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MEASURING MUSICAL INSTRUMENTS DIRECTIVITY PATTERNS WITH SCANNING TECHNIQUES

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Characterizing directivity patterns of musical instruments implicitly requires measuring non-stationary sound fields. A priori, this fact implies using multichannel methods to assess any change along time at all regarded positions. However, this paper explores the advantages of using a static reference sensor close to the musical instrument to track any variations in the sound field during the measurement, allowing scanning techniques to be performed. The method developed is based on taking transfer functions between the scanning transducer and the reference sensor. Then, the number of positions measured is limited for each frequency depending on the dynamic range acquired. Experimental results of violin directivity patterns are presented along with a discussion focused on comparing the technique proposed with conventional multichannel measurement methods.

1. Introduction

Human understanding is mainly based on seeing; leading people to create many visual representations to help understanding what is going on when things cannot be seen. Sound representations have been key aids for understanding practical problems regarding noise issues. Many different techniques have been implemented over years for charting sound fields and noise sources but always with a trade-off between cost, time and accuracy.

Already from the late 1980's scan-based methods have been introduced for mapping stationary sound fields [1]. Recent works have introduced a novel scanning method called "Scan & Paint" [2–4] for measuring sound pressure, particle velocity, intensity, sound absorption and acoustic impedance in an efficient way. The properties of the sound field are determined and visualized via the following routine: while the probe is moved slowly over the surface, pressure and velocity are recorded and, at the same time, a video image is captured. Next, all data is processed. At each time interval, the video image is used to determine the location of the sensor. The absolute position of the probe is unknown, only the 2D coordinates relative to the background image are computed. Then, an acoustic color plot is generated.

Scanning methods are proven to minimize the measurement time and cost, but conventionally constrained to mapping stationary sound fields. In the literature of Near field Acoustic Holography (NAH), several signal processing techniques have been proposed to overcome some problems derived from the degree of time stationary of the source [5–7]. However, these techniques require using multiple references along with scanning microphone arrays. In contrast, this paper is focused on presenting an effective method for directly visualizing sound radiation patterns without back-propagating the sound field to the source and using only one fixed and one moving sensor.

The sound radiation patterns of a musical instrument is a classic example of non-stationary sound source. There are no standard regulations regarding the measurement procedure required for characterizing their directivity patterns due to its practical difficulties. Directivity patters play an important role on virtual acoustics, specially when computing room auralizations [8]. Therefore, finding a measurement technique which allows to characterize the sound radiated of a non-stationary source in a fast and efficient way will simplify the process remarkably.

The following sections will present the theory and methodology required to implement the proposed measurement technique. Furthermore, the directivity patterns of a violin are presented. Finally, advantages and disadvantages of the novel technique are discussed.

2. Theory

The problem addressed by this article is depicted in Figure 1. A scanning pressure microphone is moved across a planar surface which has a separation distance R with the sound source. This section derive the theoretical basis required in order to understand how to extract directivity information using only two transducers.

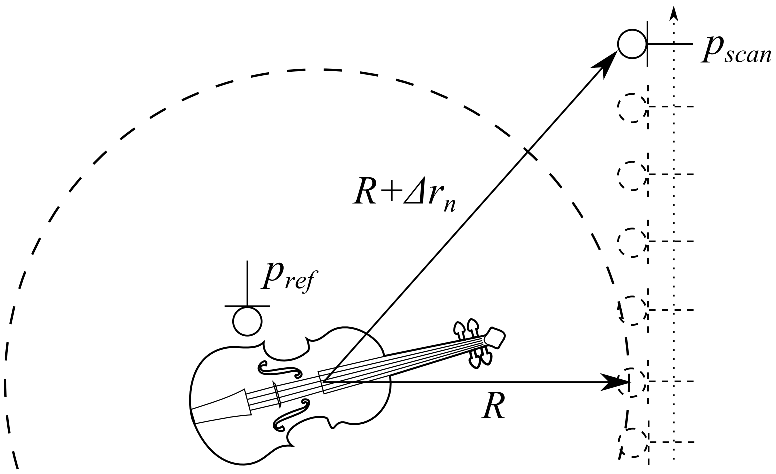

Figure 1. Sketch of the assessed problem

Any complex sound source can be represented as a set of monopoles closely distributed in the space. The superposition of two or more monopoles under free field conditions will generate a pressure sound field which can be modeled as

