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Design and implementation of a multi-octave-band audio camera for realtime diagnosis

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ABSTRACT

Noise pollution investigation takes advantage of two common methods of diagnosis: measurement using a Sound Level Meter and acoustical imaging. The former enables a detailed analysis of the surrounding noise spectrum whereas the latter is rather used for source localization. Both approaches complete each other, and merging them into a unique system, working in realtime, would offer new possibilities of dynamic diagnosis. This paper describes the design of a complete system for this purpose: imaging in realtime the acoustic field at different octave bands, with a convenient device. The acoustic field is sampled in time and space using an array of MEMS microphones. This recent technology enables a compact and fully digital design of the system. However, performing realtime imaging with resource-intensive algorithm on a large amount of measured data confronts with a technical challenge. This is overcome by executing the whole process on a Graphic Processing Unit, which has recently become an attractive device for parallel computing.

1. Introduction

Acoustic studies in various domains such as car, aircraft or train design, are mainly concerned by noise reduction. The latter is becoming stricter by both respect of regulations and passenger comfort. Most of these regulations go by standardized experiment protocols, and need the use of a Sound Level Meter (SLM). This instrument is the standard acoustic device for diagnosis, and gives an accurate description of the surrounding noise in terms of acoustic power. It provides the overall Sound Pressure Level (SPL), and octave or third-octave band levels for a finer spectral description. On another hand, source localization becomes an important tool for developers who want to reduce noise pollution from the physical origin. But no standardized protocol exists for such methods though some studies tend to it [1]. Anyway acoustical imaging remains important, but has been limited by both hardware complexity and a strong need for computing resources.

However, two recent technologies open up new perspectives for imaging systems. First, digital MicroElectroMechanical-Systems (MEMS) microphones are initially meant for general use devices implying speech acquisition *e.g.* telephones. But their high convenience for electronic design enables more versatile applications such as building an acoustic imaging array [2]. Indeed with these full digital components, the global hardware system is simplified. Secondly, General Purpose Graphic Processing Units (GPU) become an excellent solution for high parallel computing while being cost-effective and compact. The standard beamforming (BF) imaging algorithm is well adapted to parallel architecture and gives access to a fast enough code execution for a realtime visualization [3]. This has already been proved by several studies in ultrasound imaging [4–8] and in underwater acoustic imaging [9,10].

Taking advantage of these recent technologies, this paper describes a complete imaging system, from the data acquisition hardware to the signal processing and acoustic images display. Then this diagnosis tool intends to give an analysis consistent with classical SLM data. It maps, in realtime, acoustic levels according to standard octave bands. A previous study has explored the multi-frequency band imaging in realtime using GPU for versatile applications [11], using a classic acquisition system architecture.

The present work goes through two steps. First the hardware system is described; some calibration experiments performed to validate the suitability of MEMS microphones for the application in sight are also presented. Secondly, the design of the array beamformer is described, as well as its implementation on GPU. The achievability of realtime multiband imaging is discussed. Finally, we perform experimental tests in two different scenarios, to confirm the relevance of the presented diagnosis system.

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2. The data acquisition system

2.1. Hardware architecture

The acquisition system consists of two parts: the microphones array and an interface for communication with a host computer, as shown in Fig. 1. The array is made of 16 sets of 8 MEMS microphones, resulting in a 128 elements antenna. Its geometry is arbitrarily user configurable. The component chosen for pressure measurement is the ADMP441 microphone developed by Analog Device [12]. It allows a fully digital design of the system. Indeed the microchip includes the whole instrumentation chain, *ie* the transducer, an amplifier and a 24 bits $\Sigma\Delta$ converter. Compared with current systems, such an integrated sensor leads to a more condensed and simplified architecture.

The whole 128-microphone set is linked to the interface using the I2S serial protocol. It is driven with a user configurable sampling rate (f_s) common to the 128 acoustic channels, and synchronously sends the digital data. However the data flow must fit to standard computers for both convenience and compatibility issues. Therefore, the 24-bit integer samples are converted to 32-bit floats before multiplexing and transfer via the USB 2.0 serial bus. Finally a single time sample from the array consists of 512 Bytes, resulting in a total data flow rate of $512f_s$ Byte/s from the interface to the computer. With a sampling frequency $f_s = 50$ kHz the USB 2.0 bus is able to achieve a 25 MByte/s rate in *full-speed* mode.

Finally, the host computer receives the data to be processed for realtime imaging, and optionally saves them for further post-processing.

