UNIUNEA EUROPEANĂ

GUVERNUL ROMÂNIEI
MINISTERUL MUNCII, FAMILIEI, PROTECȚIEI SOCIALE ȘI PERSOANELOR VÂRSTNICE
AMPOSDRU

Fondul Social European
POS DRU 2007-2013

Instrumente Structurale
2007-2013

MINISTERUL EDUCAȚIEI NAȚIONALE
OIPOSDRU

UNIVERSITATEA TEHNICĂ DIN CLUJ-NAPOCA

Investing in people!
EUROPEAN SOCIAL FUND
Project co-funded by the European Social Fund through the Sectorial Operational Program Human Resources 2007 - 2013
Priority Axis 1: "Education and training in support for growth and development of knowledge based society"
Key Area of Intervention 1.5 "Doctoral and post-doctoral programmes in support of research"
Project: "QDOC – Improvement of the doctoral studies quality in engineering science for development of the knowledge based society"
Contract No.: POSDRU/107/1.5/S/78534
Beneficiary: Technical University of Cluj-Napoca

Faculty of Electronics, Telecommunications and Information Technology

Eng. **Flaviu Ilie BOB**

PhD Thesis

Sound Recording and Processing for Determining Source and Signal Parameters

Resume

Scientific Coordinator:
Prof. **Corneliu RUSU**

2014

Content

Keywords

Three microphones system, sound source localization, acoustic ranging, three channels recordings, time differences of arrival.

Introduction and motivation of the thesis

The human auditory system is very complex. Using the great computational power of the brain, one can estimate quite precisely the distance towards a sound emission. To do this, there is prior knowledge needed for environmental and source information. Such information is stored inside the human brain, like in a real database.

Similar to the human auditory system, artificial systems are also able to localize in terms of distance a sound source. A processor is needed to act as a "brain", a large amount of memory for the "database" and a set of audio transducers for the "ears". Even if it sounds simple, the current state of research provides several solutions that are either voluminous, expensive, complex or dedicated for certain applications. Other inconveniences are that they have a limited range or they need previously obtained statistics about the sound, the source or the environment.

For artificial intelligence to make assumptions about the environment properties and retain such a great amount of data for sound recognition is really challenging and implies unjustified costs. Unknown single acoustic emission localization in terms of distance is a complex problem. Simple solutions that do not require prior training are not able to estimate the distance at high ranges. Most solutions available are limited in range (up to 2 meters) for systems that have the same size (of 2 meters). Slightly bigger range can be determined using complex systems. These are expensive and difficult to use considering size and cost.

Thesis Objectives

The main objective of this thesis is to design a solution that uses the minimum resources in order to achieve the goal of single unknown sound source localization. This solution should be compact and portable, localization relying only on the received sound, but without prior knowledge about the sound emission or environment.

The second objective is to prove that the designed solution is viable. A set of preliminary experiments will prove the capabilities of the proposed system. In order to obtain better results and to improve the embedded system, a communication interface with a computer will be used. It allows signal recording for further use and offers the possibility of studying the processed signals. Simulations will be performed, highlighting the available possibilities and leading to enhancement of the localization system.

For the final objective another experiment session will be conducted at higher ranges with the improvements obtained before. All the experimental instances will be recorded for future usage, resulting in a full feature database. To claim the performance of our solution the set of recorded signals will be processed using some of the most popular methods, leading to the achievement of best results.

Thesis structure

The thesis is structured in two parts and five chapters comprising the following information:

- The first part contains general knowledge about the problem that is being investigated:

- In Chapter 1 is presented the general sound source localization problem. Objectives of the thesis are proposed before stating the thesis outline and the author's personal contributions.
- The current state of research is presented in Chapter 2. This chapter aims to provide the reader with the necessary background information while familiarizing him with the existing sound source localization systems, methods and algorithms.

• The second part presents the author's contributions. The proposed method is tested, improved and tested again in order to obtain the best possible results.

