

Selective Listening

ECE 445 Project Proposal - Spring 2021

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Contents

1	Introduction	2
1.1	Objective	2
1.2	Background	2
1.3	Physical Design	3
1.4	High-level Requirements	4
2	Design	4
2.1	Listener	4
2.1.1	Binaural Headphones	4
2.1.2	Microphone Array	4
2.2	Audio Codec	5
2.2.1	Pre-Amplifiers	5
2.2.2	ADC	5
2.2.3	DAC	5
2.3	Volume Control	5
2.3.1	Volume Sliders	5
2.3.2	Microcontroller	5
2.4	Audio Processing Board	6
2.4.1	FPGA	6
2.4.2	ARM Core	6
2.5	Block Diagrams	7
2.6	Risk Analysis	8
3	Ethics and Safety	9
4	References	10

1 Introduction

1.1 Objective

Imagine you are in a noisy environment (a restaurant, concert venue, etc) and you want to listen to one person or “source” in particular. For individuals who struggle with hearing loss this can be a challenging situation, even with modern hearing aids. We want to continue work done by Ryan Corey on the Augmented Listening Project and create a system that will allow the user to listen to a single source at a time, when many sources are present.

Our proposed solution involves a few different subsystems to selectively hear a unique source. We will create a pair of binaural headphones along with a microphone array that the user can wear to hear the desired source. Additionally, we will create a “Volume Control Interface” to manage the volume on one or multiple sources. Finally, we will have additional circuitry using ADCs, Amplifiers, etc. to support the microphones. Much of the software/firmware of the project will be on an FPGA (continuing the work of students in the research group), and the DSP algorithms we will use come from Ryan Corey’s thesis.

1.2 Background

Hearing aid devices have long been shrouded in proprietary technology. The Augmented Listening Laboratory is creating an open source listening platform to improve hearing aid technology. Standard hearing aid devices typically contain only two microphones close to the ears. Hearing aid performance can be improved by adding multiple microphones surrounding the headpiece in the relevant space near the target source. Our demonstration will show off the power of adding many microphones by using a directional listening technique called “beamforming”.

Returning back to the noisy restaurant example presented in the beginning, with typical hearing aids the listener would have a difficult time hearing their friends in front of them. The background noise of the kitchen and other conversations would be picked up by their ear pieces. Our solution solves this problem by giving the listener control over the conversations and noises they hear.

1.3 Physical Design

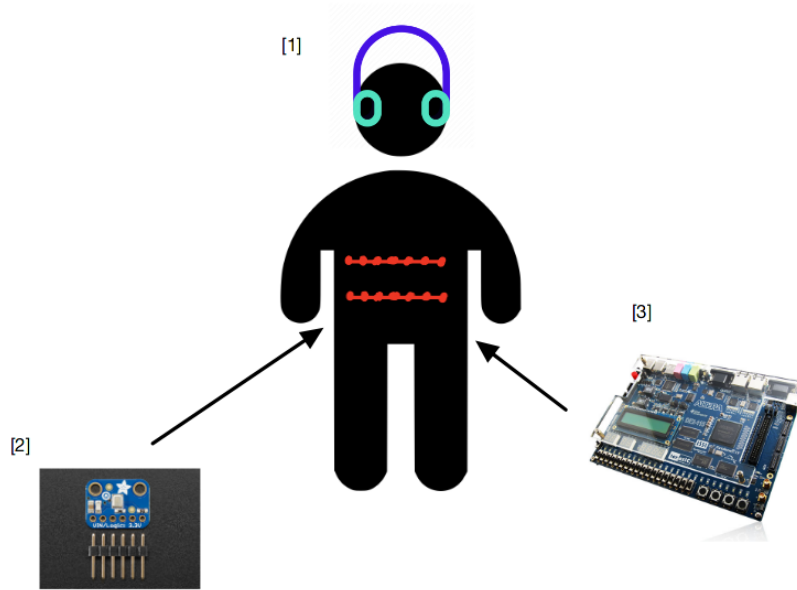


Figure 1: User diagram

1: The binaural headphones ([1] in figure 1) will be worn by the listener in order to hear in a particular direction and preserve spatial cues. In this way, the listener will be able to naturally recognize noise regardless of their orientation in the sound field.

