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NEAR-FIELD STEERED COVARIANCE MATRIX APPROACH FOR RESOLUTION AND DYNAMIC IMPROVEMENT IN ACOUSTIC CAMERA APPLICATION

Miljko Erić[1] and Milan Mišković[2]
[1]School of Electrical Engineering, University of Belgrade
Bul. Kralja Aleksandra 73, 11000, Belgrade, Serbia
[2]VTI, Ratka Resanovića 1a, Belgrade, Serbia

ABSTRACT

Concept of steered covariance matrix (STCM) was proposed in far-field wideband beamforming but to the authors' best knowledge this approach is not widely exploited in near-field beamforming applied in acoustic mapping. In this paper we will give theoretical background of steered covariance matrix approach applied in near-field signal scenario combined with classical Bartlett and high-resolution spectral method MUSIC in order to see if this approach can improve resolution properties as well as dynamic range in near-field array processing applied in acoustic mapping. Results provided by simulations and measurements will be compared and presented in this paper.

1 INTRODUCTION

Near-field beamforming is one of known techniques applied in acoustic camera. In fact it is based on array processing of spherical wave of the sound sources which are at small distances (some wavelengths) form the microphone array. Mathematical model of the signal on microphone array in near-filed scenario depends on the locations of the sound sources, so the near-field beamforming can directly identify locations of acoustic sources and provide acoustic picture in spatial sector of interests (direction of arrivals - DOAs in far field signal scenario, became locations in near-field beamforming) [1],[2],[3],[4].

In classical near-field beam-forming applied in acoustic camera there is a problem of spatial resolution and dynamic range limitation in acoustic mapping [5]. Spatial resolution is defined as the shortest distance at which two point sources can be spatially separated. From the theory of array processing it is well known that if the size of antenna/microphone array (aperture) is as big as possible, the resolution properties are better. Array aperture can be increased without increasing the number of microphones with non-uniform placement of microphone in the array, but in that case, side-lobes in beamforming appear. It is also known that spatial resolution is frequency dependent, so the lower frequency means lower spatial resolution.

Dynamic range (called also as *contrast* [5] or *plot dynamic* [6]) is the difference of signal levels between the two time and frequency overlapped sound sources that can be clearly identified and detected. If the level difference between two real sources is greater than the dynamic range, louder sources will mask quieter sources. Side-lobe levels in beamforming are a very important fact in acoustic camera application since it directly limits dynamic range in acoustic mapping. It is directly influenced by the geometry and aperture of the microphone array. Known leading producers of acoustic cameras [7],[8],[9],[10],[11] claim that dynamic range of its acoustic cameras is from 25 to 40 dB but it is not quite clear how it is provided, since they give reference to the their patented solutions.

So, improvement of those characteristics is a really challenging research problem. Those characteristics can be improved with increasing the number of microphones and increasing the array aperture of microphone array but the real technical challenge is how to improve it by array processing using the same number of microphones and array aperture.

Concept of steered covariance matrix (STCM) was proposed in far-field wideband beamforming [12],[13] but to the authors' best knowledge this approach is not widely exploited in near-field beamforming applied in acoustic mapping. In this paper we will give theoretical background of steered covariance matrix approach applied in near-field signal scenario combined with classical Bartlett and high-resolution spectral method MUSIC and we will research if this approach can improve resolution properties as well as dynamic range in near-field array processing applied in acoustic mapping.

The rest of the paper is organized as follows. Mathematical model of the acoustic signal on microphone array in near-field signal scenario is formulated in the section 2. Theoretical foundation of acoustic mapping in near-field signal scenario based on classical delay and sum beamformer is formulated in section 3. In section 4. acoustic mapping in near-field signal scenario based on the steered covariance matrix is theoretically founded. Results of measurements and simulations are presented in the section 5. Conclusions are given in the section 6.

