ECE 445

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Design Review

2/27/13

Voice tracing video camera designed for meeting recording

Team #42

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1. Introduction

1.1 Statement of Purpose

Recording an important meeting by video frequently happens in workplaces. Under usual circumstances, a person will be hired to record the entire process of the meeting, where issues of cost, convenience and security arise. Thus we will design a system that would automatically record the meeting in a medium sized room by tracing the voice of the speakers.

Furthermore, we believe during a meeting, arguments, discusses and other forms of information exchange between different speakers are worth capturing. Thus we will provide a mechanism that automatically locates and records the two most important speakers in the ongoing conversation.

1.2 Objective

a. Goals:

For this project we want to implement a voice tracking camera that can select and record up to two active speakers at any time during meetings. The product should improve the quality of meeting recording by being able to capture two speakers at the same time when necessary; it should be highly automated so not much manual adjusting is needed; it should produce high quality recording in a moderate size meeting room. Moreover, the product should be affordable and should be at an acceptable size.

b. Functions / Features:

- High-quality video recording
- Each camera is able to rotate to the 24 different directions.
- The two cameras can capture two speakers at a given time.
- Two camera work independently (i.e. no interference).
- Auto-generate splits- screens when both camera are shooting
- Smartly choose the two most important speakers when multiple people participating the conversation.

c. Benefits

- Recording meeting process without cameraman and at a lower cost.
- Provide a solution to best capture information exchanges (e.g. Q&A, argument or discuss) between two people during the meeting.
- Auto-generated split-screen simplifies post production.

d. Structure Overview

Cameras would be placed on a 360 degrees rotational base that could be placed at the center of the table. The base is surrounded by an array of three electret microphones located on the vertices of an equilateral triangle. From which direction the sound is coming from can be decided by analyzing the phase shift and delay of the signals of the three microphones (note: the calculation is done by the MCU). The 2 main speakers, if any, of the ongoing conversation can be determined. A micro-controller will control a motor to rotate the camera to face to the speaker. There would be two cameras working together and they can be rotated to different angles and record two scenes if two people are talking (e.g. one asking question & one answering). Then the information would be sent to a computer, through a USB interface, to automatically generate split-screens. Otherwise only one camera will be recording and split-screens will not be generated.

2. Design

2.1 Block Diagram

1) Function-units data flow

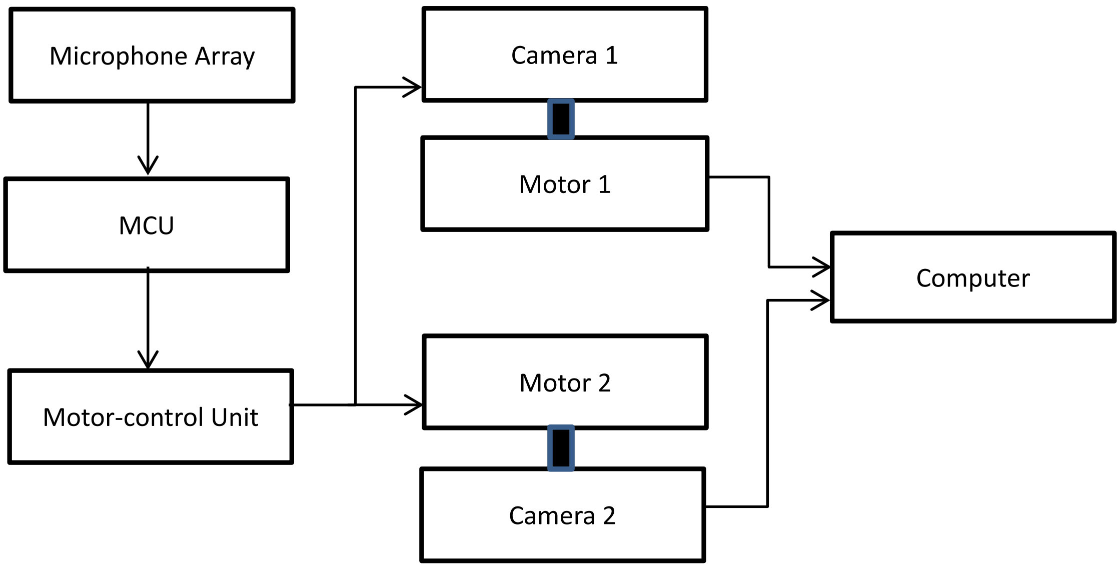


Figure1. Data flow diagram

2) power supply diagram

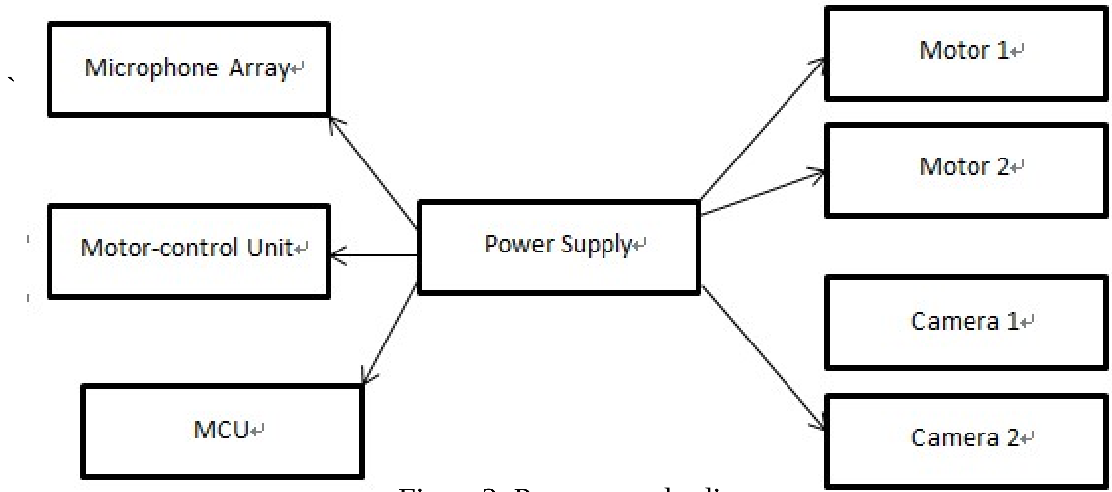


Figure2. Power supply diagram

2.2 Block Description and Design Methodology

1) Microphone Array

This module consists of 3 microphones located on the vertices of an equilateral triangle. The information of the Microphone Array would be collected and analyzed by the Micro-Controller Unit (the MCU module) to make the decision of the direction of the speaker. We choose omnidirectional microphones so that they can receive sound information from each direction equally. The signals would need to be amplified because the signals directly coming from the microphones are too small to be processed by the MCU. They would also need to be filtered to limit the frequency within the range of human voice. In this way, we can eliminate unnecessary noise in the environment.