2: The microphone array will consist of 14 digital MEMS microphones [2] directly connected to the FPGA GPIO pins. These microphones will provide additional information to accurately determine source location.

3: The FPGA [3] will be portable (albeit wired to the power source) audio processing unit capable of filtering audio and providing unique coefficients to construct the desired beamformer.

1.4 High-level Requirements

This project will be successful if the listener can distinctly hear each unique source one at a time in our demo using a mixing board. In addition, the user must be able to have control over the volume of the source they are tuning into. The total delay through our system should be no more than 10ms to avoid disorientation. The binaural headphones must preserve spatial awareness and directionality of the selected source.

2 Design

2.1 Listener

The listener is the user of our wearable microphone array and audio processing system. They will wear a pair of binaural headphones and affix a microphone grid array to their chest.

2.1.1 Binaural Headphones

The binaural headphones will serve two purposes in our design. The user will listen to the processed audio through the earbuds, and the microphones at each ear will serve as a part of the large microphone array. Most binaural headphones have analog microphones at the ears, and we will also use one with analog mics.

Requirement: Microphones are close to the ear canals and are comfortable.

2.1.2 Microphone Array

The main microphone array will be a static grid of 14 small MEMS microphones. MEMS microphones have a built in ADC so they will connect directly to the FPGA. We chose 14 microphones here to keep a total array size of 16, including the binaural microphones.

Requirement 1: Cabling between microphones must not be uncomfortable for the user.

Requirement 2: Microphones must be spaced at least 8cm apart.

2.2 Audio Codec

2.2.1 Pre-Amplifiers

A pre-amplifier is needed to amplify the two binaural microphone voltage levels to line level. Most audio codec ICs do not have a suitable amplifier circuit built in. We have evaluated some chips with a stereo amp, but it is not worth limiting our selection so much.

Requirement 1: Amplifier must increase the microphones to 1V peak-to-peak

Requirement 2: Amplifier must have inaudible self noise

2.2.2 ADC

Our design will feature 2 ADCs in the form of our audio codec IC. These are necessary to sample our analog mics to create an I²S stream to the FPGA.

Requirement 1: ADC must accept 1V_{pp} line level input

Requirement 2: ADC must sample at 48kHz/16-bit

2.2.3 DAC

A stereo DAC is necessary to drive the stereo binaural headphones. The audio codec IC will have a pair of DACs built in, which we will connect up to the FPGA I²S output.

Requirement 1: DAC can convert 48kHz/16-bit samples

Requirement 2: DAC must have inaudible self noise

2.3 Volume Control

2.3.1 Volume Sliders

The user interface will be made up of four volume sliders. Three will be dedicated to adjusting each of the source volumes, and the fourth will be the master volume out level. We will use fader potentiometers for these sliders.

2.3.2 Microcontroller

Our microcontroller will only serve to read the level of the volume sliders and relay that information to the FPGA. Most microcontrollers have both an ADC and an I²C interface which we will need.

Requirement 1: Sufficiently high resolution ADC (most likely 8-bit) for smooth volume adjustment

Requirement 2: Standard mode or better I²C interface

2.4 Audio Processing Board

2.4.1 FPGA

Our audio processing system is comprised mainly of an FPGA and a single ARM core. The FPGA has the purpose of interfacing with the I²S microphones, recording audio to memory, and accelerating the FIR filtering.

Requirement 1: Sub 10ms latency through filter banks

Requirement 2: 512 length FIR filters

Requirement 3: -20dB attenuation of sources not being listened to

2.4.2 ARM Core

The actual signal processing algorithms will be run on the ARM core. For our specific demo, the ARM core will be used to generate a set of filter coefficients to be loaded to the FPGA FIR banks. This will be a typical use of the platform.

Requirement 1: 50ms time to calculate filter coefficients

Requirement 2: No pops when reloading filter banks

Requirement 3: Ability to use python for computing filter coefficients

2.5 Block Diagrams

Figure 2 shows a block diagram of our entire system. The main components to look at here are the microphone array and the audio processing subsystem. Samples are fed from the microphones, through the FIR banks in the FPGA, and back to the headphones for the user to listen to. This basic platform has many uses, but for the sake of this class we will be showing the selective listening demo.