Motivation for this research is related to the Serbian national project of Ministry of Science and Technological Development TR 32026 called "*Integration and Harmonization of Sound Insulation in Buildings in the Context of Sustainable Housing*". In this project we tend to apply technical solutions of acoustic camera for development and verification of technical solutions for sound insulation in buildings. To the author's best knowledge acoustic camera has not so far been used for this purpose so it seems that it is our research challenge.

2 MATHEMATICAL MODEL OF ACOUSTIC SIGNAL ON MICROPHONE ARRAY IN NEAR FIELD SIGNAL SCENARIO

We consider near field acoustic multiple incident signal scenario, Fig.1., with K acoustic sources at the unknown locations denoted by the set of vectors $\mathbf{r}_k \in$ ˇ $^3; k = 1, K$. Suppose also that acoustic signals are continual and they are fully or partially overlapped in time and frequency. Summary acoustic signal is received by the microphone array with L microphones on the known locations denoted by the set of vectors $\mathbf{p}_l \in$ ˇ $^3; l = 1, L$.

Time samples $x_l(n\Delta t)$ of the signal received on the l-th microphone can be modeled in the next form [1]:

$$x_l(n\Delta t) = \sum_{k=1}^{K} b_l(\mathbf{r}_k) s_k^0(n\Delta t - \tau_l(\mathbf{r}_k)) + \eta_l(n\Delta t); n = 1, N \qquad (1)$$

where $b_l(\mathbf{r}_k)$ and $\tau_l(\mathbf{r}_k) = |\mathbf{p}_l - \mathbf{r}_k| / v$ denote attenuation and propagation time delay of the k-th signal, $s_k^{(0)}(.)$ denotes signal of the k-th acoustic source on its location, N denotes total number of signal samples and v denotes velocity of acoustic wave.

Received signal on microphone array could be formulated in spectral domain in such matrix form:

$$\mathbf{x}(\omega_n) = \mathbf{A}(\omega_n)\mathbf{s}^0(\omega_n) + \boldsymbol{\eta}(\omega_n) \qquad (2)$$

where $\mathbf{x}(\omega_k) \in$ Ł ^{Lx1} and $\boldsymbol{\eta}(\omega_k) \in$ Ł ^{Lx1} denote vectors with spectral samples of the signal and noise spectrum, respectively, $\mathbf{A}(\omega_n) \in C^{LxK}$ is a matrix whose columns are steering vectors $\mathbf{a}(\mathbf{r}_k, \omega_n) \in$ Ł ^{Lx1} whose elements have a general form:

$$a_p(\mathbf{r}_k, \omega_n) = b_l(\mathbf{r}_k) e^{-j\omega_n \tau_l(\mathbf{r}_k)} = b_l(\mathbf{r}_k) e^{-j\omega_n \|\mathbf{p}_l - \mathbf{r}_k\| / v} \qquad (3)$$

Vector $\mathbf{s}^{(0)}(\omega_n) = [S_1^{(0)}(\omega_n) \quad S_2^{(0)}(\omega_n) \quad \ldots \quad S_K^{(0)}(\omega_n)]^T$ is a vector with spectral samples of all K acoustic signals. It can be noticed that, regardless of far-field signal, near-field signal model given by (3) depends on the signal source locations and signal levels on those locations. It can be noticed also that steering vectors $\mathbf{a}(\mathbf{r}_k, \omega_n)$ are frequency dependent so this model describes wideband nature of acoustic signals in array processing context.

Suppose also that set of vectors $\mathbf{x}(\omega_n); n = 1, N$ are available, provided by acquisition of vectors with N time samples on each microphone channel, and application of Hilbert Transform (HT) and Discrete Fourier Transform (DFT) on those vectors.

The main task is in acoustic camera application is form acoustic map of the near-field signal scenario to estimate locations of powers of the acoustic sources.

The main tasks of acoustic camera applications are identification and quantification of noise/sound sources (estimation of their locations and absolute powers) by array processing of available vectors $\mathbf{x}(\omega_n); n = 1, N$ and to perform a picture (visualization) of acoustic environment by overlaying that acoustic picture to the video picture.

