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Journal of Sound and Vibration

journal homepage: www.elsevier.com/locate/jsvi

Constrained Spectral Conditioning for spatial sound level estimation

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ARTICLE INFO

Article history:

Received 4 November 2015

Received in revised form

16 June 2016

Accepted 7 July 2016

Handling Editor: K. Shin

Available online 28 July 2016

Keywords:

Phased array

Spatial filtering

Preprocessor

Beamforming

Linear constraint

ABSTRACT

Microphone arrays are utilized in aeroacoustic testing to spatially map the sound emitted from an article under study. Whereas a single microphone allows only the total sound level to be estimated at the measurement location, an array permits differentiation between the contributions of distinct components. The accuracy of these spatial sound estimates produced by post-processing the array outputs is continuously being improved. One way of increasing the estimation accuracy is to filter the array outputs before they become inputs to a post-processor. This work presents a constrained method of linear filtering for microphone arrays which minimizes the total signal present on the array channels while preserving the signal from a targeted spatial location. Thus, each single-channel, filtered output for a given targeted location estimates only the signal from that location, even when multiple and/or distributed sources have been measured simultaneously. The method is based on Conditioned Spectral Analysis and modifies the Wiener–Hopf equation in a manner similar to the Generalized Sidelobe Canceller. This modified form of Conditioned Spectral Analysis is embedded within an iterative loop and termed Constrained Spectral Conditioning. Linear constraints are derived which prevent the cancellation of targeted signal due to random statistical error as well as location error in the sensor and/or source positions. The increased spatial mapping accuracy of Constrained Spectral Conditioning is shown for a simulated dataset of point sources which vary in strength. An experimental point source is used to validate the efficacy of the constraints which yield preservation of the targeted signal at the expense of reduced filtering ability. The beamforming results of a cold, supersonic jet demonstrate the qualitative and quantitative improvement obtained when using this technique to map a spatially-distributed, complex, and possibly coherent sound source.

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1. Introduction

Arrays of microphones are used as spatial and temporal filters for the pressure signals which they measure. Processing of the array outputs yields spatial information of the sound source being studied which is not available when the microphones

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Nomenclature		Greek	
Latin			
$\hat{\phi}_{mm'}$	Measured cross-spectral phase between m and m'	$\hat{\gamma}_{mm'}^2$	Measured ordinary coherence between m and m'
ϕ_{sm}	Phase between s and m	Δx	Euclidean distance between scanning grid points in the <i>x</i> -direction
$\phi_{s,mm'}$	Relative phase difference between m and m' due to source at s	$\Delta x'$	Euclidean distance between scanning grid intra-grid points in the <i>x</i> -direction
<i>c</i>	Speed of sound in propagation medium	Δx_s	Euclidean source dimension in the <i>x</i> -direction
ch_m	Pressure time record of microphone m	ε	Normalized error
CH_m	Fourier transform of ch_m	σ	Standard deviation
e_s	Steering vector matrix for all M for location s	<i>Abbreviations</i>	
e_{sm}	Steering vector from s to m	CSC	Constrained Spectral Conditioning
<i>f</i>	Frequency	CSM	Cross-Spectral Matrix
F_s	Sampling frequency	FDBF	Frequency-Domain Beamforming
$\hat{G}_{mm'}$	Measured cross-spectrum between m and m'	RE	Random Error
$G_{s,mm'}$	Autospectrum induced on m from source s	SD	Standard Deviation
<i>K</i>	Number of data blocks that ch_m is divided into	SVE	Steering Vector Error
m	Euclidean position vector of a microphone	<i>Subscripts</i>	
m' _{op}	Optimum reference microphone	op	Optimum
<i>M</i>	Total number of microphones in array	<i>Superscripts</i>	
M	Total set of m	*	Complex conjugation
r_{sm}	Euclidean distance between spatial location s and microphone m	\widehat{}	Measured value (over variable)
s	Euclidean position vector of scanning location (targeted point in space)	†	Conjugate transposition
s ₀	Euclidean position vector of a source	˘	Different value of the same variable which does not carry this superscript
s'	Euclidean position vector of an intra-volume grid point for s	it	Iteration
<i>S</i>	Total number of sources in a measured field		
S	Total set of s		
<i>T</i>	Length (in samples) of data block obtained from ch_m		
W_b	Blocking/filtering weight term		

are used independently. However, the accuracy of the spatial mapping obtained is still being improved upon in the aero-acoustic community, and many separate efforts have been made to develop algorithms which post-process the array data to more accurately map the measured sound field [1–12]. One of the simplest and most commonly used processing formulations for synthesizing the sensor output data in order to obtain spatial information about the measured field is data-independent beamforming [13], which has been widely used in aeroacoustic testing since 1975 [14–20]. Optimal forms of beamforming (e.g., the Generalized Sidelobe Canceller (GSC, [21,22]) and Minimum Variance Distortionless Response [23]) which depend on the measured data (data-dependent) improve upon the data-independent formulation when properly constrained [24,25]. However, due to their nonlinear formulation and output uncertainty, only a few examples of their usage exist in the aeroacoustic literature [6,26–28].

Most optimal, data-dependent beamformers produce synthesized outputs, i.e., sound level estimation at targeted locations in the field is only available once the individual channel data are (manipulated and) summed. An exception can be found in [29] which provides a reformulation of the GSC that operates on the single-channel data in the time domain before array synthesis. The work presented in this paper can be viewed as a frequency-domain version of the technique from [29], and uses a more intuitive method of preventing cancellation of the targeted signal in the filtering process. This processing scheme linearly filters single-channel, frequency-domain data obtained from microphone array measurements in the manner of an optimal, data-dependent beamformer. The filtering preserves signal from user-defined spatial locations by leveraging the orthogonality between the targeted signal and all others present on the channels. Thus, single-channel outputs are produced at each targeted location which estimate only the signal from that location, even when multiple and/or distributed sources have been measured simultaneously. These single-channel, filtered outputs are available for further processing, examples being data-independent or -dependent beamforming [13], subspace methods [6,9,28], deconvolution [1–4,6–8,10–12], decomposition [5,9,30,31], or single-channel processing which does not employ array synthesis [32].

Section 2 defines the filtering equations used and their geometrical formulation for the preservation of targeted signal while minimizing the total signal on the channels. As the filtering requires other array channels which are used as

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