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SMALL: SPHERICAL MICROPHONE ARRAYS LITTLE LIBRARY

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ABSTRACT

This paper presents the Spherical Microphone Arrays Little Library (SMALL) [1]. The latter consists of an open-source set of Computer Aided Designs to easily build Spherical Microphone Arrays prototype for research or home use. The designs are made with opensCAD, they are customizable and the different parts can be directly 3D printed. Extra standard mechanical assembly elements may be required such as screws or metal rods. Two main configurations are proposed: open sphere or rigid sphere. The customization parameters are the array radius, the number and position of the microphones with a set of predefined configuration, their type and extra options such as the possibility to embed an USB sound card inside the array. The microphone arrays signal processing for a Spherical Harmonic decomposition is provided as a code in the Faust language. Also, Python codes are provided for the calibration of the Spherical Microphone Arrays using a robotic arm. Preliminary measurements of the decomposition onto Spherical Harmonics are carried out for a 2$^{\text{nd}}$ order prototype.

Keywords: *spatial audio, ambisonics, spherical microphone arrays, fast prototyping*

1. INTRODUCTION

The use of Spherical Microphone Arrays (SMA)s to capture acoustic pressure fields is a proven and widely used technique [1]. The principle is to decompose the acoustic pressure field on the Spherical Harmonics (SH)s basis and process the resulting signals to analyse it. Various applications have been proposed, from capture manipulation and reproduction of 3D acoustic fields, i.e. Ambisonics [2], to beamforming [1] or spatial active noise control [3]. More recently, applications involving a distributed network of SMAs are studied, such as virtual navigation [4]. Several SMA prototypes are studied in the literature [5–7] and some commercial solutions exist with embedded acquisition system, such as the EigenMike© em32 or the Zylia© ZM-1. However, the prototypes in the literature are not easily reproducible due to the lack of open-source Computer-Aided Design (CAD) drawings and the commercial solutions do not always meet the requirements of the targeted application: array geometry, latency of the microphones and embedded acquisition system, maximal SHs decomposition order, price. In this context, the Spherical Microphone Arrays Little Library (SMALL) has been designed to facilitate the prototyping of SMAs for experimental applications in research laboratories or for personal use by hobbyists. The democratization of rapid prototyping equipment such as 3D printers and Micro Electro Mechanical Systems (MEMS) microphone breakout and acquisition boards allows to realize SMA prototypes with the SMALL library in a fast way and at low cost. Thus, it is hoped that this library will facilitate experimental research involving such devices in the field of spatial audio. The SMALL library is composed of three parts:

1. A set of CAD files allowing to easily design the mechanical structures for SMAs,

2. Software codes written in the Faust [2] language [8] that allow real-time processing of the microphone signals to produce the SHs components. These codes are part of the ambitools [3] plugin-suite [9],

[1] http://sekisushai.net/git/sekisushai/small/

[2] http://faust.grame.fr

[3] http://www.sekisushai.net/ambitools

3. Python codes for the calibration of SMAs using a robotic arm.

The paper presents the three aspects of the library in Sec. 2, Sec. 3 and Sec. 4 respectively. Conclusions and future works are discussed in Sec. 5.

2. SMA MECHANICAL STRUCTURE

The first part of the SMALL library is composed of CAD files for the SMAs mechanical structures. They are designed with the openSCAD software[4]. The latter allows parametric designs from code. Two configurations of SMA are available: rigid or open. A set of pre-established SMA configurations files is available to the user through JavaScript Object Notation (JSON) files that can be loaded into openSCAD. These files contain the coordinates of each microphone and various configuration parameters. The following sub-sections detail the specific features of each configuration.