3 ACOUSTIC MAPPING IN NEAR FIELD SIGNAL SCENARIO BASED ON CLASICAL WIDEBAND DELAY AND SUM BEAMFORMER

Idea of near-field acoustical mapping based on classical wideband delay and sum beamformer is to steer and focus beamformer in the point of space denoted by the vector $\mathbf{r}$ and to estimate average power at the output of the steered beamformer. In order to provide acoustical mapping, this procedure is repeated for the grid points in spatial sector of interests.

Classical wideband delay and sum beamformer can be formulated in time domain as:

$$y(\mathbf{r}, n\Delta t) = \sum_{l=1}^{L} b_l(\mathbf{r})^{-1} x_l(n\Delta t + \tau_l(\mathbf{r})) \tag{4}$$

where $y(\mathbf{r}, n\Delta t)$ denotes time samples of the output of beamformer.

From the array processing point of view acoustic signals have to be modeled as wideband signals, so acoustic beamforming is broadband beamforming and it can be basically implemented in spectral domain.

Mathematical model of delay and sum beamformer can be formulated in spectral domain as:

$$Y(\mathbf{r}, \omega_n) = \sum_{l=1}^{L} b_l(\mathbf{r})^{-1} e^{j\omega_n \tau_l(\mathbf{r})} X_l(\omega_n) \tag{5}$$

or in the next matrix form as:

$$Y(\mathbf{r}, \omega_n) = \mathbf{w}^H(\mathbf{r}, \omega_n)\mathbf{x}(\omega_n) \tag{6}$$

where vector of beamformer coefficients $\mathbf{w}(\mathbf{r}, \omega_n) \in C^{Lx1}$ is defined as:

$$\mathbf{w}(\mathbf{r}, \omega_n) = [b_1(\mathbf{r}, \omega_n)^{-1} e^{-j\omega_n \tau_1(\mathbf{r})} \quad b_2(\mathbf{r}, \omega_n)^{-1} e^{-j\omega_n \tau_2(\mathbf{r})} \quad \ldots \quad b_L(\mathbf{r}, \omega_n)^{-1} e^{-j\omega_n \tau_L(\mathbf{r})}]^T \tag{7}$$

It can be seen that this vector depends on the frequency ω_n and location of focused point in the space denoted by the vector $\mathbf{r}$. Vector $\mathbf{x}(\omega_n) = [X_1(\omega_n) \quad X_2(\omega_n) \quad \ldots \quad X_L(\omega_n)]^T$ is a vector with spectral samples of the signals on microphones.

Average output power $P_y(\mathbf{r}, \omega_n)$ of the delay and sum beamformer on the frequency ω_n can be expressed as:

$$P_y(\mathbf{r},\omega_n) = E\{y(\mathbf{r},\omega_n)y(\mathbf{r},\omega_n)^H\} = \mathbf{w}(\mathbf{r},\omega_n)^H E\{\mathbf{X}(\omega_n)\mathbf{X}(\omega_n)^H\}\mathbf{w}(\mathbf{r},\omega_n) = \\ = \mathbf{w}\left(\mathbf{r},\omega_n\right)^H \mathbf{R}(\omega_n)\mathbf{w}\left(\mathbf{r},\omega_n\right) \tag{8}$$

where $\mathbf{R}(\omega_n)$ denotes Cross-Spectral Covariance Matrix (CSCM), [13] end $E\{.\}$ denotes operator of mathematical expectation.

Total average output power of the delay and sum beamformer can be expressed as:

$$P_y^{BART}(\mathbf{r}) = \sum_{n=1}^{N} P_y(\mathbf{r},\omega_n) = \sum_{n=1}^{N} \mathbf{w}^H\left(\mathbf{r},\omega_n\right)\mathbf{R}(\omega_n)\mathbf{w}\left(\mathbf{r},\omega_n\right) \tag{9}$$

So, total output power $P_y^{BART}(\mathbf{r})$ is provided by incoherent combining of the delay and sum output powers on the set of frequencies $\omega_n; n = 1:N$. It is in fact near-field wideband variant of the so-called Bartlet estimator defined in DOA estimation literature [14].

