GLOBAL ACTIVE NOISE CANCELLATION FOR CELL PHONE PRIVACY

Project Proposal

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ECE 445: Senior Design

February 6, 2013

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1.0 INTRODUCTION

1.1 Title

Global Active Noise Cancellation for Cell Phone Privacy

Cell phones have allowed for a great level of connectivity giving us the ability to communicate with each other whenever and wherever we so chose. However, this ability to speak to anyone at any time has essentially put our lives on display for anyone within earshot to hear. This project was chosen because there is currently no product that will provide privacy while speaking on a cell phone in public places. In addition, both project members have previously worked on a project involving digital signal processing and acoustics. As such, it is felt that both group members contain the background knowledge necessary to undertake this problem. It will be very exciting to perform a study on the practicality of using active noise cancellation to solve this problem as well as to apply the DSP knowledge learned in previous courses to a real world application.

1.2 Objectives

The intent of this project is to demonstrate the feasibility of using active noise cancellation to globally silence the voice of a cell phone user during a phone call. Four goals have been established that are critical to the success of this project:

- design microphone amplification and filter circuit
- implement digital signal processing algorithm to offset phase of source by 180°
- characterize spatial pattern of human speech intensity over the frequency band of interest (300 – 3000 Hz)
- evaluate optimum speaker location for best overall noise cancellation

Our finished project has the following intended functions:

- actively cancel out a cell phone user’s voice in far field
- perform cancellation globally in order to provide privacy

Benefits to the customer include:

- provides ability to have private phone calls in public locations
- eliminates disruptive background conversations for general public

Desirable product features include:

- low power consumption
- small package size (ability to be incorporated into current cell phones or cell phone cases)

2.0 DESIGN

2.1 Block Diagram

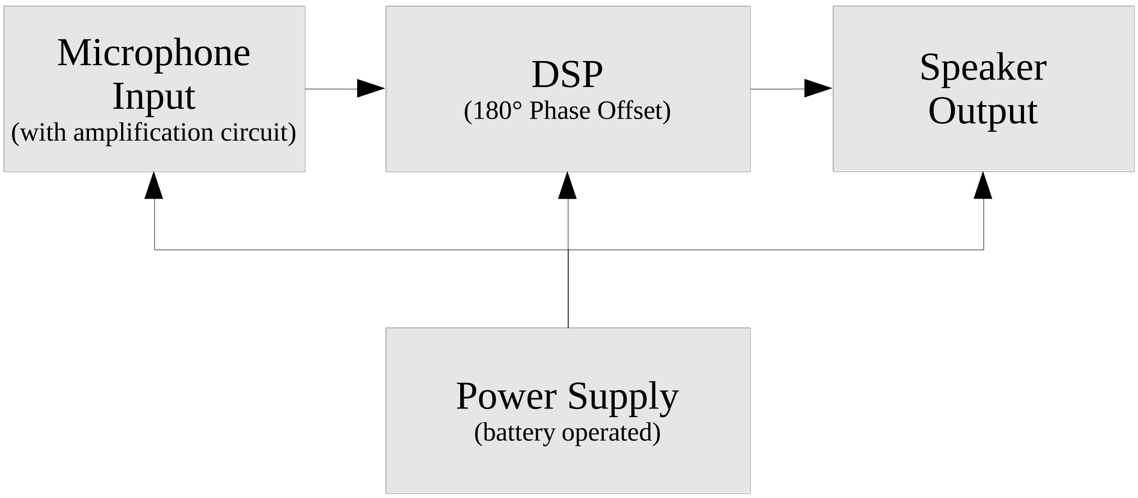

Figure 1. Top level block diagram

2.2 Block Descriptions

Microphone input:

The microphone input will consist of a single microphone along with a filter circuit (low pass below 3 kHz to cut out high frequency noise) and an amplifier circuit if the expected signal levels will be lower than required for input into the DSP chip. The function of the microphone will be to acquire the speech signal of the person using the cell phone, low pass filter this signal, and possibly amplify this signal for processing. The microphone input will provide the DSP chip with the voice signal to be phase shifted for cancellation. The microphone input will interface with both the DSP chip and the power supply. The power supply will provide the bias for the microphone and power to the amplification circuit. The output of this amplification circuit will connect to the analogue to digital converter input of the DSP chip where the signal will be digitized for signal processing. The microphone block is important to the overall design because without it, we would not be able to acquire the input signal. If we cannot read in the initial sound source, it would be impossible for us to cancel it out.