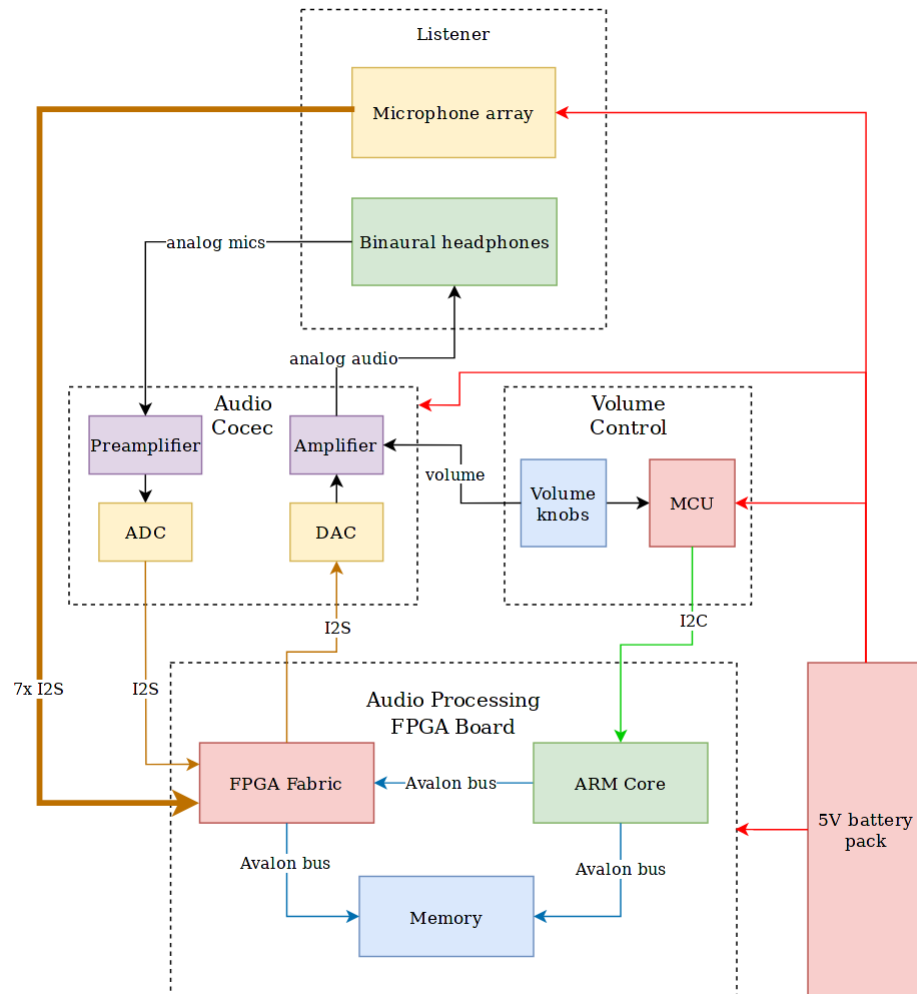


Figure 2: Block diagram of full listening platform

Figure 3 shows a block diagram of the basic DSP flow we described for the first block diagram. There isn't too much to look at, and most of the interesting signal processing work happens on the ARM core.

