set is then reemitted every 15 min in order to study the evolution of the sidelobe level $\mathrm{SL}_{(\mathrm{dB})}$ against time. Results are presented in Fig. 8. During the measured period (48 h), people move and work in the room, creating slight modifications of the propagating medium. In these conditions where temperature remains constant (variations do not exceed 0.5 °C), focusing through time reversal proves to be very robust. However, time-reversal focusing is very sensitive to cumulative variations of the propagation medium characteristics. Figure 8 shows that for example focusing quality has a strong dependence with temperature (i.e., with variations of sound speed). This problem can easily be overcome in practical applications by using banks of data corresponding to the different conditions and link the choice of the data bank to the temperature in the room.

C. Temporal focusing

In the time domain, results presented in Fig. 9 show the effect on time reversal for the compression of the impulse response of the room. As shown in Eq. (2), the response obtained at the focal point corresponds to the coherent sum of the autocorrelation of the impulse responses for each loudspeaker. For this reason, the response is symmetric and is a good approximation of a delta function. However, the shape of the temporal response at the focal point for the two focusing techniques appears to be very different: Time reversal optimizes the ratio between the main focusing lobe and sidelobes in a symmetric way, whereas delay-law techniques concentrate noise after the main lobe, giving a noisier but more natural response, since it is similar to what would be obtained for the propagation of a single pulse in the room.

Thus, from the temporal point of view, time reversal appears to give an interesting result in term of S/N ratio. This result is directly linked to the fact that time reversal provides both a temporal[12] and spatial[13] matched filtering, corresponding to the real experimental conditions. However, the symmetrical form of the impulse response at the focal point, although optimal in terms of S/N ratio, makes the system difficult to use for audio applications. Indeed, the sidelobes preceding the main lobes create strange perspective effects,

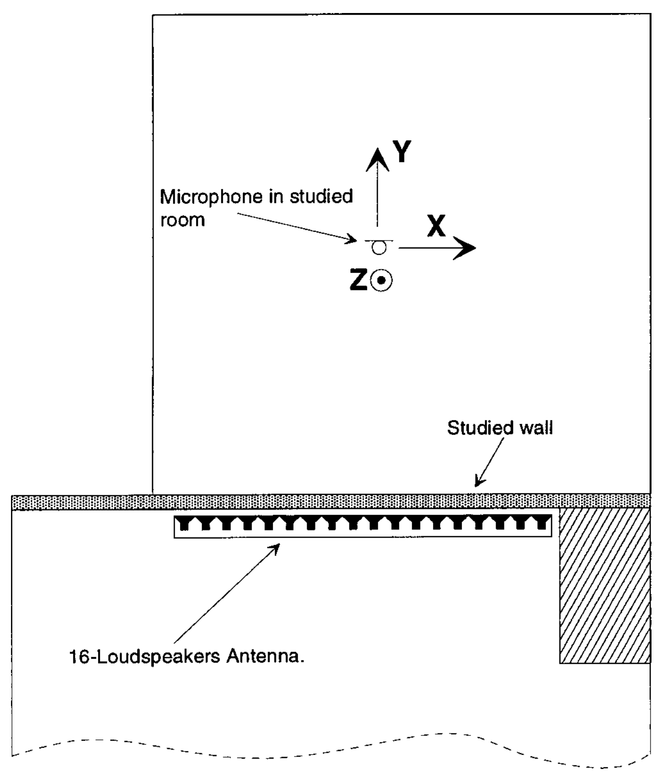

FIG. 10. Experimental setup for focusing through the wall.