$$p(\theta, \varphi, r, t) = \frac{\mathbf{A}}{r} D(\theta, \varphi) e^{j(\omega t - kr)} \tag{1}$$

where $\mathbf{A}$ is a complex time independent term which relates source characteristics such as volume velocity, specific acoustic impedance and wavenumber; ω is the angular frequency; r is the distance between source center and measurement plane; and $D(\theta, \varphi)$ is a directivity term which can take arbitrary values for different radiation angles of azimuth (θ) and elevation (φ). Evaluating the above expression for simple sources such as a monopole source, $D(\theta, \varphi)$ will simplify to unity for all possible angles. In contrast, D(θ, φ) could also be expressed using first order Bessel functions if the source behaves as a baffled circular piston [9]. In contrast, musical instruments cannot be described using general analytical expressions. Consequently, suitable measurement procedures for characterizing complex sound sources are required.

Let us now evaluate Equation (1) for the two transducers shown in Figure 1. First of all, the fixed reference position can be defined such as

$$p(\hat{\theta}, \hat{\varphi}, \hat{r}, t) = p_{ref}(t) = \mathbf{B}e^{j(\omega t - k\hat{r})} \tag{2}$$

where $\mathbf{B}$ is a complex time independent term which relates source characteristics perceived at a given position $\hat{\theta}, \hat{\varphi}, \hat{r}$. It is important to highlight that the number of independent variables have been reduced but the time dependency is maintained. Secondly, it is necessary to define the projection of each of the measured pressures across the measurement plane in a sphere of radius R, i.e.

$$p_{sph}(\theta, \varphi, t) = p(\theta, \varphi, r, t)\left(\frac{R + \Delta r}{R}\right)e^{j\Delta r} \tag{3}$$

where Δr represent the euclidean distance between the corresponding sphere projection and the measurement position. Figure 2 shows an example of the sound field produced by a monopole source measured in 6 equidistant planar planes and the corresponding directivity pattern obtained after projecting the data into a sphere.

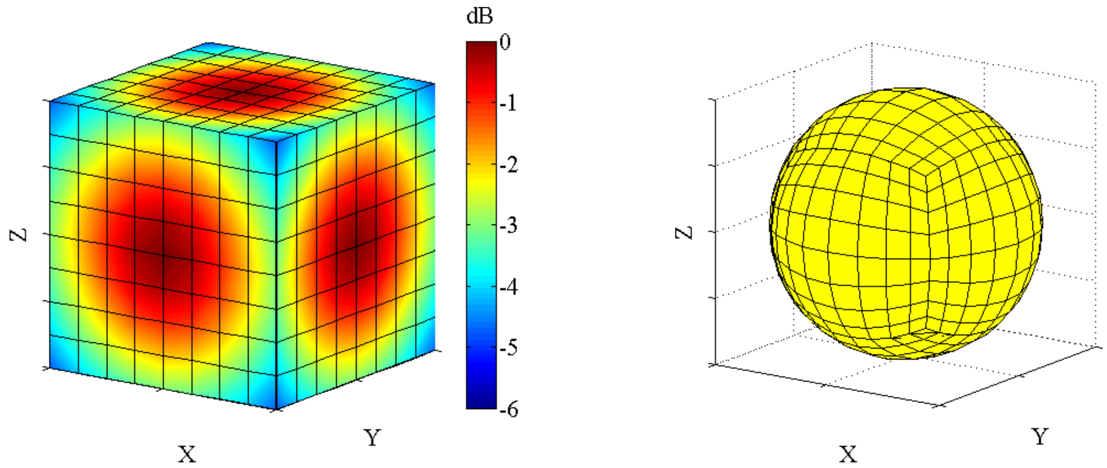

Figure 2. Example of a monopole radiattion measurements transformed to a spherical directivity pattern