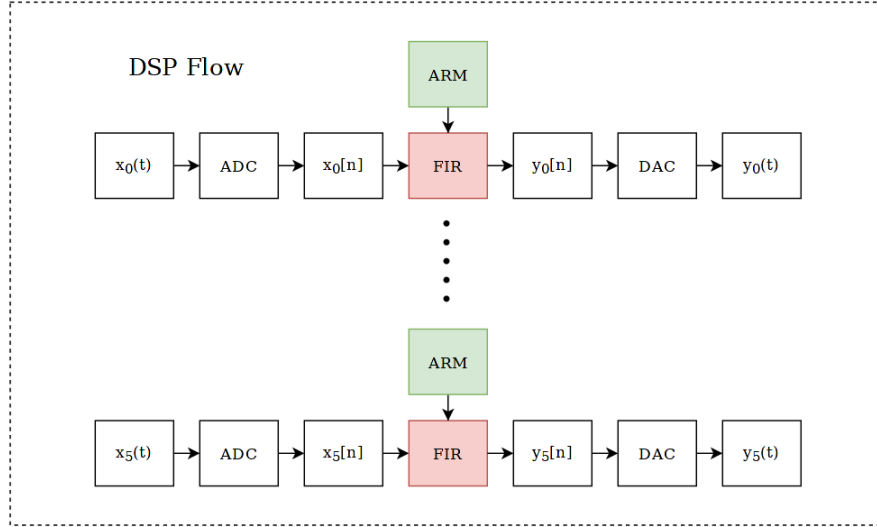


Figure 3: DSP flow diagram of samples through the platform

Figure 4 shows the software flow that runs on the ARM core. This is specific to the selective listening demo we are planning, and will likely look slightly different when testing other algorithms on the platform.

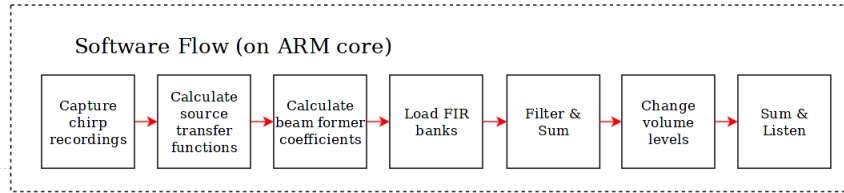


Figure 4: High level overview of the ARM beam forming program

2.6 Risk Analysis

The software running on the ARM core presents the greatest risk to a successful project. The digital signal processing algorithm derived from the problem presented is exceptionally unique. Ryan Corey, a PhD at the University of Illinois, devised this algorithm to improve hearing aid performance in specific listening directions. In order to solve the traditional “noisy cocktail problem,” a

beamforming algorithm is necessary in order to isolate one or multiple listening sources. Computational speed, accuracy, and mathematical complexity pose the greatest challenges to a successful demonstration.

The mathematical rigor of the algorithm is a concern for our group. We do not yet guarantee that the algorithm to compute the beamforming coefficients will finish in enough time to prevent noticeable delay. We are confident that the processing can be done on time on a PC, but on the FPGA hard processor it is still unknown. Though we fully understand the mathematical computations involved and the steps required to generate said coefficients, it remains to be seen whether the ARM core can run this algorithm in time to ensure a smooth listening experience. In numerical terms, the ARM core must generate coefficients within 50 milliseconds in order for the listener to expect the system to behave in “real time.”

3 Ethics and Safety

Our project could present multiple safety hazards. One such hazard is our power system. We are using a rechargeable battery pack that could cause a fire hazard if not cared for properly. If the power system experiences any physical damage there poses a risk of short circuiting the device, or overcharging the device [4]. If the device is short circuited or overcharged it could overheat and damage the board or even pose a fire hazard. Before plugging our battery pack into our system (i.e. the FPGA) we will test the battery pack to ensure it doesn’t exceed 6V. Another risk that could pose a safety hazard to users of our platform is hearing damage. The users of our device will be wearing binaural headphones, with microphones located close to the ear canal. We will ensure that listeners using our product will not experience noises exceeding 85 dB [5]. One key activity that contributes to hearing loss is listening to sounds through headphones that are too loud [6]. Our team will take measures to eliminate the risk of hearing damage while using our platform.

As a team, we fully understand the importance of upholding the IEEE Code of Ethics and we take it very seriously [7]. We strive to develop our platform to be constructed sustainably, and to protect our users privacy in accordance with IEEE Code of Ethics 1. Our project, beyond ECE 445, will be released as an open source platform for hearing aid researchers and manufacturers to take advantage of to improve accessibility of hearing aid technology to benefit all communities. No user’s audio data will be collected on our system in accordance with IEEE Code of Ethics 5 and 6. We will take every precaution to uphold the IEEE Code of Ethics as we develop and release our platform to the public to benefit people who struggle with hearing impairments. The users of our platform will face no discrimination of any form in accordance with IEEE Code of Ethics 7.

Our mission as a team is to make the most accessible and safe open-source hearing aid technology to researchers and hobbyists worldwide. We strongly believe that our system can contribute significant welfare to the public in a

variety of life-changing applications people use everyday.

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