We have come up with two methods to amplify and filter the signals, one using transistors and one using op-amps. We will test both circuits and decide which one we would use in our final design depending on the performances we get from each.

1st method:

Since the voice of people is around 1 kHz, we add a 0.47 capacitor at the output terminal of MIC and C72 (showed in below picture) to respectively control the passage of low frequency and high frequency.

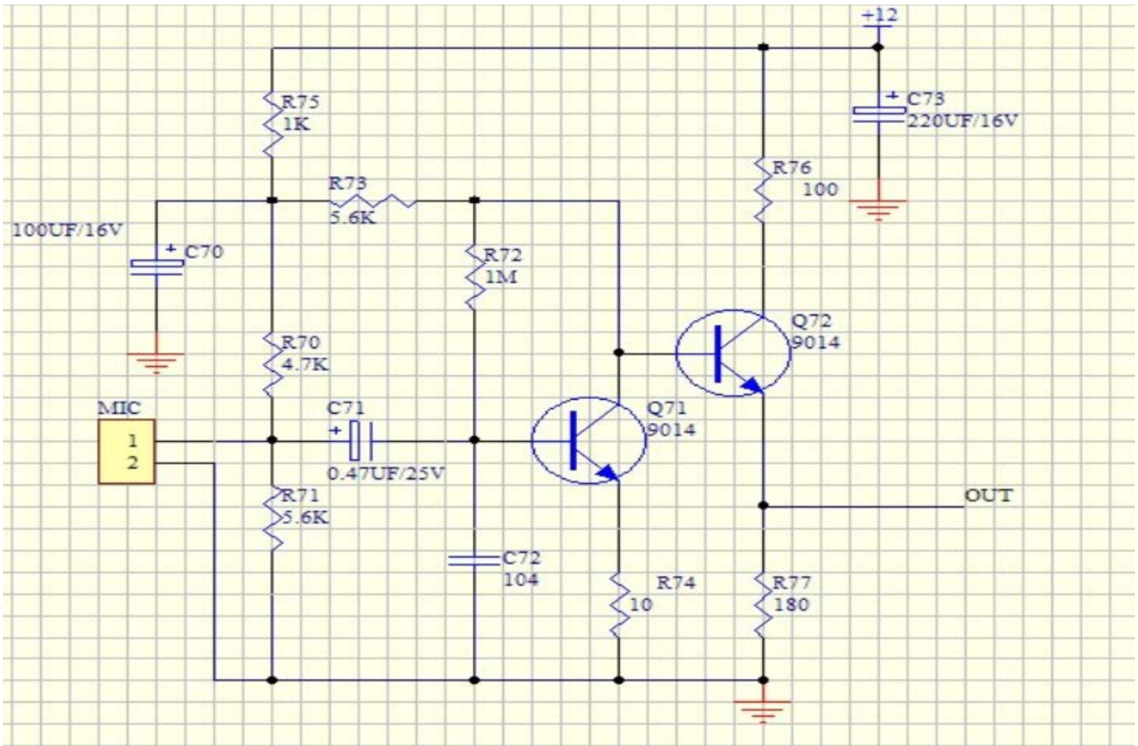
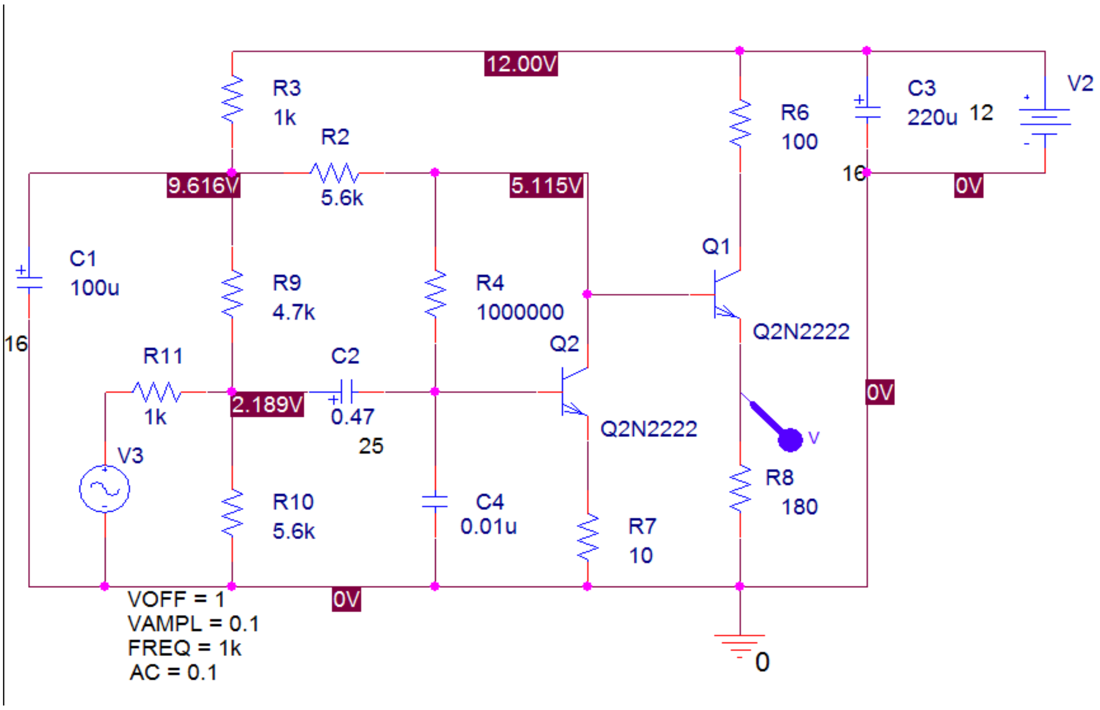


Figure3. Amplify & BPF circuit (1^{st} method)

We adjust gain of the amplifier by changing the value of R70 and R71. 0.47F capacitor at the output terminal of MIC and C72 are respectively responsible for low band pass filter and high band pass filter. We decrease the value of C72 to restrict high frequency and increase the 0.47F capacitor to restrict low frequency.

