Global Active Noise Cancellation for Cell Phone Privacy

Final Presentation

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ECE 445: Senior Design

Project #27

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Agenda

- Introduction
- Objectives
- Original design and fabrication
- Modifications to Original Design
- Requirements and verification
- Voice Characterization
- Speaker Array Evaluation
- Further testing
- Summary and conclusions
- Recommendations for future work
- Credits
- Questions

Introduction

- Active noise cancellation (ANC) is currently used in many noise cancelling headsets
- Wanted to explore active noise cancellation to reduce cell phone users voice in the far-field
- Requires cancelling noise at source instead of at destination

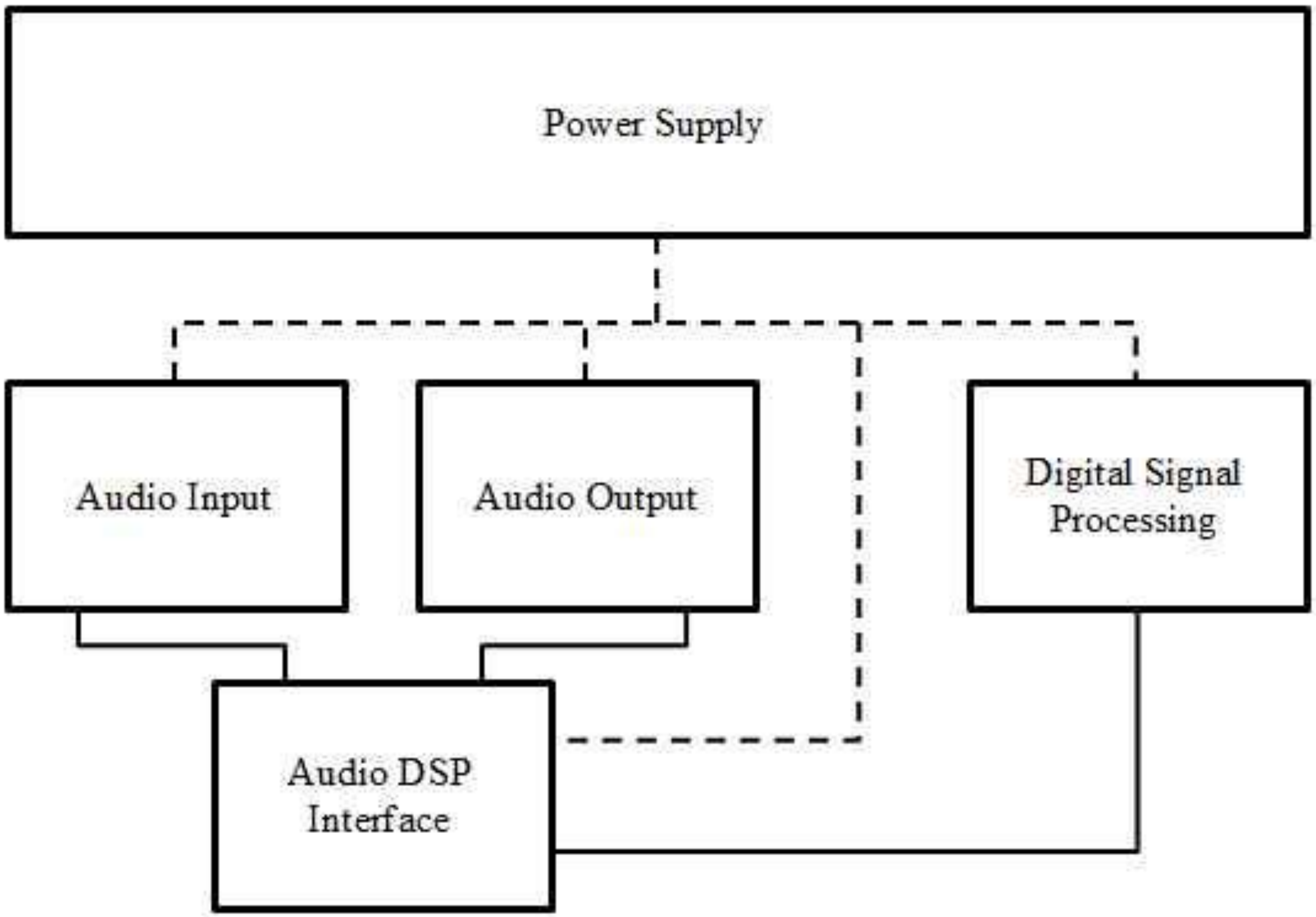
Objectives

- Explore active noise cancellation to reduce cell phone users voice in the far-field
- Characterize human voice radiation pattern
- Assess optimal speaker arrangements to maximize destructive interference

Original Design

- Acquire raw voice signal through microphone, amplifier, and filter circuitry
- Phase shift by 180° using DSP
- Emit phase shifted signal through speakers
- Characterize speaker and human voice radiation patterns to aid in speaker array design