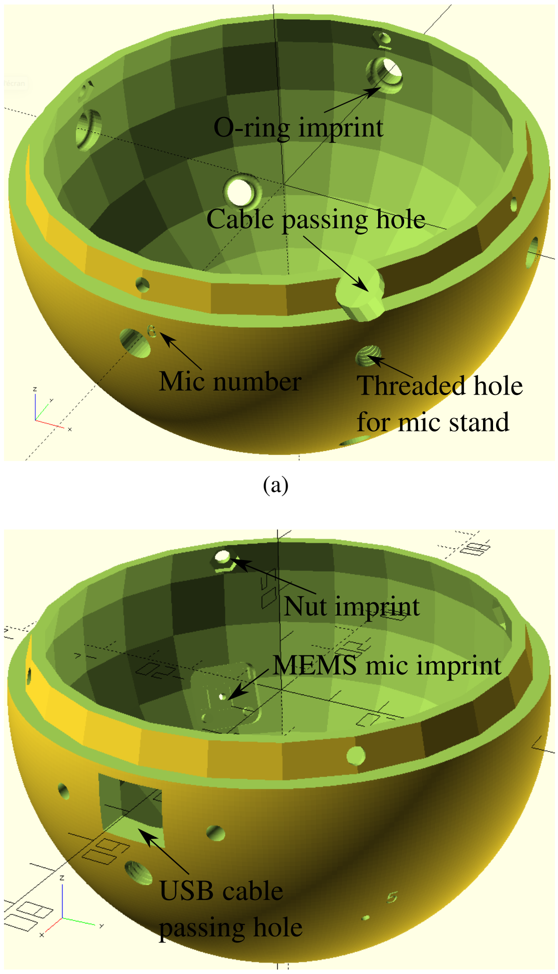
2.1 Rigid SMA

For the rigid SMA configuration, the user specifies the radius of the sphere and the direction of each microphone. The mechanical structure is made by assembling two fractions of spherical shells as shown in Fig. 1. The shell thickness is set by the user, and the latitude of the cutting plane is also adjustable. The position of a threaded hole, cable passing hole and screw and nuts imprints are adjustable in the vicinity of the cutting plane. In the current version, two shapes of microphone imprints are available:

- Cylindrical shaped microphones, for example 1/4” microphones with their preamplifier, such as B&K© 4958 mics,
- MEMS microphone soldered on their breakout board, such as Adafruit© MEMS breakout boards.

The identifying number of each microphone is engraved on the outer surface of the shell. The cylindrical-shaped microphones are held in place with an O-ring mounted inside the shell (see. Fig. 1a). The latter realizes the air sealing. For MEMS microphones (see Fig. 1b) a silicone gasket is applied after assembly for gluing and air sealing. If one uses digital MEMS microphone breakouts board with Integrated Interchip Sound (I2S), Pulse Density Modulation (PDM) or Time-Division Multiplexing (TDM) protocol, it is possible to use a miniDSP© MCHStreamer USB

[4] https://openscad.org/



(a)

(b)

Figure 1: Two examples of the bottom part of an SMA for the rigid configuration: (a) for cylindrical-shaped microphones, (b) for MEMS PDM microphones and USB embedded sound card.

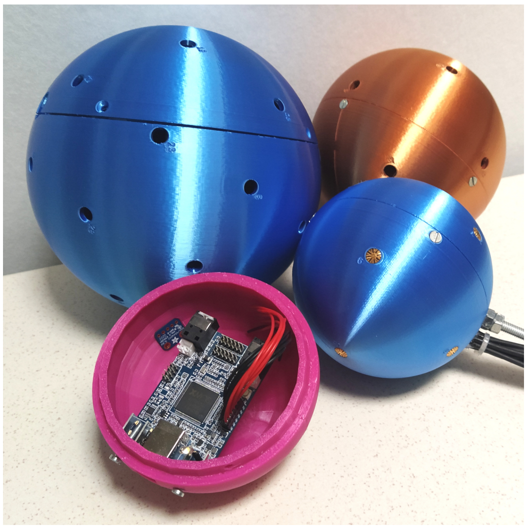


Figure 2: Several prototypes of rigid SMAs: the blue ones and the orange one are intended for 1/4” microphones. The pink one is intented for Adafruit© PDM microphone breakout boards and embeds a miniDSP© MHStreamer sound card. The shells are made in plastic with an 3D printer.

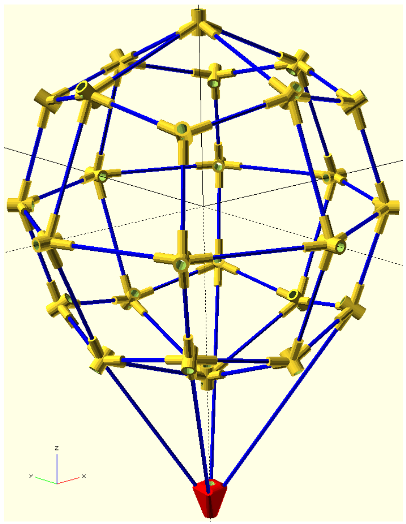


Figure 3: Example of an open SMA for a Lebedev mesh with 26 microphones. The 1/4” microphones are mounted in the holding parts (yellow). These parts are connected to each other with metal rods (in blue). A threaded part for attaching the SMA to a microphone stand (in red) is also connected to the adjacent microphones.