So far, arbitrary signals in the time domain have been assessed. However, by defining a time harmonic excitation we can relate Equation (2) and Equation (3) in the frequency domain by computing the transfer function estimator H_1 such as

$$H_1(\theta, \varphi, \omega) = \frac{\mathbb{F}\{p_{sph}(\theta, \varphi, t)\}}{\mathbb{F}\{p_{ref}(t)\}} = D(\theta, \varphi)\left(\frac{\mathbf{A}(\omega)}{\mathbf{B}(\omega)}\frac{e^{-jkR}}{R}\right) \tag{4}$$

where $\mathbb{F}\{\cdot\}$ denotes time Fourier transform. Since the right hand side of Equation (4) is independent of the position of the moving sensor we can demonstrate that the transfer function estimator H_1 could be understood as a scaled version of the directivity term $D(\theta, \varphi)$, hence

$$H_1(\theta, \varphi, \omega) = D(\theta, \varphi)\gamma(\omega) \tag{5}$$

where $\gamma(\omega)$ is a scaling factor which depends on the position of the reference sensor, radius R and the source characteristics. From Equation (5) can be inferred that measuring transfer function between fixed and projected signals leads to a time independent expression for calculating the directivity of any sound source. This key factor allows using scanning techniques even though the source excitation is non stationary. Tracking the position of the moving sensor across time make possible to represent transfer function variations across space, which are directly related with the radiation patterns of the sound source. Nevertheless, errors could appear if not only the absolute level varies but also the spectral content changes along time. An algorithm for selecting the spatial positions depending on the dynamic range of the assessed frequency is presented in the next section in order to overcome this potential problems.

In summary, it has been shown analytically that the transfer functions between a fixed and a moving transducer gives a time independent ratio which carries information about the directivity of the sound source assessed.

3. Positional discrimination algorithm

So far it has been pointed out that taking transfer functions between a fixed and a moving transducer allows to characterize time independent relative variations across a sound field. Nonetheless, a detailed description of the measurement scenario is required to understand how to apply a positional discrimination algorithm. Figure 3 shows a schematic view of a generic experimental setup. This figure illustrates how a continuous time signal is driving the sound source while the scanning microphone is moving across the sound field. A finite grid of positions can be created relative to the location of the moving transducer. Then, the time intervals $(t_1, t_2, ..., t_n)$ can be linked with their corresponding measurement positions $(x_1, x_2, ..., x_n)$.