Original Design: Top Level



Original Design: Audio Input

- Acquires and amplifies input waveform
- Mic has flat response from 100Hz to 15 kHz
- Amplifier gain of 5 V/V

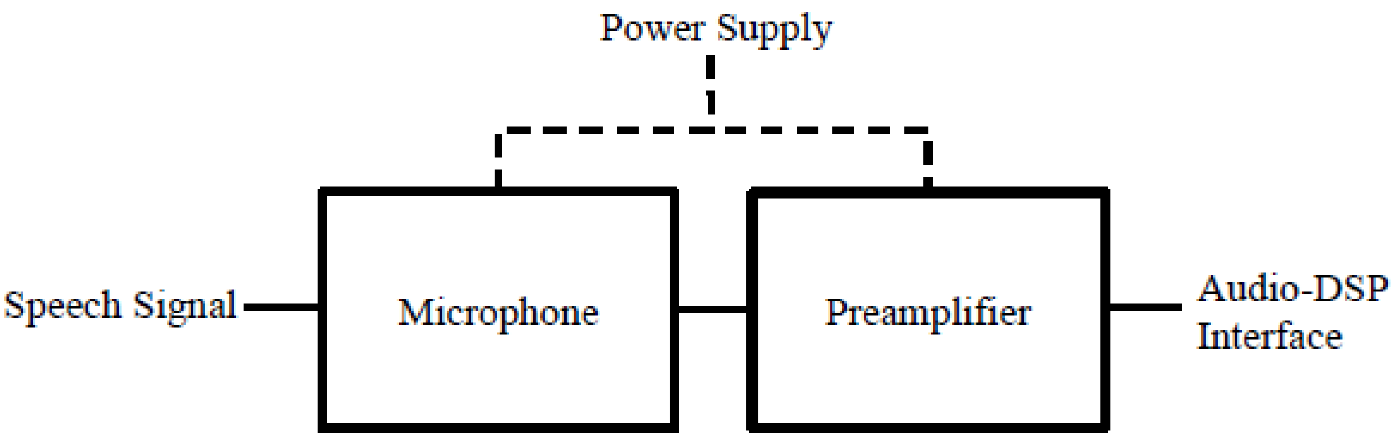
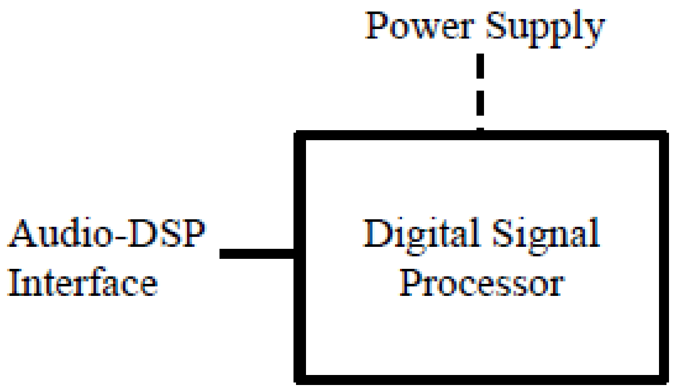