2.2. Acoustic assessment of the MEMS microphones

The ADMP441 microphones are initially intended for voice acquisition applications of general use [12], however their performances for aerial acoustic imaging applications are unknown. Previous works have made use of such components [2] but without concern for imaging. In this section the performances of these general purpose microphones are established in the specific field of acoustic imaging. Since the component datasheet does not provide accurate values for these particular specifications, a set of calibration experiments is performed.

2.2.1. Design of experiments

Four important specifications are investigated: (i) sensitivity, (ii) directivity, (iii) frequency response and (iv) self noise. Indeed the imaging algorithm rely on strong hypothesis: all the sensors are supposed to have the same sensitivity and to be omnidirectional; meeting at best these criteria guarantees better reconstruction capability. Besides, frequency response determines the spectral limits of the diagnosable sound field, and self noise sets the hearing threshold of the system.

The experiments are performed in the anechoic chamber of the *Laboratoire National d'Essai* (LNE) [13]. The MEMS microphones are assembled on a compact frame (Fig. 2). This frame is located 5 m away from a controlled source. This configuration allows to consider the impinging wave front to be of constant amplitude over the array, which makes a statistical analysis possible.

Using a standard loudspeaker emitting a white noise signal, 128 SPLs are measured, directly revealing the sensitivity variation of the microphones. Using the same source and rotating the frame over 180 degrees with a 10 degrees step, 128 azimuthal directivity patterns are derived. The frequency response is referenced to a reference microphone type *Bruel* & *Kjaer 4190*. The transfer function estimation method is followed [14], using an exponential chirp to provide high signal to noise ratio (SNR). Finally self noise is evaluated by acquiring component outputs in silence.

2.2.2. Results and discussion

In this section averaged results are presented in order to exhibit the global behavior of MEMS microphones (Fig. 3). Individual results are also presented in order to assess the statistical variation over the array.

The relative sensitivity variation is presented on the histogram, showing the 128 measured SPLs from the array during the experiment (Fig. 3a). This result straightly reflects the homogeneity among the microphones. In order to quantify the variation, a normal distribution (red curve in the figure) is fit over the histogram, its standard deviation equals 0.8 dB. This small variation can easily be handled by means of simple equalization coefficients.

The normalized frontal directivity is presented in Fig. 3b; the average pattern shows a maximum 1.7 dB deviation. Considering this small variation, the hypothesis of omni-directionality of the array is acceptable. Moreover the normalized individual patterns remain similar. So the averaged directivity pattern is representative of the global behavior of the microphones, and can be used if directivity correction is wanted according to the processed steering direction.

The frequency response is assessed on a frequency range which covers standard SLM octave bands implying typical community noises. Fig. 3c shows the evaluated frequency response of MEMS

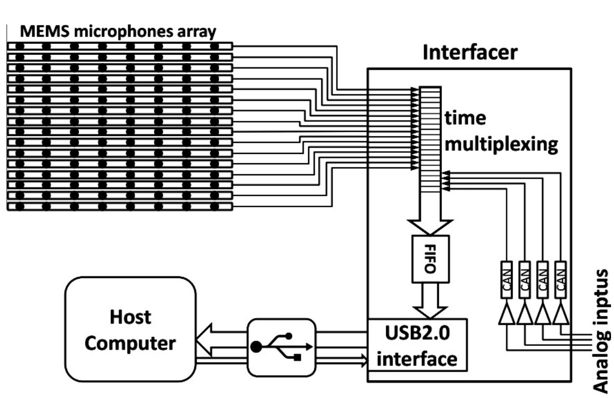

Fig. 1. Hardware architecture of the data acquisition system.

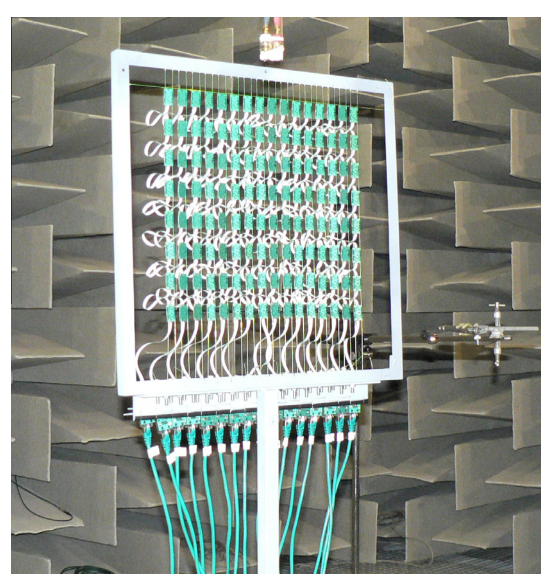

Fig. 2. MEMS assessment experimental setup in the anechoic chamber LNE – Frame dimensions: 50 cm × 50 cm.