



Automatic 3D scanning surface generation for microphone array acoustic imaging



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ABSTRACT

This work presents a new technique for automatically generating the 3D scanning surface for acoustic imaging using microphone arrays. Acoustic images, or maps, of sound coming from spatially distributed sources, may be generated from microphone array data using algorithms such as beamforming. Traditional 2D acoustic maps can contain errors in the near-field if the object being imaged has a 3D shape. It has been shown that using the 3D surface geometry of an object as a scanning surface for beamforming can provide more accurate results. The methods used previously to generate this 3D scanning surface have either required existing CAD (Computer-Aided Design) models of the object being acoustically imaged or have required separate equipment which is generally bulky and expensive. The new method uses one or more cameras in the array, a data projector, and structured light code to automatically generate the 3D scanning surface. This has the advantage that it is inexpensive, can be incorporated as an add-on to existing microphone arrays, has short scan time, and is capable of being extended to imaging dynamic scenes. This technique is tested using beamforming and CLEAN-SC (CLEAN based on spatial Source Coherence) algorithms for a spherical array and an Underbrink multi-arm spiral array. For sound sources located about 1.2 m from the array, the mean position errors obtained are 6 mm. This is a quarter of the diameter of the mini-speakers being used as a sound sources.

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1. Introduction

The microphone phased array (also known as the acoustic camera) has been developed as a tool to enable the position and magnitude of sound sources to be identified. Microphone phased arrays are widely used by industries such as aeroplane and automotive manufacturers to identify sound sources. A commonly used acoustic imaging algorithm is beamforming [1,2], which uses delaying and summing of microphone channel data to obtain acoustic images or 'maps'. However, the beamforming maps contain an interference pattern artefact referred to as sidelobes. Image sharpening or deconvolution techniques have been developed to remove these sidelobes and attempt to obtain the true sound source distribution. Examples of these algorithms are DAMAS (Deconvolution Approach for the Mapping of Acoustic Sources) [3] and CLEAN-SC (CLEAN based on Source Coherence) [4]. The other methods used to generate acoustic maps from microphone phased array data are acoustic holography [5] and inverse methods [6].

1.1. Acoustic imaging using traditional 2D scanning surfaces

Acoustic imaging techniques, such as beamforming and deconvolution, have traditionally used the assumption that the acoustic sources lie on a plane. A 2D scan surface is used which is oriented perpendicular to the array's principle forward direction (the array Z-axis). This can lead to errors in the resulting acoustic maps if the sound sources are offset from the 2D scanning surface. These errors appear as projection/parallax errors in the plotting of the acoustic maps [7] and incorrectly estimated sound pressure levels (SPL) and location of sound sources [8–10]. These beamforming magnitude and position errors result from incorrect focus (time delays) being used for the beamforming.

1.2. Acoustic imaging using 3D scanning surfaces

Beamforming and deconvolution have been performed using a 3D grid [11–14]. However, unless the microphone arrays surround the object being imaged, there is poor resolution in the array Z-axis. Another problem is that these 3D grids can contain a large number of scan points, making deconvolution of these 3D grid beamforming maps very computationally expensive.

An alternative technique that has been developed is to use a scanning surface for beamforming which corresponds to the 3D

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surface geometry of the object that is being acoustically imaged. This method is said to provide the correct beamforming focus (time delays) for sound propagating from the surface of the object and should provide more accurate measured sound pressure levels, positions, and plotting. Different methods have been used to generate the 3D scanning surfaces for acoustic imaging. *Gfai* uses a CAD model of an object [15] or a laser scanner for acoustic imaging of the interior of rooms [16]. Pininfarina Full Scale Wind Tunnel tests obtained the 3D surface geometry of the exterior of cars, using stereoscopic imaging and a static pattern projected onto the object. Irimia et al. [10] (*MicroB/LMS*) uses a laser scanner, attached to a solid spherical array, to obtain the 3D geometry of the interior of cars for acoustic imaging. Another acoustic imaging technique that uses the surface geometry of an object is the acoustic holography technique referred to as conformal mapping. *Brüel & Kjær* measure the surface geometry of an object using a sonic contact measurement tool. However, this process can be slow and, therefore, is generally only applied to small areas [17].

1.3. New 3D acoustic imaging technique

The automatic 3D acoustic imaging technique presented in this work was developed as part of research comparing 2D and 3D acoustic imaging methods [26,29,30]. This work was the first to investigate deconvolution (image sharpening) of beamforming maps which were generated using the 3D surface geometry of an object as a scanning surface. A key requirement was the ability to accurately determine this 3D surface geometry relative to the microphone array. This paper presents a technique which was developed to allow the 3D scanning surface to be automatically generated using structured light scanning [18,19]. Structured light scanning was decided on because it is robust, inexpensive, fast, and capable of imaging dynamic scenes. Most microphone phased arrays contain a camera and hence only the addition of a data projector and software is required to convert these microphone arrays into a structured light scanner.