The main limitation of this approach is that statistically stable estimation of frequency-dependent cross spectral covariance matrix $\mathbf{R}(\omega_n)$ requires a relatively long observation interval.

4 ACOUSTIC MAPPING IN NEAR FIELD SIGNAL SCENARIO BASED ON STEERED COVARIANCE MATRIX APPROACH

Steered covariance matrix approach is proposed in literature in the context of DOA estimation of wideband signal sources but as the best knowledge of the authors it is not widely used in near-field source localization. The key idea of steered covariance matrix approach is to formulate rank-one model of the wideband signal [12],[13]. This approach provides estimation of covariance matrix with greater statistical stability when observation interval is limited.

Such rank-one model requires a preprocessing of multidimensional signal on sensor (microphone) array.

In order to formulate rank-one model of the wideband signal, Wang and Caveh in [15] proposed a Coherent Signal Subspace (CSS) processing based on the estimation of spectral focused covariance matrix. By this preprocessing this method basically provides coherent combining of the contribution of all spectral sub-bands in wideband DOA estimation.

Krolik and Swingler proposed in [12] steered covariance matrix (STCM) approach in DOA estimation of wideband signals, based on the idea '*that time domain sensors outputs are simply steered in each direction of interest prior to forming wideband covariance matrix*' [13].

This steering preprocessing transformation in near-field signal scenario is performed in frequency domain in such a way:

$$\mathbf{x}^{st}(n\Delta T,\mathbf{r}) = \sum_{n=1}^{N} \mathbf{T}(\omega_n,\mathbf{r})\mathbf{x}(\omega_n)e^{j\omega_k n\Delta t} \tag{10}$$

where vector $\mathbf{x}^{st}(n\Delta T,\mathbf{r}) \in C^{Lx1}$ represents vector of the time samples on the microphone array after the steering preprocessing transformation. Matrix $\mathbf{T}(\omega_n,\mathbf{r}) \in C^{LxL}$, which is so called steering matrix, is diagonal matrix with elements $b_l(\mathbf{r})^{-1}e^{j\omega_n\|\mathbf{p}_l-\mathbf{r}\|/v}$ on the main diagonal.

Furthermore, steered covariance matrix $\mathbf{R}^{st}(\mathbf{r}) \in C^{LxL}$ can be estimated as:

$$\mathbf{R}^{st}(\mathbf{r}) = E\{\mathbf{x}^{st}(n\Delta T,\mathbf{r})\mathbf{x}^{st}(n\Delta T,\mathbf{r})^H\} = \sum_{n=1}^{N} \mathbf{T}(\omega_n,\mathbf{r})\mathbf{R}(\omega_n)\mathbf{T}(\omega_n,\mathbf{r})^H \tag{11}$$

It can be seen that steered covariance matrix $\mathbf{R}^{st}(\mathbf{r})$, regardless of the cross-spectral covariance matrix $\mathbf{R}(\omega_n)$, depends on the location but not on frequency. This matrix can be estimated directly from the vectors $\mathbf{x}^{st}(n\Delta T,\mathbf{r}) \in C^{Lx1}$ but also from the set of previously estimated cross-spectral covariance matrices $\mathbf{R}(\omega_n); n = 1, N$.

Variant of near-field steered-Bartlett estimator based on use of steered covariance matrix, can be defined as:

$$P_y^{st\_BART}(\mathbf{r}) = \mathbf{v}^H\mathbf{R}^{st}(\mathbf{r})\mathbf{v} \tag{12}$$

Furthermore, near-field steered-MUSIC estimator can be defined as:

$$P_y^{st\_MUSIC}(\mathbf{r}) = \frac{1}{\mathbf{v}^H\mathbf{E}^{st}(\mathbf{r})\mathbf{E}^{st}(\mathbf{r})^H\mathbf{v}} \tag{13}$$

Vector $\mathbf{e}(\mathbf{r}) \in \check{\ }^{Lx1}$ in eq. 12. and 13. has a form:

$$\mathbf{v} = [1 \quad 1 \quad \ldots \quad 1]^T \in \check{\ }^{Lx1} \tag{14}$$

Matrix $\mathbf{E}^{st}(\mathbf{r})$ is the noise subspace matrix of the steered covariance matrix $\mathbf{R}^{st}(\mathbf{r})$.