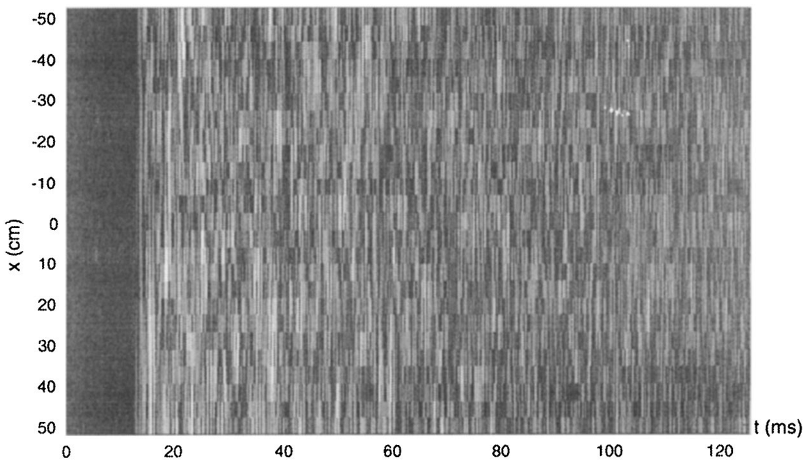

FIG. 11. B scan for the signal coming from loudspeaker No. 8: $h_8(x,t)$.

especially with percussive sounds and plosives consonants, whose rendering quality is poor.

Experiments conducted in the room demonstrated the efficiency and robustness of time reversal compared to the classical time-delay technique to achieve focusing in a reverberant room, both in terms of temporal compression and spatial focusing. A more challenging configuration for time reversal is investigated in the next section.

III. TIME REVERSAL THROUGH A WALL

Focusing sound in a reverberant room appears to be a nearly ideal situation for time reversal: Attenuation is due mainly to reflections, and remains quite small. In order to study the robustness of the time-reversal process for audible range acoustics, the focusing experiment described in the preceding section is now conducted under the conditions where attenuation is strong and propagation much more complex. The experimental setup is described in Fig. 10: The 20-loudspeaker antenna array is now directed towards the wall and we try to focus through the wall in the other room.

Propagation of acoustical energy from one room to the other is complex, because sound propagates through multiple paths, such through the wall, of course, but also through doors and windows. As shown in Fig. 11, there is no direct wavefront, and no specific path for propagation can be identified. In the conditions of experiment, the acoustic attenuation (acoustic attenuation is defined as the difference in mean sound levels in each room for white noise) between the central loudspeaker of the antenna and the central position of the microphone is 20 ± 3 dB. The wall by itself is made of hollow bricks and plaster, and is 8 cm thick. Such a wall provides a normalized acoustic attenuation of $\sim$41 dB.[14] Thus, the most important part of acoustical energy that manages to reach the focal point comes from "leaks," probably via doors and windows but also from lateral propagation through walls. The problem is nevertheless very interesting because the attenuation is strong and strongly frequency dependent, as shown in Fig. 12. (For example, some cuts are down 30 dB.)

Those conditions are very tricky for time reversal. Indeed, its ability to focus should be degraded by attenuation, as the technique is based on the fact that no dissipation occurs during propagation. Moreover, time reversal does not provide any compensation for the amplitude of the frequency response, and this point will necessarily lead to reductions in the focusing quality.

A. Focusing experiments

A typical PSF obtained in the configuration described above is plotted in Fig. 13. The results obtained in such a configuration are very similar to those obtained in a simple room, especially in terms of focal spot width: -6-dB width is equal to $\lambda/2$, where λ corresponds to the central frequency of the loudspeakers. On the other hand, secondary lobe level is higher (around -18 dB) than the one obtained for an equal number of loudspeakers in the room. This phenomenon is