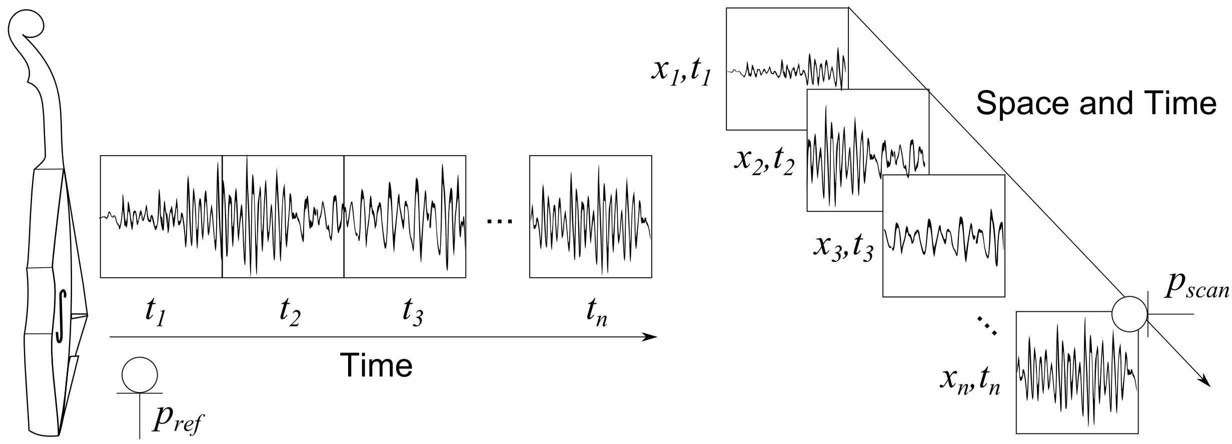

Figure 3. Schematic view of the measurement scenario

Next, the time sequences measured with the moving and static sensors are combined by computing the transfer function estimator H_1 (see Equation (4)). This allows to assess how the sound radiation changes across the space. Nonetheless, it is required to have sufficient spectral excitation at the source so as to evaluate this power spatial changes. Hence, the Power-Spectral Density (PSD) of the fixed transducer can be studied for different time intervals so as to neglect transfer functions with poor signal to noise ratio.

A maximum dynamic range has been establish depending on the maximum transfer function found for each frequency band. Consequently, any position with insufficient signal excitation for a given frequency has been disregarded. This leads to have an irregular spatial grid which size changes across the frequency domain. Figure 4 shows a representation of the procedure.

In summary, transfer functions have been mapped across space depending on the excitation signal. The spectrogram of the reference sensor is assessed for each position to see whether there is enough excitation on the source for each particular frequency.

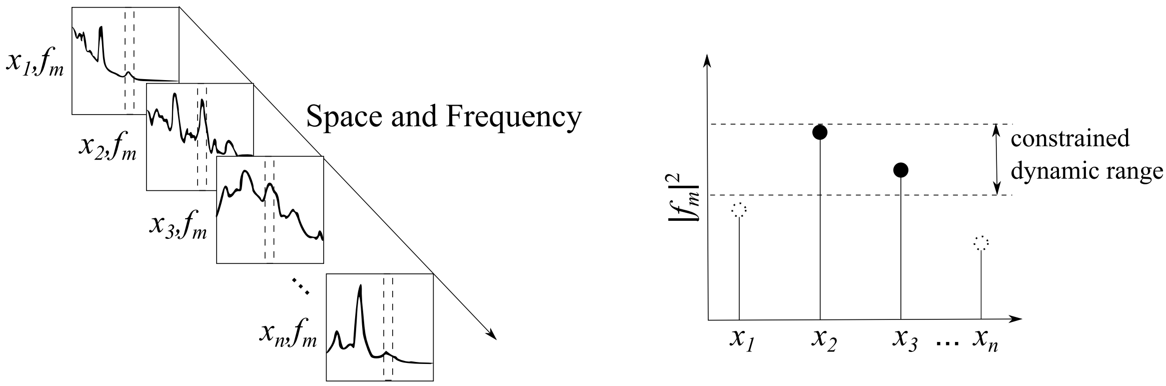

Figure 4. Diagram of the positional discrimination procedure

4. Instrumentation and experimental setup

Scanning measurements were carried out using a Microflown PU probe which contains a pressure microphone along with a particle velocity sensor. The reference pressure microphone used was a Read Instrument Microphone Fiddle Pro. This hyper-cardioid microphone was attached to the C bout of the violin by a stainless steel spring clip. In addition, a camera “Panasonic Lumix TZ7” was required for recording a video of the measurements.

Measurements were performed in the large anechoic chamber of the ISVR (Southampton, UK) for achieving free-field conditions. The sound radiation of a violin was measured carrying out sweeps one meter away from the musical instrument along surfaces of two meters by two meters.

Six different planes were scanned without modifying the measurement setup: the musician was turned around the radiation center for the front, left, right and back plane and then she had to perform lying on a table so as to capture the top and bottom radiation. The excitation signal measured was a traditional music piece. The time expended in each scanning was about 4 minutes.

In addition, Figure 5 presents a picture of the measurement setup in the ISVR anechoic chamber (left) along with a spectrogram sample extracted from one the measurements performed. As can be inferred from this figure, the statistical properties of short time segments of the excitation signal are not representative of the whole time series. Consequently, it can be conclude that the measurements were performed under non time stationary conditions.