- In Chapter 3 the author proposes a solution for the established goal, a sound source ranging system and its associated realizations: principle, method, developed embedded software and hardware.
- As the proposed solution has to be verified, in Chapter 4 are presented the preliminary experiments. After the experimental setup, the results are followed by conclusions that highlight the sound source ranging capabilities of the system.
- Chapter 5 exposes improvements brought to the system. The computer communication and application design have the greatest impact in our research. They lead to an upgrade of the hardware, mechanic design, embedded software and signal processing. In the following, studies and simulations of the system are performed in order to further improve its capabilities and establish its limitations.
- Another set of performed experiments is presented in Chapter 6, this time using increased distances and making use of the improvements stated in Chapter 5. The experimental setup is followed by the obtained results. Some methods of determining the *Time Delays Of Arrival* available in the literature are used here and compared against each other in order to obtain the best possible results. In the end some conclusions are drawn.

Personal Contributions

The work that has been done for this thesis permits the statement of the following contributions in the field of sound source localization:

1. The most important contribution is the implementation of a system capable of performing two-dimensional localization (in terms of distance) of a single unknown sound emission in an homogeneous and free-field environment. The solution is compact and uses the minimum required resources. This achievement has several sub-contributions, as follows:

   (a) An embedded software solution was designed for usage with three conversion channels for three microphones. We also implemented the distance and angle calculation algorithm in the embedded software, despite the constraints of the μController. A hardware design was provided and the design step was concluded by the physical implementation of the system. The implemented solution was subject of preliminary tests at different angle orientations for distances up to almost 6 meters, above the ones presented in the specialized literature.

   (b) A Matlab® interface was designed for ease of both debugging and development, opening doors for further improvements and for further more complex signal processing. Improvements were brought to the tested solution in order to better evaluate its performance and further improve the results. Also simulations that highlight the system's capabilities were performed.

(c) Final experiments provide a database of 3-channel recordings, at a high sampling rate, for distances up to 50 meters and at different angle orientations. The experimental results confirm that the unknown sound source ranging system is suitable for distances up to 35 meters, at angles from -45 to 45 degrees with *mean errors* and *maximum errors* not exceeding 10% and respectively 25%.

2. Several methods for determining the *Time Delays Of Arrival* that are found in the literature are tested and compared using our database of recorded sounds. The most robust method has proven to be the *General Cross Correlation* without any filtering.

3. Another contribution worth mentioning is the bibliographic study of synthesizing the systems, methods and algorithms used in sound source localization. It provides a general perspective over the domain and the directions that can be exploited in order to propose new methods for improving their quality.

Publications List

Articles

F. I. Bob, "Hearing: From Sound to Nerve Response", *Novice Insights in Electronics, Communications and Information Technology Magazine*, vol. 11, no. 11, pp. 9–18, 2011, iSSN: 1842-6085.

F. I. Bob, "Three Microphones Embedded System for Single Unknown Sound Source Localization", *Carpathian Journal of Electronic and Computer Engineering - CJECE*, vol. 5, no. 1, pp. 19–24, September 2012, iSSN 1844-9689.

F. I. Bob, "Long Range Sound Source Localization Experiments", *Acta Technica Napocensis - Electronics and Telecommunications*, vol. 55, no. 2, June 2014, submitted paper (under evaluation).

Proceedings conferences

F. I. Bob, N. C. Pampu, and L. T. Chira, "Improving Analog-to-Digital Converter's Resolution Using the Oversampling Technique", in *Signal Processing and Applied Mathematics for Electronics and Communications - SPAMEC 2011*, Cluj-Napoca, Romania, August 26-28 2011, pp. 57–60.

F. I. Bob, "Embedded Solution for Universal Acoustic Source Distance Localization Using Three Microphones", in *International Symposium on Electronics and Telecommunications - ISETC 2012*. "Politehnica" University of Timisoara, November 2012, iSBN 978-1-4673-1175-5.

Other scientific activity publications

F. I. Bob, "Dynamic Equilibration System Using a μController", *Novice Insights in Electronics, Communications and Information Technology Magazine*, vol. 10, no. 10, pp. 120–124, 2011, iSSN: 1842-6085.