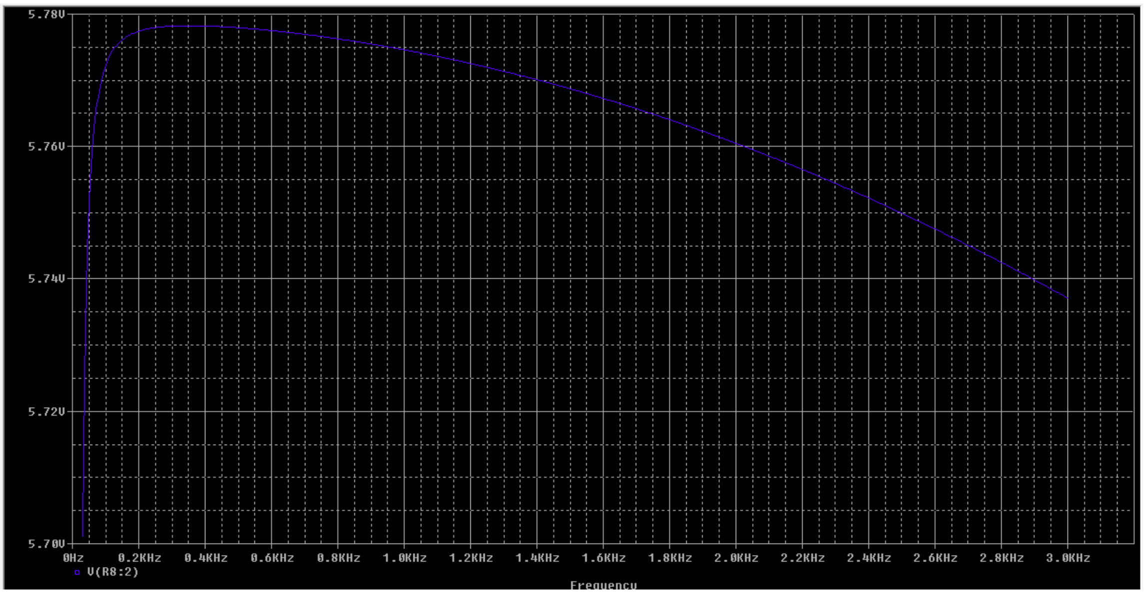
Simulation of amp & filter circuit (method 1) using PSPICE:

We simulate the circuit using transistors with PSPICE. The circuit diagram is as follows.



We could not find the transistor 9014 in the library. So we use Q2N2222 to replace it. The results are quite satisfactory. We use a sine wave voltage source to replace the microphone input. To compensate for the input resistance of the microphone, we add a 1k Ohm resistor in series with the voltage source. For the simulation type, we use AC sweep. The frequency range is from 0Hz to 3k Hz, which is approximately the frequency range of human voice.

The simulation result is as follows. We can see that the simulation has a very nice cut off at 3k Hz.



2nd Method:

We mainly use the op-amp 358 to amplify the signal since it is generally better for human voice amplification.

C2 and C1 are respectively responsible for low band pass filter and high band pass filter. We decrease the value of C1 to restrict high frequency and increase the C2 to restrict low frequency.

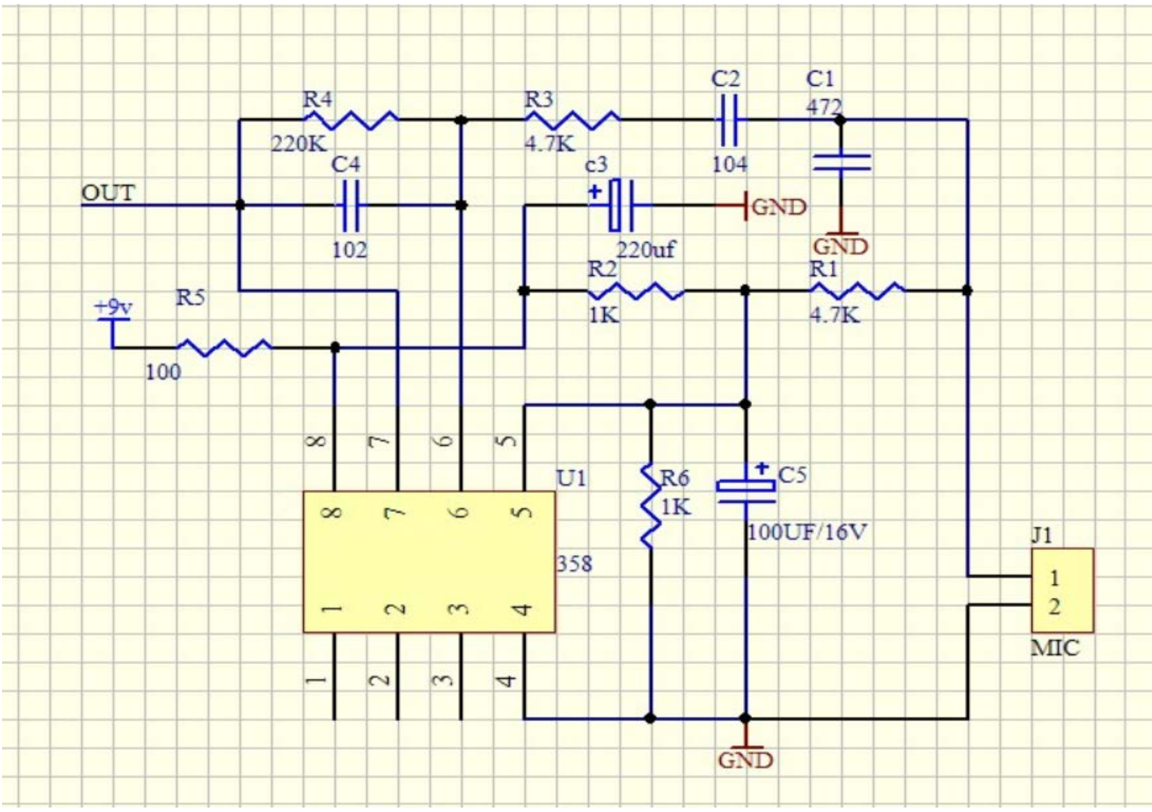


Figure 4 Amplify & BPF circuit (2nd method)

2) MCU (Arduino ATMega328)

This module is the central control unit of the system. It collects data from the Microphone Array, decides in which direction the speaker is located, and controls the Motor-control Unit. It would tell the Motor-control Unit how to direct the motor movement. A learning algorithm will be running on the MCU to determine the main speakers (for this project, up to 2 main speaker will be located). In the following we described the voice source detection algorithm we will use. The Calculation section describes the detailed algorithm runs on the MCU

We will use Arduino ATMega328 board to perform data sampling, cross correlation and the two speaker tracking algorithm discussed above.

The Arduino ATMega328 board has 6 analog inputs, which are sufficient for our implementation (we need only 3 of them). We can use the AnalogRead() function to perform analog to digital conversion.