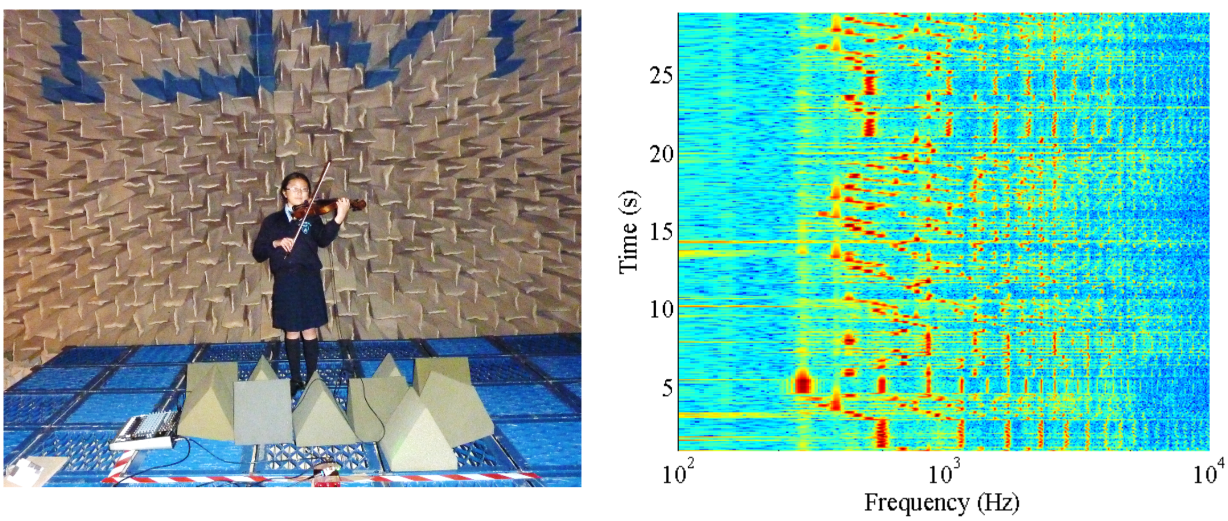

Figure 5. Experimental measurement setup (left) and spectrogram sample (right)

5. Presenting results

One common visualization methods for presenting 3D directivity patterns is using colormaps overlay on a mesh of the geometry used. It gives a direct feedback about radiation maxima but it is not enough clear to follow the pattern variations across the space. Consequently, it is common practice using polar coordinates scaling the radius of the point aimed to represent according to the value represented by the color. Figure 6 illustrates the two representation methods explained. The data used in both cases corresponds to the scaled directivity described in Section 2.

