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## Introduction

With the advent of spatial audio for virtual reality and 360° video, software tools have enabled users to synthesize higher order ambisonic material in Digital Audio Workstations (DAWs) in order to create immersive, full-sphere, sound fields.

The availability of microphones capable of capturing such signals has not grown at the same pace, or affordability. This work aims to provide the tools needed to successfully develop such arrays; gaining a greater understanding of these performance capabilities and limitations.

## An Ideal Ambisonic Microphone

A theoretically ideal ambisonic microphone array would have...

- Coincident Microphone Capsules
- Capsule w/ Flat Frequency Response (*Regardless of the angle of incidence*)

It isn't physically possible to have coincident capsules as multiple capsules cannot occupy the same physical space. Additionally, capsules even when matched, won't exhibit the desired frequency and polar response across the audible frequency range.

**Q : How can we *Attempt* to account for these physical inadequacies?**

**A : Informed Design & Pre Filtering / Post Filtering**

## MATLAB® Array Simulation

A MATLAB® routine has been developed that simulates the response of an array given a specific set of design attributes, these are set by the user via a graphical user interface (GUI). These attributes (shown below) are used to generate a set of impulse responses (IRs) that can be used to evaluate an arrays theoretical performance.

- |                      |             |                           |               |
|----------------------|-------------|---------------------------|---------------|
| ○ Ambisonic Order    | N           | ○ Desired IR Length       | (samples)     |
| ○ Array Radius       | (mm)        | ○ Sample Rate             | (kHz)         |
| ○ Array Type         | (2D / 3D)   | ○ Virtual Source Distance | (m)           |
| ○ Sampling Scheme    |             | ○ Source Increment        | (Degrees °)   |
| ○ Directivity Factor | (d = 0 - 2) | ○ Capsule Directivity     | – (d = 0 - 2) |

## The Ambisonic Array Design Tool (AADT)

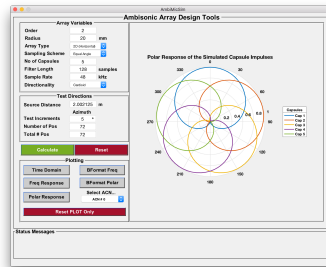


Figure 1 – AADT – Polar Response

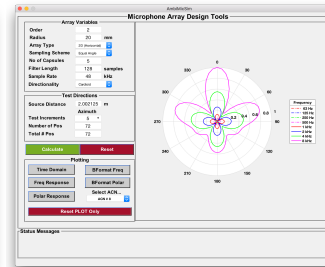


Figure 2 – B-Format Polar Response

The AADT is an application that packages the array simulation routine into a standalone GUI using the MATLAB® GUI Development Environment [1].

The current incarnation of the AADT utilizes the Equal-Angle sampling scheme in the horizontal plane only (thus simulating a circular arrays). The reason for opting for horizontal only capability in the first instance was so factors such as spacing, calibration and filtering could be evaluated with greater ease. Findings could then be used to inform the development of a 3D spherical array.

The AADT is currently capable of generating plots of the following five responses.

### Simulated Capsule Signals

1. Time Domain
2. Frequency Domain
3. Polar Response (See Fig 1)

### B-Format Signals

1. Frequency Domain
2. Polar Response (Fig 2)

## The Ambisonic Array Evaluation Tool (AAET)

The Ambisonic Array Evaluation Tool differs to its counterpart in that its purpose is to evaluate / validate a physical arrays performance against a comparable simulation.

In conjunction with the development AAET application a five channel prototype array was developed (Fig. 3). A set of IRs were measured in an hemi-anechoic chamber, These IRs were loaded into the AAET via a .mat file for direct comparison.

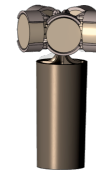


Figure 3 - Model of 3D Printed Prototype

## AAET (Continued)

Along with the array performance being evaluated it is possible to visualise the response of the individual capsules, pre and post filtering.

Two initial approaches to pre-filtering have been implemented in this application. These are the optimisation of the capsule signals by means of on-axis capsule calibration, with the desired on-axis response being either flat across the audible frequency range or matching of capsules to an individual capsule response.

These filters are calculated within the application by utilising Nelson - Kirkeby Inversion with Regularisation [2].

The effect of this Pre-Filtering can be viewed by selecting the desired filtering approach from a drop down menu

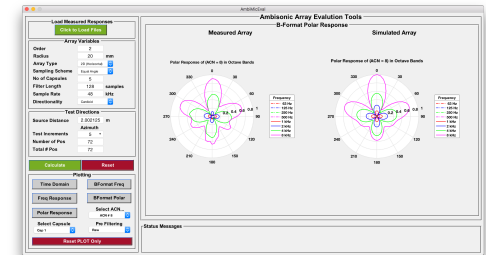


Fig. 4 – AAET – B-Format Polar Response

## Future Work

### Implementing Additional Features

- |                                 |  |
|---------------------------------|--|
| ○ Full 3D Simulation capability | ○ Frequency dependent directivity factor |
| ○ Scattering on a Rigid Sphere  | ○ Additional Pre-Filtering Methods       |
| ○ Sampling schemes              | > Averaged Gain Matching                 |
| > Near Uniform                  | > Diffuse Field Equalization [3]         |
| > Gaussian                      | ○ Post Filtering of SH Signals           |
| > Spherical T-Designs           | ○ Capsule Specific Radius Calculator     |

## References

- [1] Mathworks Inc, "[MATLAB]® App Building R2018a" <http://bit.ly/MathworksGUI>
- [2] H. Tokuno, O. Kirkeby, P. Nelson, "Inverse filter of sound reproduction systems using regularization" IEICE Transactions on Fundamentals of Electronics, Communications and Computer Sciences, Vol. 80, No. 5. 809-820. (1997).
- [3] Heller, A.J; Benjamin, E. M; Calibration of Soundfield Microphones using the Diffuse-Field Response. In Audio Engineering Society Convention 133, 8711, 2012.

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