The MCU will take 3 inputs from the microphone array and will output 5 signals to the motor controller.

MCU (ATMega328) pin mapping:

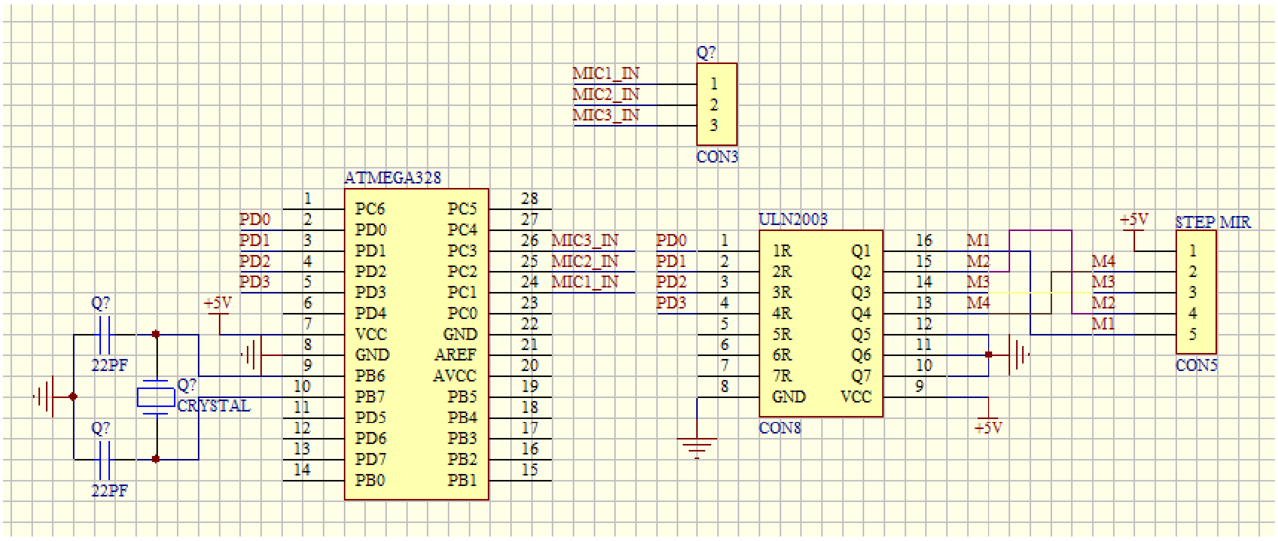


Figure5 MCU (ATMega328) pin mapping

3) Motor Driver Unit

This module receives data from the MCU module and controls the two motors to move to a certain direction so that the camera would be directed at the speaker.

We use the chip ULN2003 to drive the motor. It gets pulse inputs from the Micro Controller and amplifies the signals. It will output voltages high enough to drive the stepper motor. The pull-up resistors are used to ensure the inputs to the chip settle at expected logic levels when high-impedance from the Micro Controller is introduced.

4) Stepper Motor Unit:

This module would be controlled by the Motor-control Unit. Both motors can rotate 360 degrees and in both clockwise and counterclockwise as needed. They will individually carry a camera (Camera 1 and Camera 2) to direct in an appropriate direction to record the speaker.

We choose the stepper motor 28BYJ48 that is 5 lines and 4 phases. It is a gear motor of 1/64. The stepper motor will get inputs from the ULN2003 chip and rotate accordingly. The two motors are controlled independently from two Motor Drivers. The two motors will also individually carry a camera and control its rotational movement in order to track the voice source. The motors can rotate clockwise or counterclockwise according to the input pulses as specified below.

Line	1	2	3	4	5	6	7	8
5	+	+	+	+	+	+	+	+
4	-	-						-
3		-	-	-				
2				-	-	-		
1						-	-	-

If the pulse goes as 1 2 3 4 5 6 7 8, it will rotate in one direction; if it goes like 8 7 6 5 4 3 2 1, it will rotate in the opposite direction.
The circuit diagram and pin specification are as follows. (Both for the motor driver unit and motor unit)

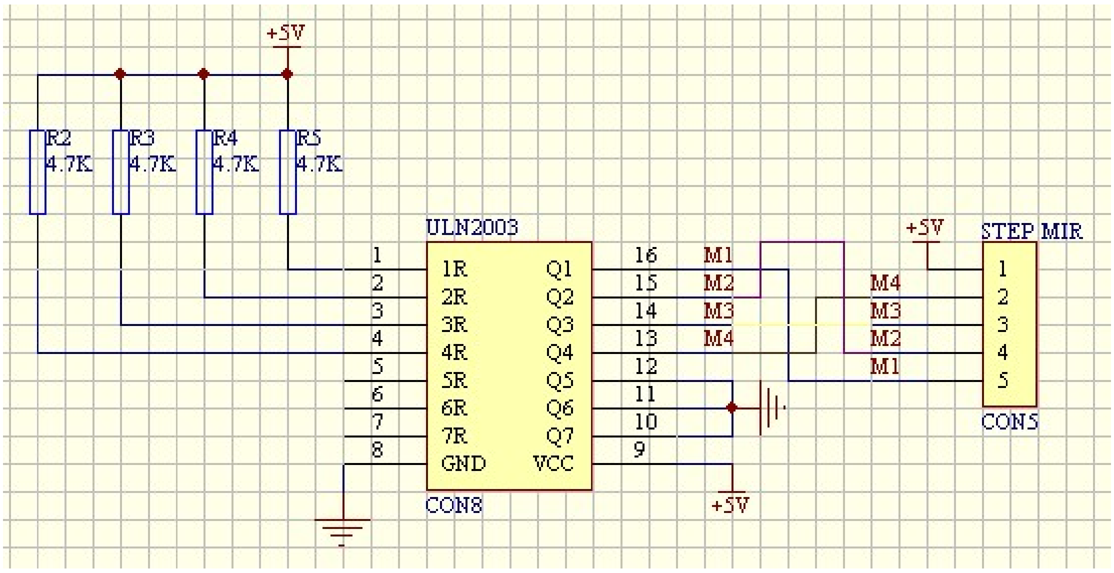


Figure6. Motor Controller pin mapping

5) Camera 1/Camera 2

This module is connected with the motors. The direction that cameras are directed is decided by the rotation of the motors. Cameras will keep recording information and send data to a computer for processing. We plan to use Webcam C210 from Logitech which is a cheap hd webcam and is relatively small compare with other similar products.

6) Computer

The recording data is sent to the computer through USB interface. If two speakers are identified, it will generate a split screen automatically. If not, it will show the image of the camera that is directed at a speaker and ignore the one that is not.