DSP:

The function of the DSP block is to perform the signal processing necessary to create a signal 180 degrees out of phase with the original input signal. This block is important to the overall design because without it, there would be no way for us to shift the initial input signal by 180 degrees. This means there would be no way of providing active noise cancellation. This block has a simple interface with the rest of the design. It will receive its input signal from the microphone block through an analogue to digital converter, it will perform the signal processing, and then it will output the processed signal to the speakers through a digital to analogue converter. In addition, it will receive the power that it needs in order to function from the power supply block.

Speaker output:

The function of the speaker output is to generate the physical sound waves which will be used to actively cancel out the initial sound source. This block is important to the overall design because without it, we would not be able to physically output the processed wave. Therefore, the initial sound source would not actually be canceled. The speaker output block only interfaces with the DSP block and the Power Supply block. It receives its input, the processed sound wave, from the DSP chip through a digital to analogue converter. It receives the power necessary for it to function from the power supply.

The speaker output will consist of multiple speakers, arranged in a pattern such that their output most closely mimics the spatial intensity pattern of the human voice. In order to achieve this, it is first necessary to characterize the spatial intensity pattern of the human voice. The method that will be employed will be to place microphones in various locations in a reverb-minimizing chamber. A person will then speak from a known point within the chamber and data will be simultaneously gathered about the frequency and intensity of the sound waves at the different microphone locations. This data will then be used to characterize the spatial intensity of the human voice and its dependence upon frequency.

Power supply:

The power supply will consist of a battery as well as a voltage regulator circuit that will provide correct voltage levels to the microphone input, DSP chip, and speaker output. The function of the power supply will be to provide the bias for the microphone input as well as the power for the microphone amplification circuit, power for the DSP chip, and power to the speaker and any amplifier circuits that might be necessary to drive the speaker. It interfaces directly with all of the other blocks in order to supply them with power. This block is important to the overall design because without it, our other blocks would not be able to function.

3.0 REQUIREMENTS AND VERIFICATION

3.1 Performance Requirements

Microphone

1. Cannot clip input signal when person's voice does not exceed normal levels (to be determined by voice characterization)
2. Correctly filters out frequencies above 3 kHz
3. Peak output signal (to be fed to DSP) must not exceed 3V for typical speech
4. Cut off frequency must be within ± 10% of chosen cutoff frequency

DSP

1. Must read in correct input signal
2. Performs calculations in real time – output waveform must be within 180° ± 18° out of phase from the original input signal for 10 dB sound cancellation

3. Must correctly output the phase-shifted signal

Speakers

1. Spatial sound pattern must match that of human voice within 6 dB for all points in space
2. Must output audio in range of 300 Hz to 3 kHz

Power Supply

1. Power supplied to each of the other components is within ±10% of the typical rating for each component

3.2 Test Procedures

Microphones

1. Display microphone output on oscilloscope to ensure there is no clipping when speaking into microphone.
2. Feed sinusoids of known frequencies into filtering circuit. Display input waveform and output waveform on oscilloscope. Verify that output waveform is attenuated for input frequencies greater than 3 kHz
3. Use oscilloscope to measure output signal and use peak detection to verify that signal is below 3V
4. Check 3 dB down point of filter using function generator and oscilloscope to verify that it is within ± 10% of chosen cutoff frequency

DSP

1. Feed in sine wave with function generator; tell DSP to output the same signal that is input; display the input and output signals on the oscilloscope and verify they are identical
2. Use DSP development environment to determine clock cycles required for execution of program. Multiply this number by the amount of time to execute each cycle. Make sure this number is less than 33.33 microseconds (since we want an error less than ± 18 degrees, we need the output signal to be sent within one tenth of the period of the input signal – since the maximum frequency we will process is 3 kHz, the minimum period is 333.3 microseconds – one tenth of this number is 33.33 microseconds)
3. Feed DSP a sine wave between 300 Hz and 3 kHz with a function generator; display input and output of DSP on oscilloscope and use the acquire function to verify that they are 180° ± 18° out of phase

Speakers

1. Use microphones to determine speaker output intensity at various locations around the speaker; compare with known pattern for intensity of human voice to ensure similarity; perform this experiment in a reverb-attenuating room to reduce noise from external sources
2. Use function generator to feed input signals between 300 Hz and 3 kHz to speakers and verify that they are output correctly (compare with computer generated audio signals of same frequency)