Near-field steered-Bartlett estimator formulated by (12) can be used for the estimation of total output power of near-field beamformer focused in the point in space denoted by the vector $\mathbf{r}$.

Near-field steered-MUSIC estimator formulated by (13) cannot be used for the estimation of total output power but it can be used for high-resolution location estimation. We expect that steered-MUSIC estimator does not only have better resolution properties than conventional near-field steered-Bartlett estimator but that it also provides better dynamic range in acoustic mapping, So we will try to prove that by the comparative results of measurements and simulations presented in the next section of the paper.

5 RESULTS OF SIMULATIONS AND PRACTICAL MEASUREMENTS

Circular low-cost microphone array with diameter 0.75 m and 8 microphones in the array was designed and realized in our laboratory of acoustics, primaryly for education but also for research purposes. The experiment was perfomed using measurement setup presented in the figure 1. which contains mentioned microphone array, multicahnnel microphone preamplifiers and signal acquisition and lap-top for the control of merasurement setup, collection of signal samples and off-line implementation of beamformers [16].

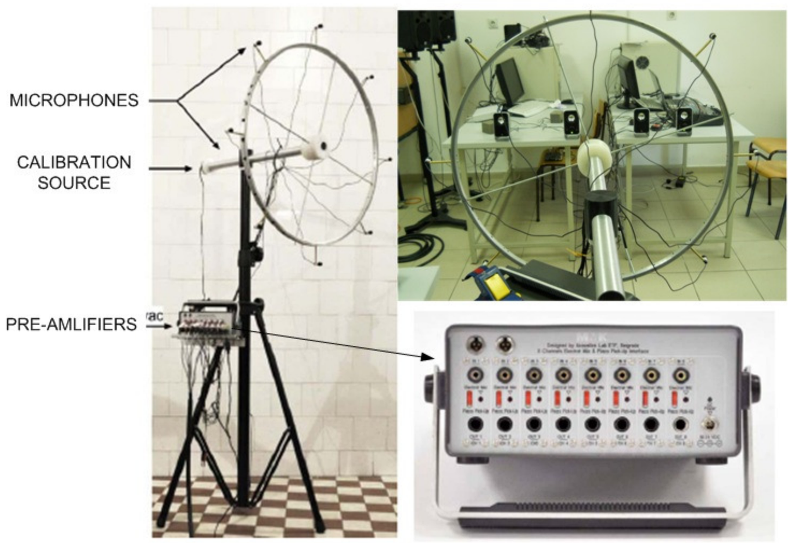

Fig.1. Measurement setup using circular microphone array

Measurements were realized in indoor environment, and no calibration procedure of the receiving channels is performed. Simulation was also realized with the same parameters of measurement signal scenario, and provided results are compared and presented

In the first measurement we generated two time and frequency overlapped test acoustic signals from two loudspeakers placed at the shortest possible distances (9.18 cm, limited by physical dimensions of loudspeakers). Test acoustic signals have equal powers and Signal/to

Noise Ratios (9 dB) and the same spectral and statistical properties (uncorrelated white Gauss noise sequences generated by MATLAB with spectral bandwidths 200Hz-10 kHz).

Experimental results and results of simulations for this signal scenario are presented in the figures 2. and 3. respectively. Steered MUSIC and Steered Bartlett functions are calculated on grid points in *x0z* plane for the value of *y* equal to the known (measured) distance of loudspeakers from the plane of microphones and for time interval equal to 6400 signal samples (sampling frequency was 24 ks/s for 200 Hz-10 kHz bandwidth, 12 ks/s for 200 Hz-4 kHz bandwidth and 8 ks/s for 200 Hz-3 kHz bandwidth).