7) Power Supply

This module handles the power supply of the Microphone Array module, the MCU module, the Motor-control Unit module, the motors and the cameras. The microphone would need 1.5V DC. The Aduino Uno, ULN 2003 chip, and the stepper motors will all be operated at 5V DC.

We will use an AC-DC converter to transform the power from a standard wall socket to 5V DC. And then a switching regulator circuit will step down the 5V to 1.5V in order to supply power to the microphone arrays. We will need three such regulators.

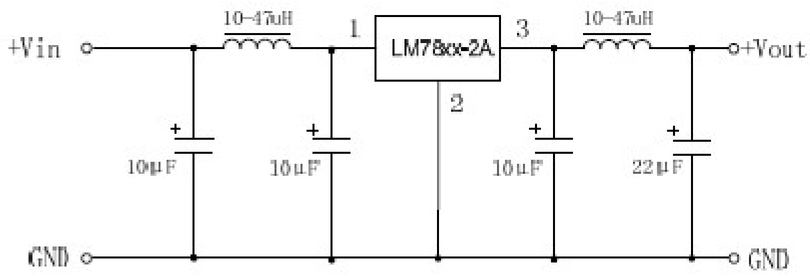


Figure7 Power supply circuit [3]

2.3 Voice source detection algorithm

a. Microphone array geometry

The geometry we use is an equilateral triangle with 1 microphone on each vertex. For each pair of microphones, the distance between them is 30cm.

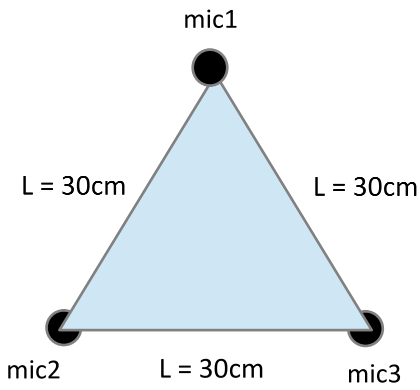


Figure8. Microphone array geometry

b. Basic Voice Direction Detection Algorithm

We adopted the voice source detection algorithm from Knudsen & Lessans' project paper [1]. The 2D-space is equally divided into 6 segments (figure 5.1), each with 60 degree angle centering at the midpoint of the triangular.

Let sig1, sig2, sig3 denote the voice signals received at mic1, mic2 and mic3 respectively.

An example will illustrate how to roughly determine the target angle of camera rotation: Say sig1 and sig2 has smallest phase difference, the voice source is in the blue region in the graph since the blue region is most perpendicular to the line connecting mic1 and mic2. Then, say sig3 lags behind both sig1 and sig2, and the source must be in the top blue section. If sig1 leads sig2, then the source is in the top-left blue segment; otherwise, the source is in the top-right blue segment. The maximum error is limited to 30-degree in this case. For detailed analysis of the algorithm, please reference the Knudsen & Lessans' paper.

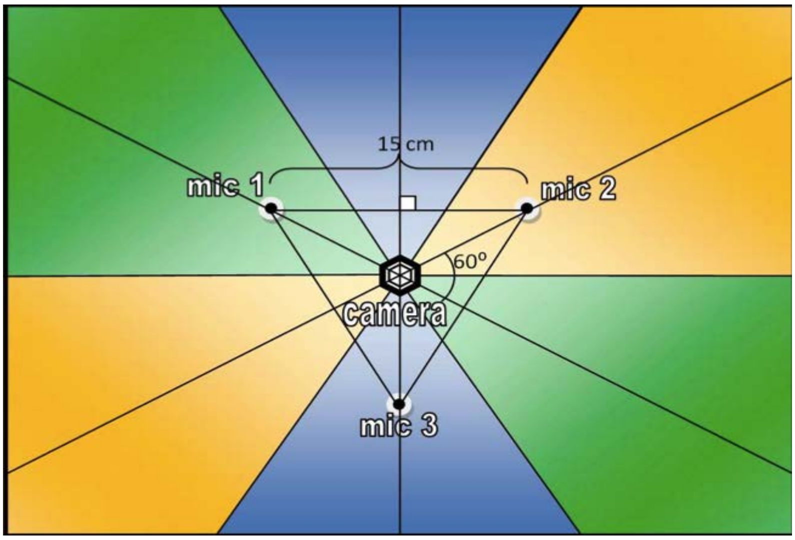


Figure9. Separation of space [2]

c. Further modification to improve resolution

For simplicity, we decided to use discrete values for camera targeting direction. Based on the value of phase difference, we can manually divide each of the 6 segments into even smaller sub-segments. In the below figure, all 24 red arrows indicates all 24 possible camera target direction. We choose to use discrete values for target direction based on the following observation:

1. easy to implement motor control unit using a step motor.
2. easy to implement 2 source tracking (will discuss later)
3. Most web cam would give at least 30 degree of view. A resolution of 24 is sufficient to ensure the quality of the video (i.e. speaker is in the camera frame, but not necessarily centered).

How to divide each region into smaller sub-regions will be discussed later when we introduce the cross correlation and “m value”.

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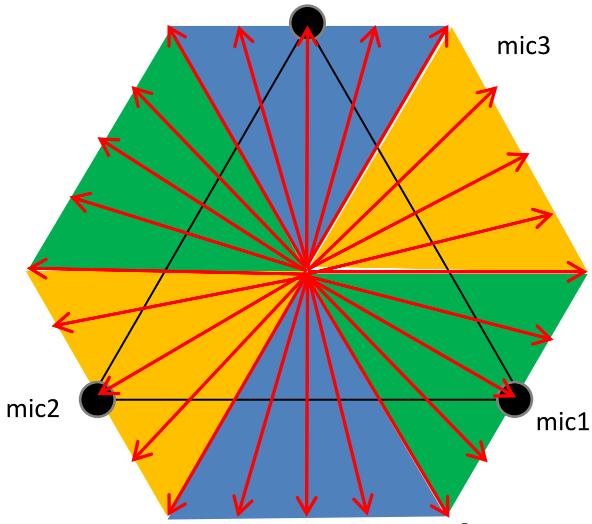

Figure10. Discrete targeting angles

d. Phase difference calculation

The relative phase between each pair of mic-signal is estimated using cross correlation. Let $S_1(n)$ and $S_2(n-m)$ denotes the sample of sig1 and shifted sig2 (shift to the left by m samples) at some sample frequency f.

$$C_{1,2}^{m} = \frac{1}{N}\sum_{n=0}^{N-1} S_1(n) * S_2(n-m)$$

C value indicates the similarity between the two waves. In this case, a large C value for some m indicates a high similarity between sig1 and sig2 shifted by m. Thus, we can estimate the delay between sig1 and sig2 by finding the m which produces largest C. The range of possible m and sample frequency f can be calculated as follow:

Distance between microphone pair: L = 0.3 m
Speed of sound: v = 342 m/s
Maximum delay: T = L/v = 0.00087s
Speech frequency range: usually between 100 – 3000 Hz
Nyquist Rate: f = 6000Hz Sample interval: t = 1/f = 0.00017s

M = T / t = 0.00087/0.00017 = 5; that is, the phase difference between a pair of microphone signals are 5 samples (can be either leading or lagging). Thus m∈[-5, 5], f = 6000Hz.