Power Supply

1. Use multimeter to measure voltage and current delivered by power supply to each component.

3.3 Tolerance Analysis

The goal of the tolerance analysis is to verify the correct behavior of the filter circuit within the microphone input. This will be important for the proper function of our project because the filter will remove high frequency noise, improving our signal to noise ratio as well as limiting aliasing in the frequency analysis of the signal in the DSP chip. We will be designing an RC low pass filter. The correct values of the resistors and capacitors within this circuit must be maintained in order to achieve the correct cutoff frequency. If the cutoff frequency is too low, we will lose important vocal data, while if the cutoff frequency is too high we will not be filtering out as much noise as possible. A tolerance analysis of one capacitor of the filter will be performed in order to find a tolerance range for the capacitance that ensures that the filter's cutoff frequency is within ± 10% of the chosen cutoff frequency. In order to verify that the capacitor has a capacitance value that is within the acceptable tolerance range, we will measure the -3 dB cutoff frequency of the filter using a function generator and an oscilloscope. More specifically we will use the function generator to input signals of known frequency and amplitude into our filtering circuit. We will display the output on the oscilloscope and use the measure function to acquire the amplitude of the output wave. We will find the frequency at which the amplitude of the output wave is 3 dB less than the amplitude of the input wave. We will vary the capacitor values to find the range of capacitances for which the measured -3 dB cutoff frequency is within ± 10% of our chosen cutoff frequency.

4.0 COST AND SCHEDULE

4.1 Cost Analysis

Labor:

Employee	Labor Costs
Joel Godard	\$50/hr * 12 hrs/week * 12 weeks * 2.5 = \$18,000
Hershed Tilak	\$50/hr * 12 hrs/week * 12 weeks * 2.5 = \$18,000
Total	\$36,000.00

Parts:

Item	Quantity	Unit Cost (\$)	Total Cost (\$)
TI TMS320C5409APGE16 (DSP)	1	\$27.30	\$27.30
25J10K (10 kΩ Resistor)	10	\$1.65	\$16.50
MFR100FRF52200K (200 kΩ Resistor)	10	\$1.50	\$15.00
4066PHCT-ND (10 µF Capacitor)	10	\$0.75	\$7.50
4201PHCT-ND (100 µF Capacitor)	10	\$2.71	\$27.10
445-8575-ND (5 pF Capacitor)	10	\$0.25	\$2.50
1C25Z5U223M050B (20 nF Capacitor)	10	\$0.15	\$1.50
WM-61A (Panasonic Microphone)	2	\$1.92	\$3.84
AS01808MR-R (Speaker)	6	\$3.78	\$22.68
LM386N-1 (Op-Amp)	5	\$0.53	\$2.65
1294 (Keystone 9 Volt PCB Mount Holder)	1	\$1.65	\$1.65
PC16049V (Duracell 9V Battery)	2	\$1.46	\$2.92
PCB	4	\$33.00	\$132.00
Total			\$263.14

Grand Total:

Labor	$36,000
Parts	$263.14
Total	**$36, 263.14**

4.2 Schedule

Week	Task	Responsibility
2/10	Design preliminary amplifier circuit	Joel
	Design preliminary low-pass filter circuit	Hershed
2/17	Build and test amplifier circuit	Joel
	Build and test low-pass filter circuit	Hershed
2/24	Design Review	Joel
	Create pseudo-code for DSP algorithm	Hershed
3/3	Collect data on spatial intensity of human voice	Joel
	Begin programming DSP chip	Hershed
3/10	Process data from previous week	Joel
	Finish programming DSP chip	Hershed
3/17	Create interface to connect speakers	Joel
	Designing PCB Board in Eagle	Hershed
3/24	Work out preliminary speaker placement configurations	Joel
	Solder PCB / revise and resubmit if needed	Hershed
3/31	Perform experiments with speaker placement (early in week)	Joel
	Analyze data from speaker placement experiments (later in week)	Hershed
4/7	Finalize speaker placement	Joel
	Tolerance Analysis	Hershed
4/14	Create final speaker mounting assembly	Joel
	Prepare Demo	Hershed
4/21	Demo	Joel
	Prepare Presentation	Hershed
4/28	Final Paper	Joel
	Presentation	Hershed