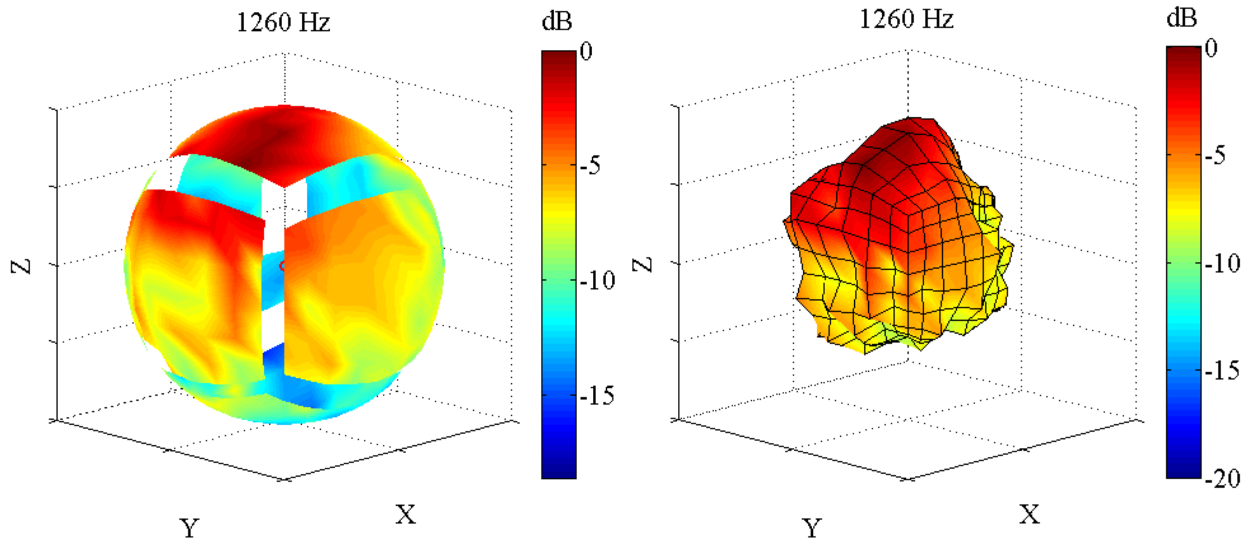

Figure 6. Example of a monopole radiattion measurements transformed to spherical directivity pattern

6. Directivity patterns

Figure 7 presents several examples of the measured directivity patterns at four different third octave frequency bands: 315 Hz, 500 Hz, 1000 Hz and 1588 Hz. The X plane represent the frontal measurement plane. Hence, only the frontal, top and right directivity can be seen in the given figure. According to the measurement results the violin has an omnidirectional behavior at the lower frequency bands. In contrast, as frequency increases the directivity patterns shapes change dramatically, moving the radiation lobe from the front of the musician to the side, depending on the frequency band assessed.

7. Advantages and disadvantages of the measurement procedure

Current methods for measuring non-time stationary sound field rely on using large sensor arrays. The most common solution uses one sensor for each measurement position. Alternatively, methods based on NAH can reconstruct the entire sound field by placing multiple reference transducers and then scanning an area with a large array [5]. The novel technique proposed in this paper only requires two sensors: one static while the other is manually moved. Time, cost, simplicity, flexibility and accuracy are the main issues evaluated in this section which determine the advantages and disadvantages of choosing a measurement technique.

Time required for setting up the instrumentation and performing the measurement is always a big issue. Manual sweeps of a single probe are a fast procedure for directly obtaining information about a sound field. Each measurement plane was evaluated undertaken sweeps for less than 4 minutes, which can be seen as a reasonable amount of time for characterizing the sound field produced by a non-stationary sound source such as violin.