Sound Card embedded inside the spherical shell. Figure 2 shows the realization of different rigid SMA prototypes.

2.2 Open SMA

The open SMA configuration is realized by means of microphone holding parts connected to each other with thin metal rods. In the current version, only cylindrical microphones can be used. Two examples can be seen, in Fig. 3 for a Lebedev mesh with 26 nodes and in Fig. 4 for a tetrahedral mesh. The user indicates the coordinates of each microphone, as well as the associated connection matrix. The latter informs for each microphone the neighbors to which they should be connected. The microphone holding parts are then automatically generated. In fact, this design approach allows realizing open arrays of arbitrary geometry. In particular, it is possible to add additional microphones inside the open sphere, for the configuration of Fig. 4. The SMA is fixed to stand by the mean of a threaded part, attached to neighbouring microphones, as shown in Fig. 3 and 4.

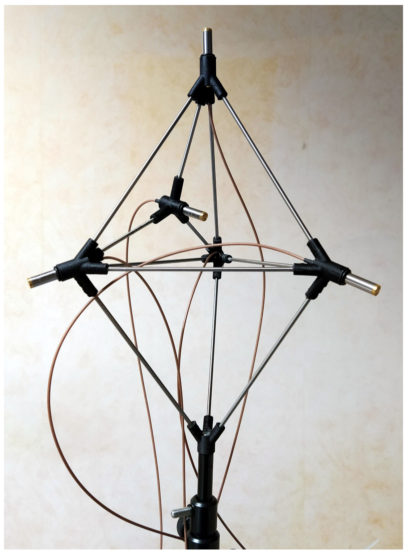


Figure 4: Prototype of an open SMA configuration with 4 B&K© 4958 microphones in a tetrahedral arrangement plus a fifth microphone pointing at the sphere center.

3. SMA SIGNAL PROCESSING WITH FAUST

The second part of the SMALL library is composed of Faust codes, from the ambitools plugin-suite [9] to convert the microphones signals onto the SHs signals on the fly. The Faust language allows describing the Digital Signal Processing (DSP) algorithms in the time domain and compile them to a multitude of audio plugins formats or even for embedded system such as Field Programmable Gates Array (FPGA) [10]. Thus, it becomes possible to make real-time applications involving one or more SMAs, such as spatial active noise control [11]. The rest of this section gives more insight to the SMA DSP to retrive the SHs signals.

3.1 Microphone signals to theoretical SHs signals

One considers a spherical array of radius r with N omnidirectional microphones disposed on a mesh able to perform Discrete Spherical Harmonic Transform (DSFT) up to order L [1]. In the frequency domain, the pressures captured by the microphones are denoted $\mathbf{s} \in \mathbb{C}^{N\times 1}$ with:

$$\mathbf{s} = [s_1, \cdots, s_N]^T \tag{1}$$

where ^{T} is the transpose operator. Then, the decomposition onto the SHs is done in two steps: Firstly, the DSFT, denoted $\mathbf{Y}^{\text{"}-1\text{"}} \in \mathbb{R}^{(L+1^2)\times N}$ is operated on $\mathbf{s}$. Secondly, the resulting vector is weighted with radial filters, denoted $\mathbf{E} \in \mathbb{C}^{(L+1)^2\times(L+1)^1}$ such that:

$$\tilde{\mathbf{b}} = \mathbf{E}\mathbf{Y}^{\text{"}-1\text{"}}\mathbf{s} \tag{2}$$