Here is an example of data flow that determines the phase difference between signals at mic1 and mic2:

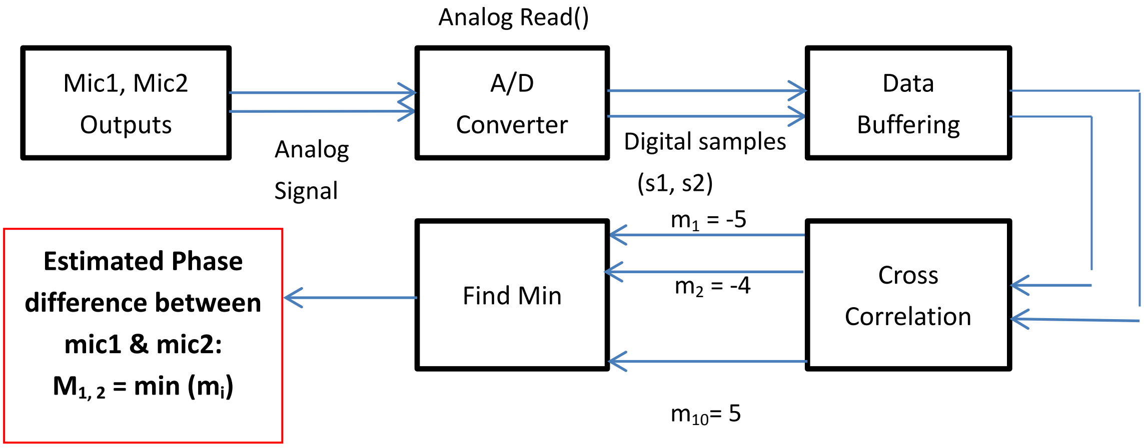

Figure11. Data flow to determine phase difference

We will be sampling at 6000Hz, and uses every 3000 samples as one group (i.e. voice source direction is updated every 0.5s).

If we plot the C value against different m values for a mic pair, a single peak would normally suggest a single voice source, or 2 sources but close enough so that they merges into one peak. A threshold is applied to the C value so that the peak produced by background noise when no people are speaking will not be considered as voice sources.

The threshold value needs to be determined and verified by manually testing in the future to insure the background noise to be discarded.

Multiple peaks on the C-m graph suggest multiple speakers speaking within the same time interval. We realize it is hardly possible to separate these voice sources only using the current available information. So for the 0.5s interval in which there are multiple peaks (all above threshold) in the C-m plot, the plot will not be used to estimate voice source location.

We will get 3 phase differences for 3 microphone pairs (i.e. M_{12}, M_{23}, and M_{31}). Then we can use Knudsen & Lessans' algorithm to roughly locate the voice source in a colored region. Again assume the source is in top blue region of figure1.3. Then we look at the value of M_{12}:

M_{12}	Camera Direction (figure1.5)
-5, -4	A
-3, -2	B
0, -1, 1	C
2, 3	D
4, 5	E

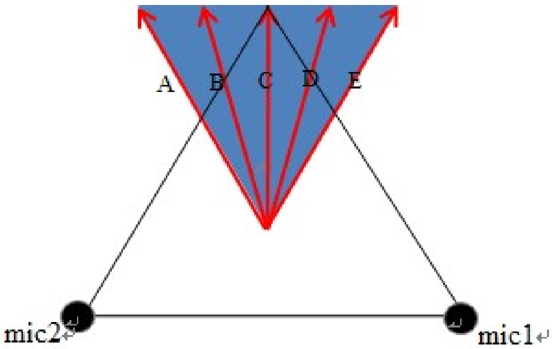

Figure12.

Note: The accuracy of this estimation need to be verified in the future testing.

e. Two speakers tracking:

The idea is to separate different speakers sitting at different directions and find how active each speaker is during the on-going conversation; select the two most active speakers if there are any, and face the camera towards them.

As discussed above, we split the 360 degree into 24 discrete targeting angles, and index them through 1 to 24. An array (let's call it Y-heap) of 24 elements will be initialized to all 0s.

For each 0.5 seconds, the voice detection algorithm will produce a number from 1 to 24 that indicates the current speaker (or 0 if silence). The corresponding array element will increase its value using the forgetting function:

$$Y_{new} = a*Y_{old} + (1-a)*C \qquad \text{Where C is a constant}$$

All other element will be updated using:

$$Y_{new} = a*Y_{old}$$

The weight of old record will exponentially decrease. The 'a' value needs to be carefully picked so that the Y value can roughly reflect the level of participation of each direction in the on-going conversation.

Each time, we can select the 2 elements with largest Y value (which also need to be greater than some threshold) and have the 2 cameras rotated to the corresponding angles.