2. Acoustic imaging theory

Consider sound propagating through air from sound sources located at $\vec{X}_s$ to M microphones in an array having coordinates $\vec{X}_m$. The time domain microphone data may be converted into the frequency domain by dividing it into J blocks of length K , and for each data block obtaining a $(M \times 1)$ element array of FFT (Fast Fourier Transform) coefficients $\mathbf{U}$ for each frequency bin. The resulting frequency domain microphone data may be modeled as

$$\mathbf{U}(j) = \sum_{s=1}^S \mathbf{C}(\vec{X}_s) Q(\vec{X}_{sj}) + \mathbf{E}(j), \quad j = 1 \dots J \quad (1)$$

where $\mathbf{C}$ is the $(M \times 1)$ array propagation vector, Q is the sound source strength, and $\mathbf{E}$ is an $(M \times 1)$ element array of uncorrelated channel noise coefficients. A generic term for the array propagation vector may be given by

$$C_m(\vec{X}_s) = \frac{\exp(-i\omega\sigma_m(\vec{X}_s))}{A_m(\vec{X}_s)}, \quad (2)$$

where ω is the angular frequency, $\sigma_m(\vec{X}_s)$ is the acoustic propagation time from the sound source at $\vec{X}_s$ to a microphone at $\vec{X}_m$, and $A_m(\vec{X}_s)$ is a term to allow for spherical spreading [20].

A frequency domain beamforming $(N \times 1)$ acoustic map $\mathbf{b}$ may then be generated by defining a grid of scan points located at $\vec{\xi}$ and calculating

$$\mathbf{b}(\vec{\xi}_n) = \langle |\mathbf{w}^\dagger(\vec{\xi}_n) \mathbf{U}|^2 \rangle, \quad \vec{\xi}_n = \vec{\xi}_1 \dots \vec{\xi}_N \quad (3)$$

where $\langle \cdot \rangle$ is the time average and $\mathbf{w}$ is an $(M \times 1)$ array steering vector. A generic term for the m th element of the array steering vector may be given by

$$w_m(\vec{\xi}_n) = \frac{\exp(i\omega\tau_m(\vec{\xi}_n))}{D_m(\vec{X}_s)}, \quad (4)$$

where ω is the angular frequency, τ_m is the beamforming time delay, and D_m is an amplitude correction factor. Using Eq. (1) and assuming a unit magnitude source, one may model Eq. (3) as

$$\mathbf{b}(\vec{\xi}_n) = \mathbf{w}^\dagger(\vec{\xi}_n) \mathbf{C}(\vec{X}_s) \mathbf{C}^\dagger(\vec{X}_s) \mathbf{w}(\vec{\xi}_n). \quad (5)$$

Expanding this equation and substituting in Eqs. (2) and (15) gives

$$\mathbf{b}(\vec{\xi}_n) = \sum_{m,m'=1}^{M,M} \frac{\exp(i\omega[d\sigma_{mm'}(\vec{X}_s) - d\tau_{mm'}(\vec{\xi}_n)])}{D_m(\vec{\xi}_n)A_m(\vec{X}_s)A_{m'}(\vec{X}_s)D_{m'}(\vec{\xi}_n)} \quad (6)$$

where $d\sigma_{mm'}(\vec{X}_s)$ is the difference in the propagation time from the sound source at $\vec{X}_s$ to the two microphones respectively at coordinates $\vec{X}_m$ and $\vec{X}_{m'}$ and $d\tau_{mm'}(\vec{\xi}_n)$ is the corresponding difference in the beamforming time delays used for these two microphones for scan point $\vec{\xi}_n$.

2.1. Beamforming phase error

Traditionally acoustic maps have been generated using 2D scanning surfaces. If the scanning surface is offset from the sound source location, errors can occur in the amplitude and position of sound sources in beamforming acoustic maps. These errors may result from incorrect focus (time delays) being used for the beamforming. To explain this, one may consider sound waves propagating under free-field conditions through still, homogeneous air from a sound source located at position $\vec{X}_s$ to two microphones located respectively at positions $\vec{X}_m$ and $\vec{X}_{m'}$ in an array containing M microphones. The difference in acoustic propagation time between the two microphones may be described by

$$d\sigma_{mm'}(\vec{X}_s) = \frac{\|\vec{X}_m - \vec{X}_s\| - \|\vec{X}_{m'} - \vec{X}_s\|}{c}, \quad (7)$$

where c is the speed of sound.

Now consider that the beamforming is performed for the n th scan point with coordinate $\vec{\xi}_n$. The difference between the beamforming time delays for the two microphone signals would be

$$d\tau_{mm'}(\vec{\xi}_n) = \frac{\|\vec{X}_m - \vec{\xi}_n\| - \|\vec{X}_{m'} - \vec{\xi}_n\|}{c}. \quad (8)$$

The maximum constructive interference of the two beamformed microphone signals should occur when $d\sigma_{mm'}(\vec{X}_s) = d\tau_{mm'}(\vec{\xi}_n)$.

The beamforming phase error for two channels of microphone data may, therefore, be described by

$$\Delta_{mm'}(\vec{X}_s, \vec{\xi}_p) = \frac{2\pi c}{\lambda} [d\sigma_{mm'}(\vec{X}_s) - d\tau_{mm'}(\vec{\xi}_p)], \quad (9)$$

where λ is the wavelength and $\vec{\xi}_p$ is the theoretical coordinate in an acoustic map where a peak should occur due to a sound source at $\vec{X}_s$. A phase error of zero will give maximum constructive interference, while a phase error of π will give the maximum destructive interference of the beamformed microphone signals. Using perspective projection from the center of the microphone coordinates $\vec{X}_{mc}$ to a sound source at $\vec{X}_s$, the theoretical coordinates of $\vec{\xi}_p$ may be modeled as

$$\vec{\xi}_p = (\vec{X}_s - \vec{X}_{mc}) \frac{Z}{Z_s} + \vec{X}_{mc}, \quad (10)$$

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