In Eqn (2), $\tilde{\mathbf{b}} \in \mathbb{C}^{(L+1)^2\times 1}$ are the estimated SHs signals. The matrix $\mathbf{Y}^{\text{“}-1\text{”}}$ is obtained by the inversion of the matrix of SHs evaluated in the microphones directions up to order L, denoted $\mathbf{Y} \in \mathbb{R}^{N\times(L+1^2)}$. Depending on the spherical mesh used the inversion can have the following forms [12]:

$$\mathbf{Y}^{\text{“}-1\text{”}} = \begin{cases} \mathbf{YW} & \text{cubature} \\ (\mathbf{YY}^T)^{-1}\mathbf{Y} & \text{least-square} \\ (\mathbf{YWY}^T)^{-1}\mathbf{YW} & \text{weighted least-square} \end{cases} \tag{3}$$

where $\mathbf{W} \in \mathbb{R}^{N\times N}$ is a diagonal matrix of microphones weights ensuring the orthonormality of the DSFT. In the current ambitools version, several matrices $\mathbf{Y}^{\text{“}-1\text{”}}$ are proposed, mainly for cubature rules [13]. The matrix $\mathbf{E}$ is a diagonal matrix of radial filters such that :

$$\mathbf{E} = \mathrm{diag}([e_0(kr), \underbrace{e_l(kr), \cdots e_l(kr)}_{(2l+1)\text{ times}}, \cdots, \underbrace{e_L(kr), \cdots, e_L(kr)}_{(2L+1)\text{ times}}]) \tag{4}$$

where k is the wave number and the radial filters $e_l(kr)$ are defined by:

$$e_l(kr) = \begin{cases} i^{1-l}(kr)^2 h_l'(kr) & \text{rigid} \\ \dfrac{i^{-l}}{j_l(kr)} & \text{open} \end{cases} \tag{5}$$

In practice, the radial filters of Eqn. (5) can not be implemented directly as they have infinite gain a some frequencies. In the current ambitools version, only the rigid configuration filters are available as stable parametric Infinite Impulse Response (IIR) filters where the theoretical filters are stabilized with phase-matched high-pass filters [14]. For these filters, the user can set the array radius r and maximal amplification gain at runtime.

4. SMA CALIBRATION TOOLS

The third part of the SMALL library is composed of Python codes to achieve the SMA calibration using a robotic arm. In fact, the DSP presented in Sec. 3 applies for theoretical SMAs and in practice, the microphone frequency responses, their positioning and the scattering of the mechanical structure of the SMA differs from their theoretical modeling. If one is able to measure the SMA responses to an acoustic wave with variable Directions of Arrival (DOA)s, a matrix of encoding filters $\mathbf{H} \in \mathbb{C}^{(L+1)^2\times N}$ can be obtained [6, 12, 15] such that:

$$\mathbf{b} = \mathbf{Hs} \tag{6}$$

where $\mathbf{b} \in \mathbb{C}^{(L+1)^2\times 1}$ are the SHs signals. In practice, calibration of a SMA requires an anechoic environment and a loudspeaker which can move on a spherical surface around the SMA [15]. This experimental setup can be cumbersome to implement. In [16], the author proposes to use a robotic arm to rotate the SMA around its center in both azimutal and elevation relatively to a fixed loudspeaker. This latter approach is chosen in the SMALL library.

Figure 5: A rigid SMA prototype (in blue) mounted on a 6-DOF robotic arm for directional response measurements with a fixed loudspeaker.

4.1 Robotic arm modification

The robotic arm used for the measurements is based on TrossenRobotic© X-series with 6 Degrees of Freedom (DOF)s. These robotic arms are compatible with the open-source Robot Operatic System (ROS) [5] and can be driven with a Python Application Programming Interface (API).

To adapt this type of robot for an SMA calibration purpose, the arm gripper has been removed and replaced with a threaded rod on which the SMA is fixed, as visible in Fig. 5. Accordingly, the arm forward kinematics description [17] is adapted and provided in the SMALL library. In particular, the "end-effector" (i.e. the gripper) frame is placed at the SMA center. This frame is denoted the "SMA frame" in the rest of the paper. Thus, the task for the arm is to pose the SMA frame relatively to the robot base frame at a chosen position and orientation. This allows to define an orientation of the SMA with respect to the loudspeaker and proceed with the acoustic SMA Impulse Responses (IR)s measurement. However, the inverse kinematic solution does not always exists for all possible orientations of the SMA frame around a point. For a robot geometric configuration, finding the point for the SMA frame which can be attained with a maximal number of pose is not a trivial task. An approach can be to use a reachability map of the robot, which is left for future investigation [18]. For now, the SMA frame coordinate is chosen by trial and error.