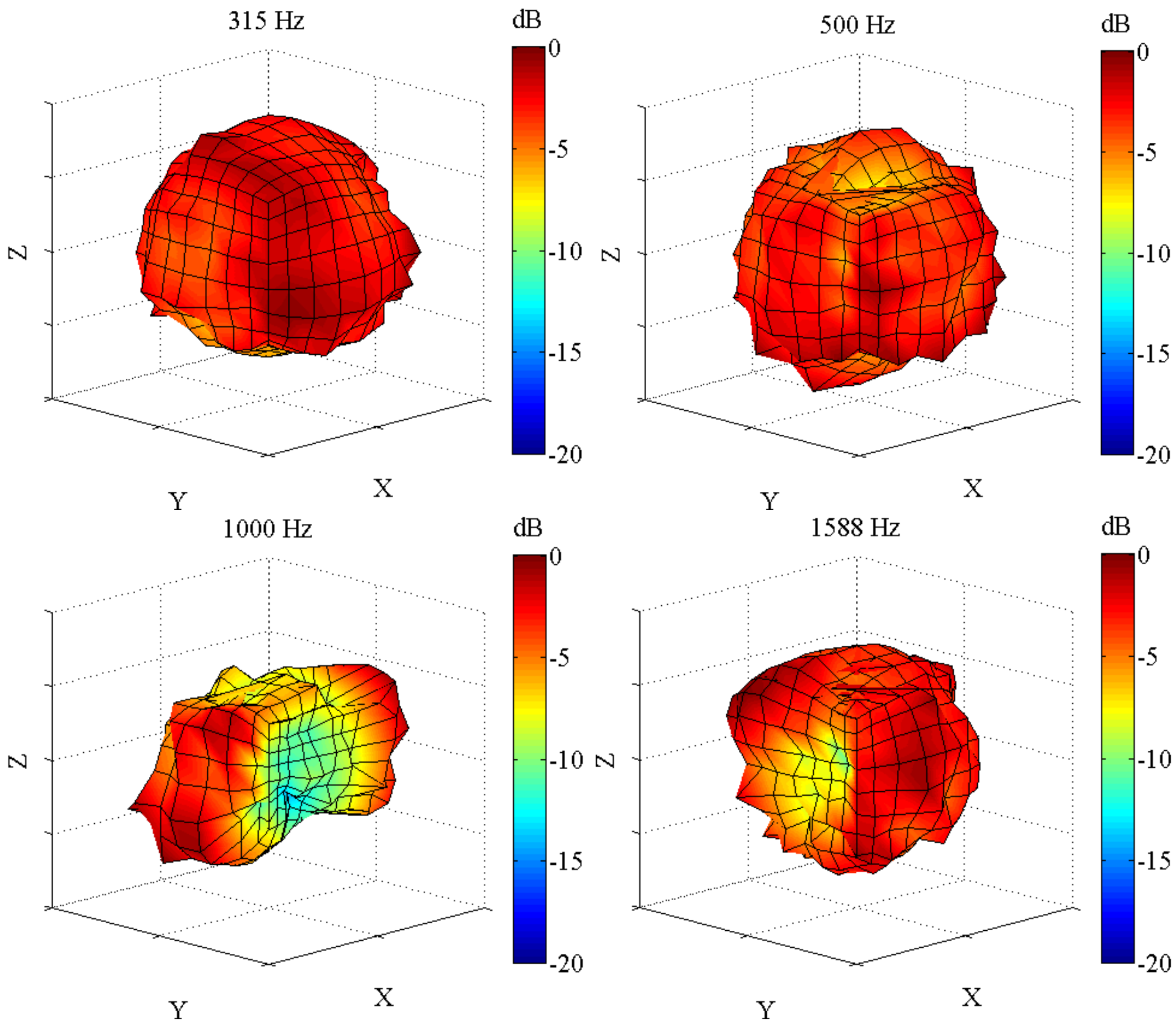

Figure 7. Directivity patterns of a violin at third octave bands of 500 Hz (top left), 1000 Hz (top right), 1260 Hz (bottom left) and 1588 Hz (bottom right) [dB]

One of the main problems of most conventional array measurement systems is the cost of the equipment. Not only the number of transducer required for performing the measurements but also the multichannel acquisition system rise the price strongly. The proposed two-sensor solution has far less requirements than most of the current large multichannel applications.

The measurement protocol and the post processing stage should be fairly intuitive. The use of a camera makes sure that all the measurement process is filmed . This has been proved to be helpful with trouble shooting. Color maps overlaid on pictures give a direct feedback that is easy to understand.

The flexibility of the proposed method is one of its stronger advantages against array-based solutions. The novel technique allows to setup all instrumentation and resize the measurement plane just by moving the camera. Furthermore, the spatial resolution of the measurement is selected after performing the measurement allowing to assess several spatial distribution a posteriori. The criteria for selecting the blocks size of the grid depends on the frequency investigated and the duration of the measurement (as the sweep get longer the grid blocks can be smaller).

The main outcome of a measurement technique is to ensure accurate and reliable results. The smooth radiation maps presented in Figure 7 support the great potential of combining the Scan & Paint measurement techniques with the positional discrimination algorithm. The more measurements per plane, the more accurate the results will be since the small errors due to human factors can be minimized when averaging all the sessions. Besides, the fact that there are not physical fixed positions leads to minimize the interference effect of placing so many objects in the sound field.

8. Conclusions

A theoretical base for measuring time independent relative changes in a sound field has been derived. Moreover, an algorithm for selecting the spatial areas with same signal excitation has been proposed. The combination of the described principles with the scanning measurement technique Scan & Paint lead to a novel method for characterizing directivity patterns of non-stationary sound sources using a scan based two-channel system. Results presented support the successful implementation of the method. The measurement technique presented reduces the number of transducers, measurement time and cost, while maximizes the flexibility of current methods.

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