It can be seen that both methods (Steered Bartlett and Steered MUSIC) resolve two acoustic sources in the considered signal scenario. Also it can be noticed that there is a good matching between experimental results and results of simulations regarding source resolvability and location accuracy estimation but there are differences between pick and side-lobe levels in measurement and simulation, as well as in steered MUSIC and steered Bartlett estimation.

We repeat this experiment with the same signal scenario, but for the spectral bandwidths 200 Hz-4 kHz and 200 Hz-3 kHz. It can be seen in the figures 4. and 5, that signal sources can be still resolved with both methods for the spectral bandwidths 200 Hz-4 kHz, but for the spectral bandwidths 200 Hz-3 kHz they cannot be resolved. This result is expectable, since microphone aperture basically determines resolution properties (smaller frequency ->smaller aperture->smaller resolution). We expected that steered MUSIC will give much better resolution properties since it is a high resolution method, but in this particular scenario the results are as presented.

In the second signal scenario we considered dynamic range since it is a very important aspect of acoustic camera application. The signal scenario was similar (same type of signals are used with spectral bandwidths 200 Hz-10 kHz, but distance between loudspeakers was 20 cm and the signal level of the second (right) loudspeaker was 8 dB lower than the power of the first (left) loudspeaker. As it can be seen from figures 8 and 9 steered MUSIC method provides clear detestability of the pick of the second user, but that pick in the case of steered Bartlett method is on the level of side-lobes. So steered MUSIC provides better dynamic range than steered Bartlett but this improvement in the considered signal scenario and used microphone geometry is not so dramatic.

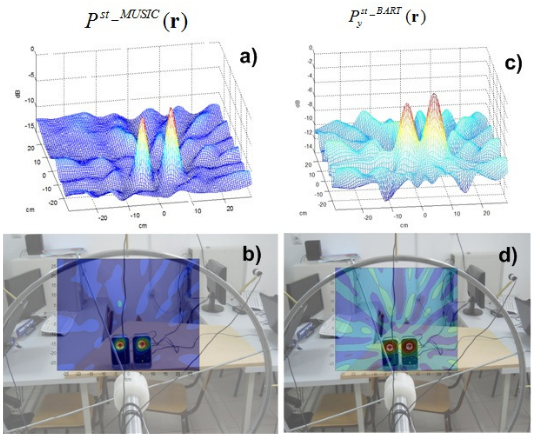

Fig.2. Experimental results, signal bw (200Hz-10kHz), SNR1/SNR2=0dB: a) mesh plot of $P^{st\_MUSIC}(\mathbf{r})$; a) contour plot of $P^{st\_MUSIC}(\mathbf{r})$; c) mesh plot of $P^{st\_BART}(\mathbf{r})$; d)contour plot of $P^{st\_BART}(\mathbf{r})$

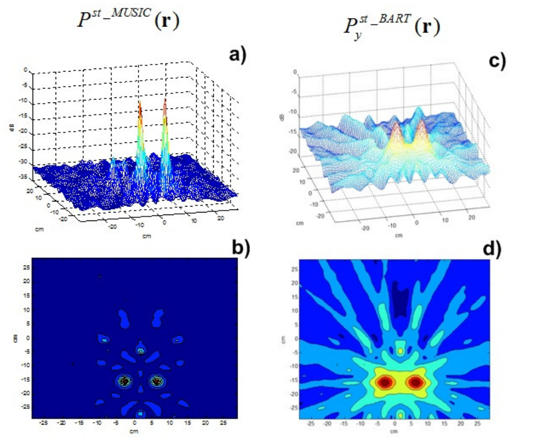

Fig.3. Results of simulations, signal bw (200Hz-10kHz), SNR1/SNR2=0dB: a) mesh plot of $P^{st\_MUSIC}(\mathbf{r})$; a) contour plot of $P^{st\_MUSIC}(\mathbf{r})$; c) mesh plot of $P^{st\_BART}(\mathbf{r})$; d) contour plot of $P^{st\_BART}(\mathbf{r})$