[5] https://www.ros.org/

4.2 Calibration measurements protocol

After alignment of the SMA and the loudspeaker frame with the help of lasers, the directional response of the SMA proceeds sequentially for all chosen orientations in two steps with a Python routine: First, the robotic arm sets the SMA frame to a specific orientation. Then, SMA IRs are measured with an exponential sine sweep method [19]. The same measurements are carried out without the SMA mounted, using a reference microphone placed at the SMA frame center. This allows to deconvolve the loudspeaker response and reduce the arm scattering. The resulting measurements allow to establish the matrix $\mathbf{H}$ of Eqn (6) [12].

4.3 Preliminary measurements with a 2^{nd} order rigid SMA

In this section, preliminary measurements are carried out for a rigid SMA prototype. The latter uses a 10-nodes Popov cubature rule which allows a DSFT up to $L = 2$ [20]. The microphones placed at the cubature nodes are 1/4" B&K© 4958. The robotic arm is a Trossen-Robotics© WidowX 250 Robot Arm 6DOF adapted to the SMA measurement purpose, as described in Sec. 4.1. A picture of the experimental setup is shown in Fig. 5. The IRs measurements are carried out for horizontal incidences $\phi \in [-180°180°]$ every 10°.The resulting microphones IRs are converted to SHs signals using theoretical model of Eqn. (2), with radial filters of Eqn. (5). The measured directivity for the SHs with a horizontal component are shown in Fig. 6 at 1.5 kHz in red solid lines. Theoretical directivities are also shown as dashed blue lines. The reconstruction of the SHs components at this frequency can be observed: the main lobes are similar in width and direction to the theory and there are no extraneous side lobes, although the DSP used is that of a theoretical microphone without defects.

5. CONCLUSIONS AND FUTURE WORKS

In this paper the SMALL library has been presented. It allows to quickly realize prototypes of rigid or open SMA, in a multitude of configurations. CAD files, tools for the decomposition on spherical harmonics are and tools

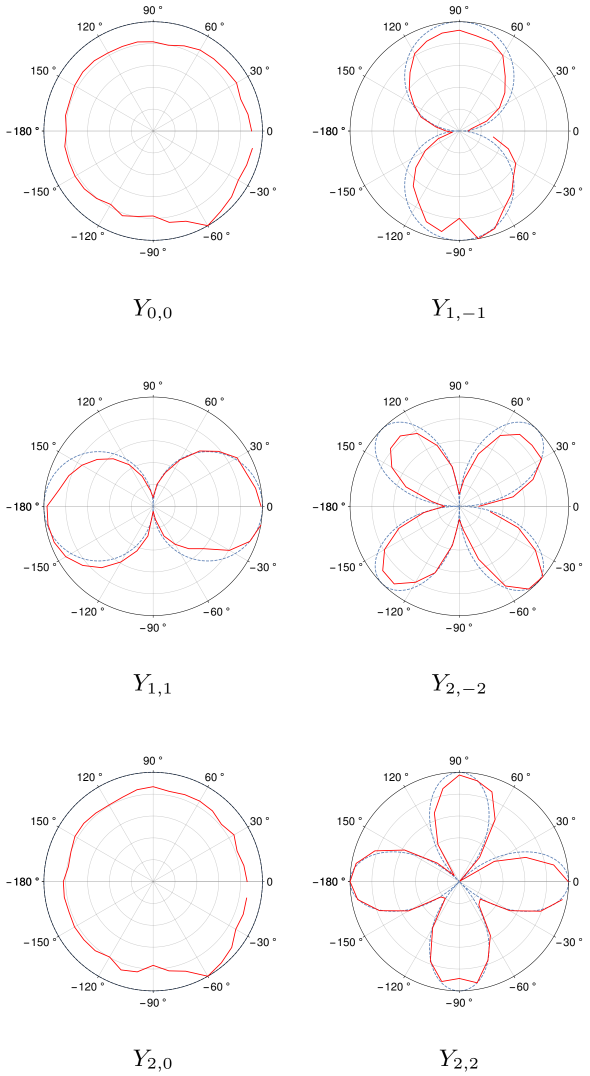


Figure 6: Normalized experimental polar responses (solid red curves) and theoretical (blue dashed curves) for several spherical harmonics at 1.5 KHz.

for calibration with a robotic arm are provided. Preliminary directional measurements for a 2nd order prototype were presented and shows promising results. Future work will focus on calibrating a series of prototypes made with SMALL and integrating the resulting filters into a database.

6. ACKNOWLEDGMENTS

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