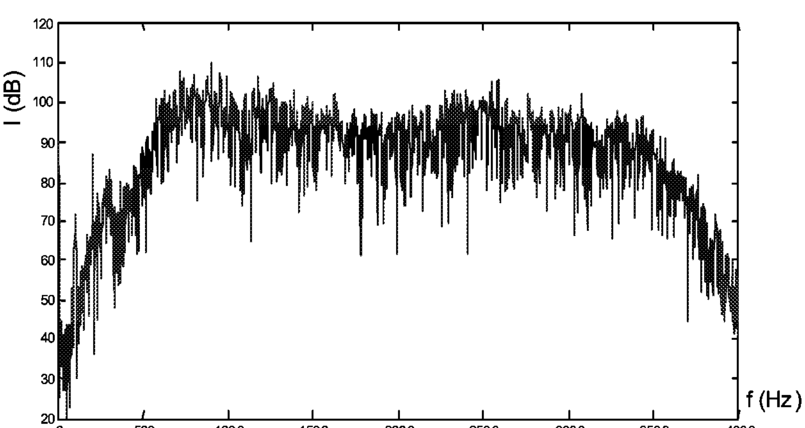

FIG. 12. Spectral density of the impulse response $h_8(r_{fp},t)$ between loudspeaker No. 8 and the desired focal point.

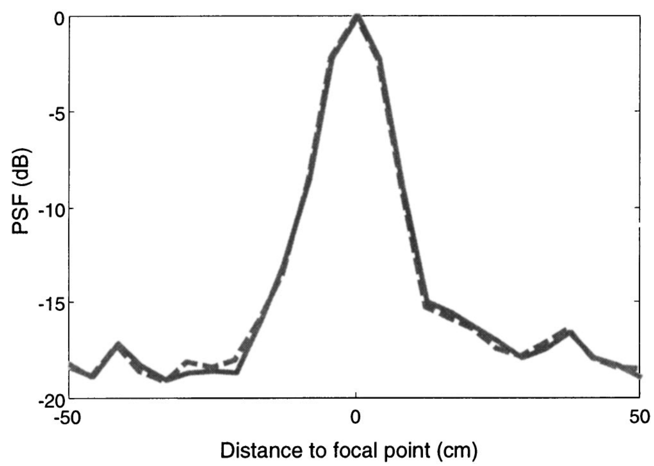

FIG. 13. Point spread functions obtained with time reversal (dashed line) and time reversal with amplitude compensation (solid line).

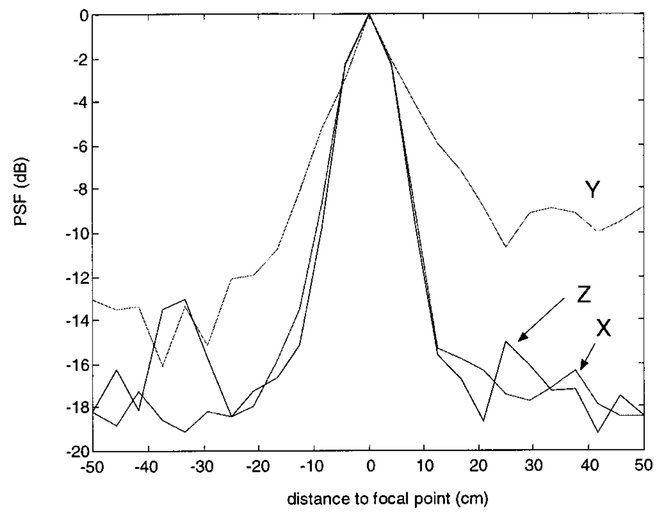

FIG. 15. Point spread functions for focusing through the wall along the three axes $\{x,y,z\}$.

linked to acoustical dissipation during propagation, which implies distortion in the spectrum and distortion of the overall amplitude coming from each loudspeaker. Figure 14 shows the consequences of these distortions for the overall quality of focusing: In many cases, no focusing can be achieved (sidelobe level is 0 dB or higher), whereas in similar conditions, focusing was possible when using the first experimental setup (see Fig. 6). Thus, loss of information during propagation results in less quality in spatial focusing, because the constructive interference system at the focal point is less effective.

From the point of view of temporal compression, results are very similar to those obtained with time reversal with the conditions of the first experimental setup, except that the temporal sidelobe level is higher: The level of first sidelobes cannot be lowered under -25 dB, whereas levels of -35 dB could be attained in rooms. In the meantime, as for spatial behavior, the width of temporal response is equal to the width obtained in rooms.