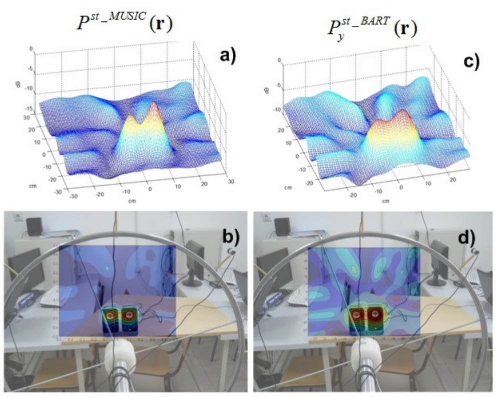

Fig.4. Experimental results, signal bw (200Hz - 4kHz),SNR1/SNR2=0dB: a) mesh plot of $P^{st\_MUSIC}(\mathbf{r})$; a) contour plot of $P^{st\_MUSIC}(\mathbf{r})$; c) mesh plot of $P^{st\_BART}(\mathbf{r})$; d) contour plot of $P^{st\_BART}(\mathbf{r})$

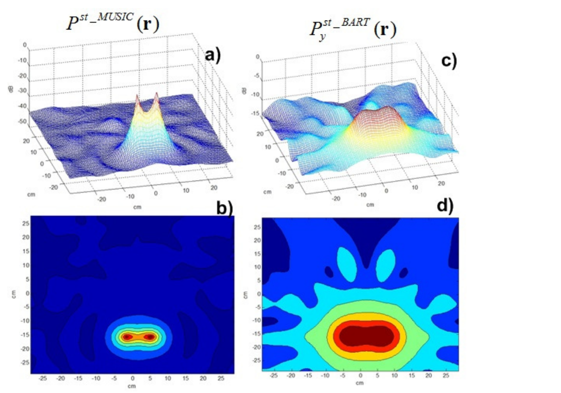

Fig.5. Results of simulations, signal bw (200Hz - 4kHz), SNR1/SNR2=0dB: a) mesh plot of $P^{st\_MUSIC}(\mathbf{r})$; a) contour plot of $P^{st\_MUSIC}(\mathbf{r})$; c) mesh plot of $P^{st\_BART}(\mathbf{r})$; d) contour plot of $P^{st\_BART}(\mathbf{r})$

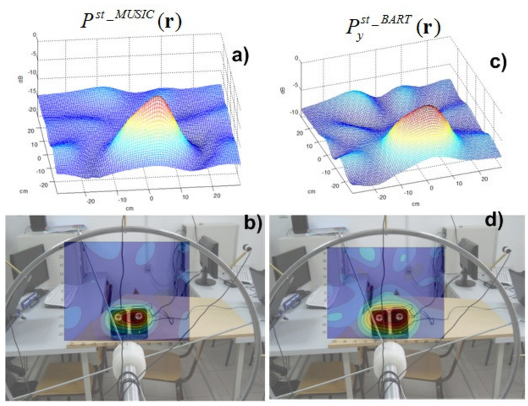

Fig.6. Experimental results, signal bw (200Hz-3kHz), SNR1/SNR2=0dB: a) mesh plot of $P^{st\_MUSIC}(\mathbf{r})$ *; a) contour plot of* $P^{st\_MUSIC}(\mathbf{r})$ *; c) mesh plot of* $P^{st\_BART}(\mathbf{r})$ *; d)contour plot of* $P^{st\_BART}(\mathbf{r})$