State Diagram for MCU

Idx_m1: the index with largest Y

Idx_m2: the index with second large Y

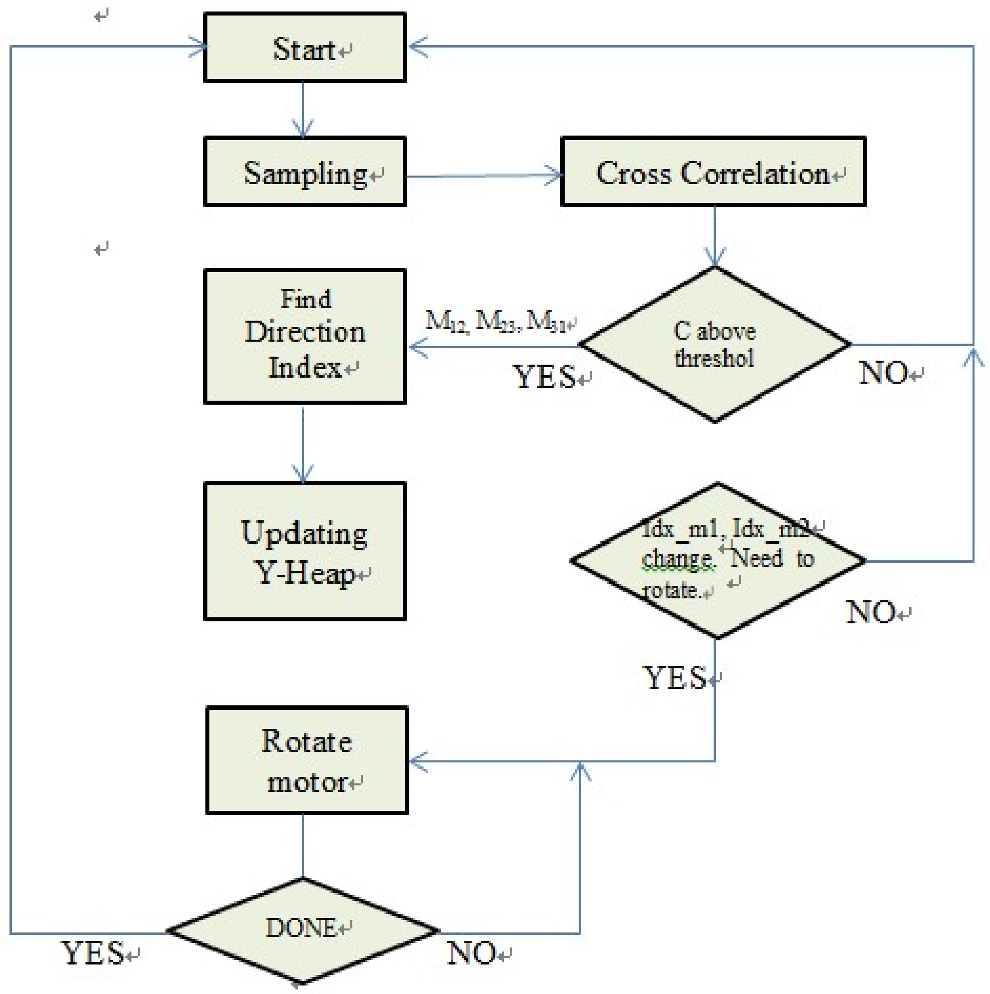

Figure13. MCU State Diagram

3. Requirements and Verification

3.1 Testing procedures

Microphone Unit

Requirements	Verification
(1) The microphones are able to accurately cross-correlate human voice while ignoring the background noise. a. Microphone test with supplied with 1.5V can pick up voltage signal around $\pm$50 mV when speaker is at 10 feet away. b. The 3dB cut off frequency of high pass filter should be 100Hz and the cut off frequency of low pass filter is close to 3KHZ.	(1) a. Connect output of a microphone to oscilloscope. The waveform of the oscilloscope should be strong and clear when the microphone is at 10 feet away from a speaker. b. Connect the output of function generate (100Hz or 3KHZ 5V) to the input of the circuit, then observe the amplitude of waveform on oscilloscope showed the output of the amplifier circuit should be close to 5$\times 0.707 = 3.537$V.
(2) The output (amplified) waveform should be clear and strong when the speaker is10 feet away from microphones. Amplifier is able to expand 50 mV to 2 V. a. The gain of amplifier is around 40.	(2) Connect the output of function generate (50 mv) to the input of the above amplifier circuit., then Observe the amplitude of waveform on oscilloscope showed the output of the amplifier circuit should be 2v.
(3) The range output voltage of microphone circuit is required from 0 to 5V.	(3) Add a 2.5 DC source in series, then obtain the range of voltage waveform on oscilloscope is from 0 to 5V. (4) Connect the 2 outputs to the 2 channels of the oscilloscope and read the

(4) When the distance between two microphones is 30cm, the phase delay close to 0.88ms (± 0.5ms) (.30m/340ms^{-1}).	delay of the 2 waveforms.

Voice Source Detection / MCU

Requirement	Verification
(1) The data sample rate should be close to 6000Hz. So all the calculation holds. 2) The C peaks produced by the background noise should be discarded. 3) When multiple C peaks appear in C-m plot, the frame (data within 0.5s interval) should be discarded. 4) The direction index calculated should be accurate enough so that the speaker can be captured in frame. 5) The forgetting coefficient 'a' should be properly chosen so that neither the camera frequently change target nor the camera seldom change target. 6) Each digital output pin of the MCU can give a digital pulse of magnitude 5$\pm$0.5V that will be used drive the motor in both directions. The pulse width will be 5ms ($\pm 1ms$) 7) MCU can generate sequence of pulses as described in Chart1. The length of each sequence should be at most 50ms.	1) Count the number of samples over some time interval and find average sampling frequency. The frequency should be roughly around 6000Hz(+-200hz). 2) Generate some background noise; Find the minimum noise level that creates a peak above the threshold. The min noise level should be considerably large (30 - 35dB). We can use speakers to generate this noise and by adjusting the sound level to modify noise level. 3) Print out the C-m plot, and mark which frame was used and all frames should only have 1 peak that is above the threshold 4) Create some random M values, and check whether the algorithm produces correct direction indices based on each input set. 5) The coefficient value determines how reactive the systems will be. We will test out a range of values (range from 0 to 1) and see which one serves us the best. 6) Hook up the output of MCU to an oscilloscope and measure the width and magnitude of the pulse. The test should be done on all 4 output pins. 7) Connect 4 output pins to 4 LEDs. Issue 100 rotate instructions, and observe

	the pulse sequence on LEDs. Should see LEDs flashing with the correct pattern described in Chart1. And the LEDs should flash no longer than 5 seconds. Test should be done for both clockwise and counter-clockwise rotation. .

Motor Driver Unit:

Requirement	Verification
(1) It can control the motor to rotate by a certain number of steps. (2) It can control the motor to rotate both clockwise and counterclockwise. (3) When it takes input from MCU, it should output current high enough to drive the motor. Specifically, it should output current at 500 mA (+/-30 mA).	(1) Write some codes to generate input for the driver. Connect the MCU output to the four inputs of the pin 1, 2, 3, 4 of the motor driver. If it goes through 64*8 cycles, the motor should experience exactly one revolution. (2) Set the codes to go two directions of the cycle described above and see whether the motor rotate in opposite direction. (3) Connect the MCU to the chip. Use the millimeter to measure the current output to make sure it stays within a certain range.

Stepper Motor Unit:

Requirement	Verification
(1) Motor should rotate exactly the steps it is instructed to by the motor driver. Specifically, to meet our design requirements, we want the motor to rotate all multiples of 15 degrees with zero error tolerance. (2) Motor should rotate in the direction as it is instructed to.	(1) Write code to the MCU to instruct the motor to rotate all multiples of 15 degrees, namely, 15, 30, 45, 60, 75, 90, 105, 120, 135, 150, 165, 180, 195, 210, 225, 240, 255, 270, 285, 300, 315, 330, 345, 360. (4) Write the code to instruct the motor to rotate clockwise or counterclockwise. See if it rotates in the direction correctly. See detailed instruction to make the motor rotate in both directions in the section of stepper motor unit.