Previously, we have studied focusing only along one axis. But, the shape of the focal spot can be studied in three dimensions. Since time reversal benefits from the multiple reflections in the three dimensions, we hope to obtain a similar focusing whatever the measurement axis is. Moreover, focal spot width should remain the same, equal to $\lambda/2$, where λ corresponds to the resonance frequency of the loudspeakers. Sidelobe levels can vary because the virtual images of loudspeakers created by reflections leads to spatially undersampled virtual antennas.

Figure 15 presents the PSFs obtained along the three dimensions $\{x,y,z\}$: As suggested above, focusing occurs in the three dimensions. One should notice that vertical and lateral focusing (axes $\mathbf{x}$ and $\mathbf{z}$) are similar and optical ($\lambda/2$ spot size), although the antenna is parallel to the $\mathbf{x}$ axis. However, longitudinal focusing proves to be more difficult to achieve, probably because of attenuation during reflections: In this direction, focusing occurs indirectly through reflections against the opposite wall in the other room, so more reflections are needed and attenuation is stronger.

We note that similar results are obtained when the active sources are placed directly in the room. These are not presented to avoid redundancy, and because the present situation is more challenging for such an experiment.

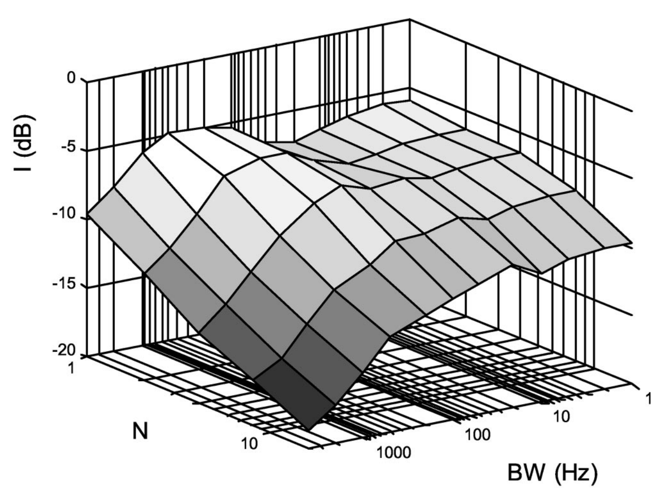

FIG. 14. Sidelobe level $SL_{(dB)}$ as a function of the number of used loudspeakers N and of the bandwidth BW. Value is the average of 32 repetitions.

B. Communication experiment

The quality of focusing obtained with time reversal through walls is too poor to be able to transmit music with good quality from one room to the other, especially because of important variations in the frequency response of the system, and because of the high level of the temporal sidelobes. However, this technique can prove very useful in a context of digital communications. For example, using the previously described experimental setup, we can transmit information in a binary form.

To achieve this transmission, a classical method of modulation and demodulation is used. In order to concentrate the emitted signals in the frequency range of the focusing system, information is modulated around a carrier frequency of 2500 Hz, using a Q-DPSK technique.[15] As it is shown in Fig. 16, measurement imprecision can be quite important both for phase and amplitude, justifying the choice of such a

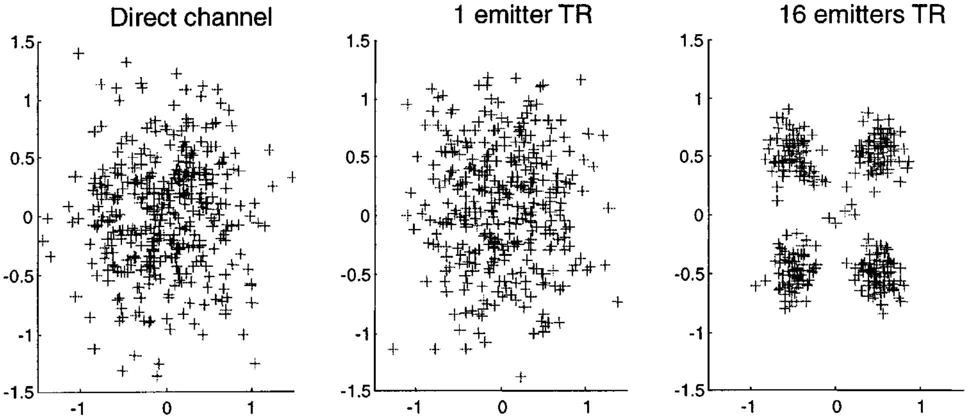

FIG. 16. Signal-space representations of the symbols acquired at the desired receiving location.