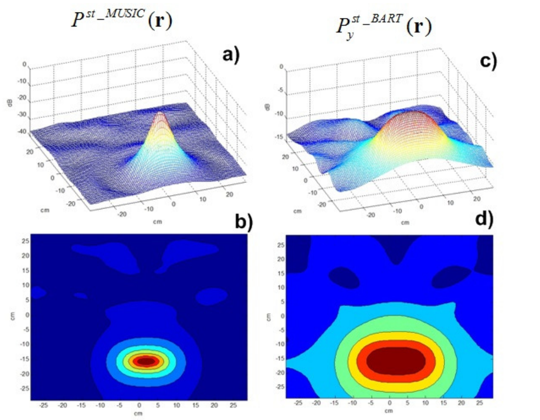

Fig.7. Results of simulations, signal bw (200Hz-3kHz), SNR1/SNR2=0dB: a) mesh plot of $P^{st\_MUSIC}(\mathbf{r})$ *; a) contour plot of* $P^{st\_MUSIC}(\mathbf{r})$ *; c) mesh plot of* $P^{st\_BART}(\mathbf{r})$ *; d) contour plot of* $P^{st\_BART}(\mathbf{r})$

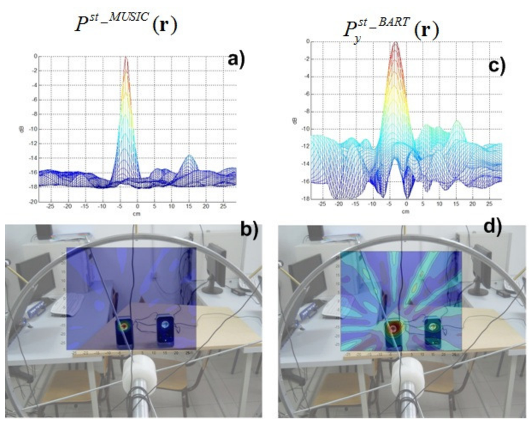

Fig.8. Experimental results, signal bw (200Hz-10 kHz), SNR1/SNR2=8dB; a) mesh plot of $P^{st\_MUSIC}(\mathbf{r})$ *; a) contour plot of* $P^{st\_MUSIC}(\mathbf{r})$ *; c) mesh plot of* $P^{st\_BART}(\mathbf{r})$ *; d)contour plot of* $P^{st\_BART}(\mathbf{r})$

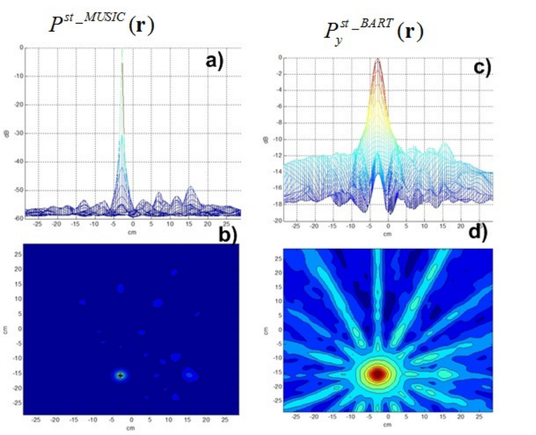

Fig.9. Experimental results, signal bw (200Hz-3 kHz), SNR1/SNR2=8dB: a) mesh plot of $P^{st\_MUSIC}(\mathbf{r})$ *; a) contour plot of* $P^{st\_MUSIC}(\mathbf{r})$ *; c) mesh plot of* $P^{st\_BART}(\mathbf{r})$ *; d) contour plot of* $P^{st\_BART}(\mathbf{r})$

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6 CONCLUSIONS

Theoretical foundations of near field acoustical mapping based on the classical near-field delay and sum beamformer and steered covariance matrix approach are presented in this paper. Steered-Bartlett and steered-MUSIC method for near-field acoustic mapping are formulated and their performances regarding resolution and dynamic range properties are compared by the results of real measurements and results of simulations. Presented results provided for the considered near-field signal scenario prove that steered MUSIC method has generally better resolution and dynamic range properties then steered Bartlett method, but it is not so dramatic in the considered near field signal scenario. Presented results also prove that there is good matching between measurement and simulation results regarding resolution and localization properties (we expected that indoor measurement scenario will significantly degrade the beamformer performances). Presented experimental also prove that realized measurement setup with low cost microphone array is a relatively good basis for our future researches.

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