3.2 Tolerance Analyses

The tolerance analysis mainly focuses on:

-The error of voice source detection caused by the geometry of the microphone array and the factor of noise. The goal is to have the difference between actual source and calculated source to be within 5% (i.e. within 20 degrees), so that the camera will still be able capture the speaker in the frame if not centered in the frame. We will test the tolerance by setting up all possible scenarios:

1. Only one speaker
2. Two speakers talking simultaneously / separately.
2. Two speakers sitting close to each other / away from each other.
3. Two speakers with some noise (machine noise or noise from other people).
4. Multiple speakers.

-The error of camera rotation:

We will analyze the cause of error of camera rotation if there is any. Again, our goal is to limit the error within 5% (less than 20 degrees) for the same reason stated above. We will measure the tolerance by feeding the motor control with pre-calculated values so we know where the camera should be pointing at after the rotation.

3.3 Ethical Issues

3) To be honest and realistic in stating claims or estimates based on available data.

We will be true and honest to ourselves and present results and data of our experiment as they are originally without any modification or cheating. We will also ensure to use our own result and don't plagiarize from others.

5) To improve the understanding of technology, its appropriate application, and potential consequences.

We will explore the most versatile possible version of our product and try to optimize its performance to serve the customers most.

7) To seek, accept, and offer honest criticism of technical work, to acknowledge and correct errors, and to credit properly the contributions of others.

We will acknowledge the portion that we take reference from others. And we will be responsible about the possible errors we make.

3.4 Safety

Our product contains potentially harmful voltage levels. So we will carefully isolate the inner circuits by shielding them with proper shells. Since this product will always be used inside the room, we do not need to consider issues such as rain and extreme temperature changes.

When building our product, we should also be aware of the lab safety procedures and avoid doing anything that we are not sure about its safety issues. Testing and debugging should also call for carefulness in terms of the possible high voltage source we will use.

4 Costs and Schedule

4.1 Cost Analysis:

Parts

Tom-3050L-R	3	3.71	11.13	microphone
P10408TB-ND	1	0.04488	0.04488	capacitor(100μF/16V)
GCM219R71E474K	2	0.0591	0.1182	capacitor(0.47μF/25V)
P5183-ND	1	0.0646	0.0646	capacitor(220μF/16V)
CG-CT1-104	1	0.0701	0.0701	capacitor(104F)
ERG_1SJ102A	1	0.03585	0.03858	Resistor(1K)
CFR-25JB-1M0	1	0.03312	0.03321	Resistor(1M)
GCM219R71E474K	2	0.04562	0.04562	Resistor(5.6K)
ERG-1SJ472A	1	0.03585	0.03585	Resistor(4.7K)
RN60C1083FB14	1	0.1078	0.01078	Resistor(108K)
EGR-2SJ104A	1	0.04868	0.04868	Resistor(100)
EGR-1DJ100A	1	0.03585	0.03585	Resistor(10)
ESIPDHE3/85A	2	0.13	0.26	Diode 9140
2947190-ND	2	9.85	18.7	PCB

ARDUINO UNO	1	27.83	27.83	Microcontroller
ADC081S021CIMF/NOPB	1	2.46	2.46	A/D Converter
CFR-25JB-2K2	1	0.01104	0.01104	Resistor(2.2K)
MCP4706AOT-E/CH	1	0.62	0.62	D/A Converter
Motor Unit				
28BYJ48	1	8.59	8.59	Motor
ULN2003		0.49	0.49	Motor controller
Camera				
Webcam C210	2	17	34	Mini Camera
total	104.64$			

Labor

Name	Hour Rate	Total Hours Invested	Total
Qi Yang	$35.00	150	$13,125
Yuxiao Lu	$35.00	150	$13,125
Jiehan Yao	$35.00	150	$13,125
Total		450	$39,375

Grand Total

Section	Total
Labor	$39,375
Parts	$104.64
Total	$39,480

4.2 Schedule

Week	Task	Responsibility
2/4	Finalize and Submit proposal	Qi Yang
	Review microphone specs and finalize selection	Jiehan Yao
	Propose voice source detection algorithm	Yuxiao Lu
2/11	Mock Design Review	Jiehan Yao
	Order MCU, microphones, motors	Qi Yang
	Finalize voice source detection algorithm	Yuxiao Lu
2/18	Finalize motor control unit design	Yuxiao Lu
	Design algorithm for voice targets selection	Qi Yang
	Finalize microphone array structure design	Jiehao Yao
2/25	Design Review, programming MCU	Qi Yang
	Testing microphone array functionality	Yuxiao Lu
	Testing motor functionality	Jiehan Yao
3/4	Finish circuit design for microphone array	Yuxiao Lu
	Finish circuit design for motor control unit	Jiehan Yao
	Testing all MCU functionality	Qi Yang
3/11	Finalize PCB design	Yuxiao Lu
	Testing input/output, identify/correct errors	Jiehan Yao
	Finish programming split screen function	Qi Yang
3/18	Order 1st revision PCB fabrication	Qi Yang
	Design/Order 1st revision rotational metal station	Jiehan Yao
	Tolerance measurement	Yuxiao Lu
3/25	Mock up demo	Yuxiao Lu
	Assemble different units on to the station	Qi Yang
	Testing overall functionality with the station	Jiehan Yao
4/1	Order final PCB fabrication	Qi Yang

	Design/Order final rotational metal station	Jiehao Yao
	Verification of specifications	Yuxiao Lu
4/8	Fix remaining bugs	Qi Yang
4/15	Ensure product completion	Qi Yang
	Tolerance analysis	Jiehan Yao
	Verification of specification	Yuxiao Lu
4/22	Finish preparing Demo	Yuxiao Lu
	Finish preparing final presentation	Qi Yang
	Finish preparing paper	Jiehan Yao
4/29	Demo / Checkout	Jiehan Yao
	Presentation	Qi Yang
	Paper	Yuxiao Lu

Reference

[1][2] Voice Tracking camera-Video conference of the future, Knudsen, Lessans, and Scholl, http://ese.wustl.edu/ContentFiles/Research/UndergraduateResearch/CompletedProjects/WebPages/sp11/JasonZachMichael/ESE%20498%20WriteUp%20Zach,%20Jason,%20Michael.pdf.

[3] Image from the datasheet of LM78xx-2A