modulation system. Furthermore, the fact that modulation is differential provides an easy way to get rid of problems with baud clock synchronization.

In order to evaluate the performances of time reversal for information transmission under such circumstances, three different filtering techniques are used: First, the signal $s(t)$ obtained from the modulator is transmitted as is with one of the loudspeakers. A second technique consists of emitting the $s(t)$ signal with one loudspeaker after filtering with the time-reversed propagation response between the loudspeaker and the microphone. Thus, we obtain a temporal matched filtering of the propagation medium with a single loudspeaker. Finally, $s(t)$ is transmitted after filtering by the different time-reversal responses between each loudspeaker and the microphone. Figure 16 shows how these different transmission techniques act on the signals constellations. When compared to direct transmission or single channel matched filtering, time reversal for multiple channels provides an improvement in both phase and amplitude measurements due to the fact that the improvement given by matched filtering is coherently summed for each of the propagation paths.

Quantitative results about this experiment are obtained by simulating the transmission a succession of random bit sequences, each of them containing 25 kbits, at a band rate of 1250 symbols/s. A bit error rate (BER) is then calculated by comparing the obtained sequence and the transmitted sequence. Each measurement is repeated 512 times and the BERs are finally averaged to obtain the results of Table I. In the case of direct transmission, no detection is possible (BER≅50%): This point is directly linked to the fact that no equalization of the channel is provided.

From these results, time reversal proves to be an efficient way to provide channel equalization and to use channel diversity. Indeed, Table I shows that a simple matched filter (case of a one-loudspeaker antenna) is not sufficient in our case of interest, whereas the use of a multiple-elements antenna allows transmission to occur under varied conditions. Thus, this technique can be very useful in environments where propagation creates multiple reflections and complex distortion of the channel, even when strong attenuation is present.

TABLE I. Evolution of the bit error rate (BER) with the number of loudspeakers in the transmit antenna.

Number of elements in antenna	1	2	4	8	16
BER	47.3%	12.7%	0.12%	3.10^{-4}	$<10^{-5}$

IV. CONCLUSION

We have presented in this paper experimental results on time-reversal properties in audible range acoustics. Thanks to an optimal use of the reverberated field, due to its spatial and temporal matched filter property, time reversal achieves high focusing quality, both from the temporal and spatial points of view. In particular, by using the time-reversal approach, a loudspeaker antenna benefits from the reverberations in the room to achieve a better focal spot than in free space. This phenomenon, known as super-resolution, is experimentally demonstrated here for room acoustics. So, according to its spatial and temporal focusing ability, time reversal could be very useful to control the local sound field in a 3D volume around a focal point, even if the focal point is not in the same room as the control antenna. This technique may find numerous applications in different transmission systems, such as virtual source imaging systems for multimedia applications or devices for digital communications in complex media, as shown in the previous section.

However, its potential remains limited for audio applications due to temporal sidelobes preceding the main signal that create strange perspective effects, especially with percussive sounds and plosives consonants, whose rendering quality is quite poor. Moreover, time reversal proves to be sensitive to attenuation and spectral content of the system. Thus, as soon as the propagation medium becomes strongly dissipative or if the loudspeaker bandwidth is nonuniform, focusing quality can be degraded. In that case, more sophisticated techniques, such as inverse filtering, as proposed by Tanter[16] or Kahana,[17] can be envisioned and are currently under investigation. The application of inverse filtering to sound focusing is described and compared to the time-reversal approach in following works.

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