Figure 2: Audio Input block diagram

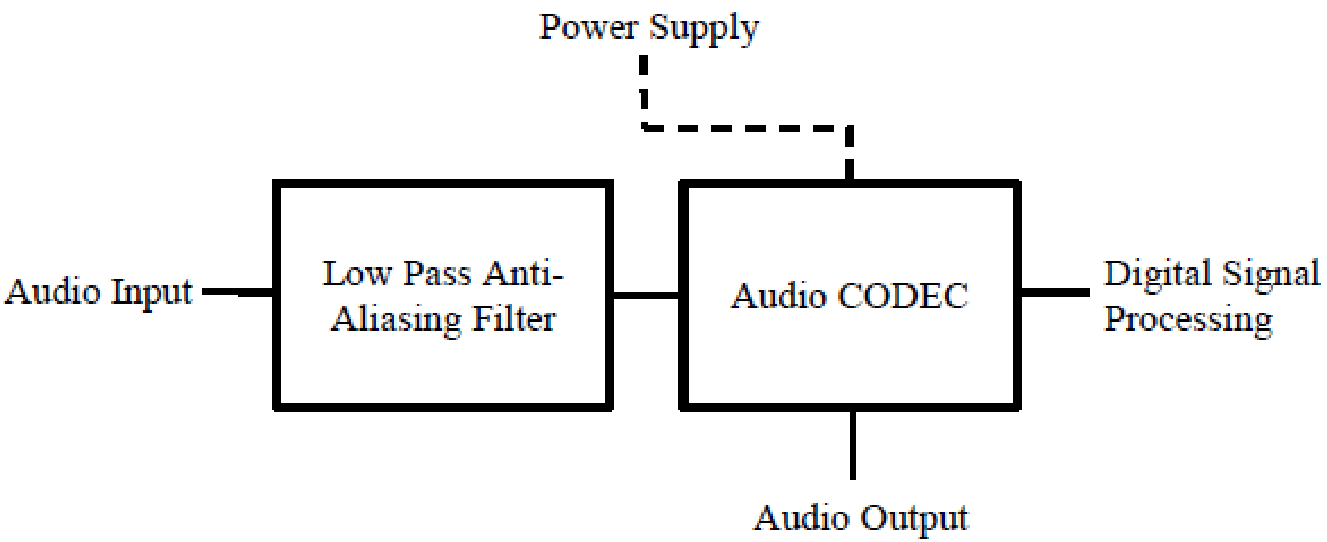
Original Design: DSP

- Generates output wave based on input
- Basic Algorithm:
  - Acquire input wave
  - Take FFT
  - Time shift and Frequency Scale
  - Take IFFT
  - Output new wave



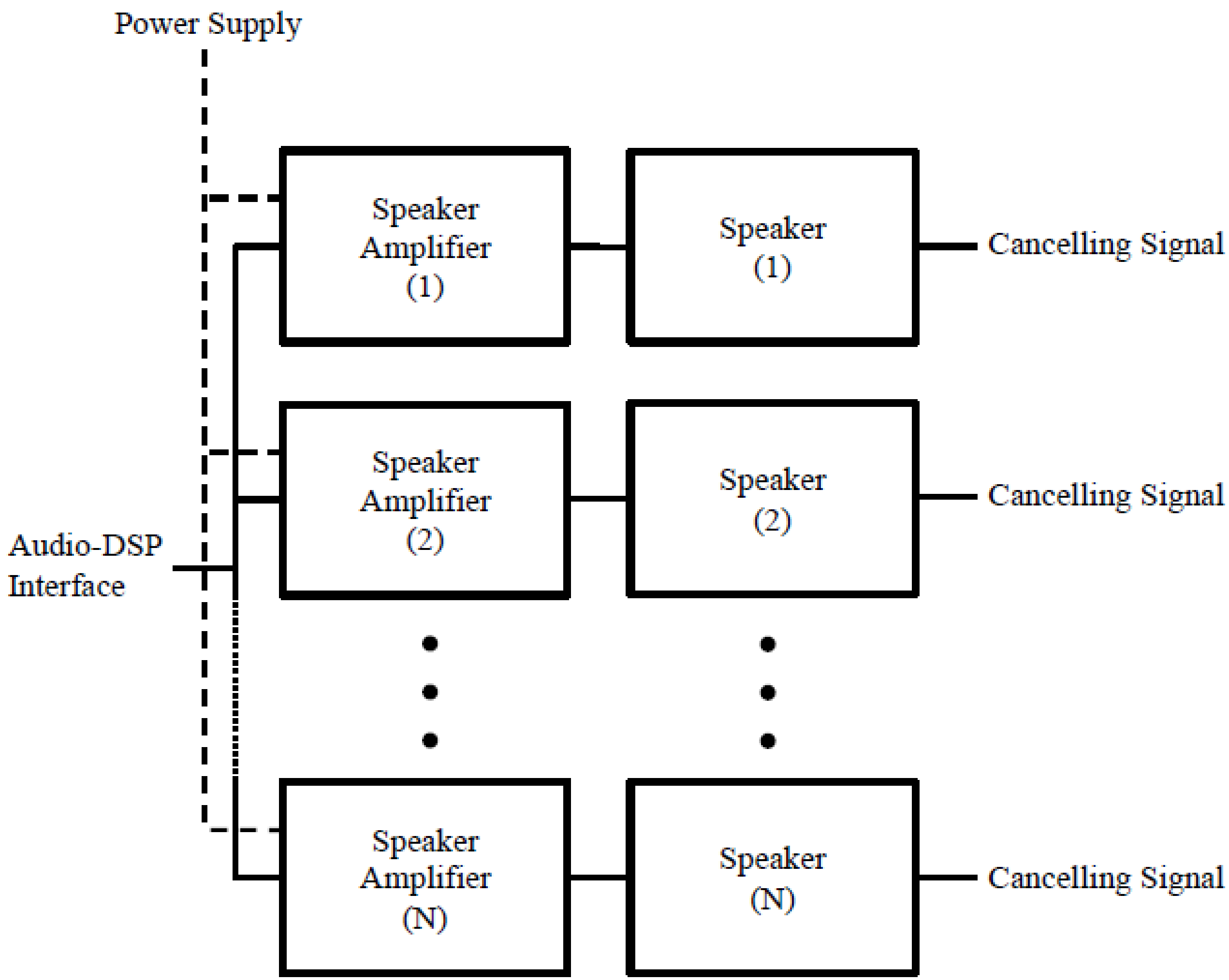
Original Design: Audio-DSP Interface

- Filters input and digitizes it for DSP processing
- Converts signal back to analog for audio output
- 1st order RC Low Pass Filter w/ 3 kHz cutoff
- Audio Codec samples at 8 kHz



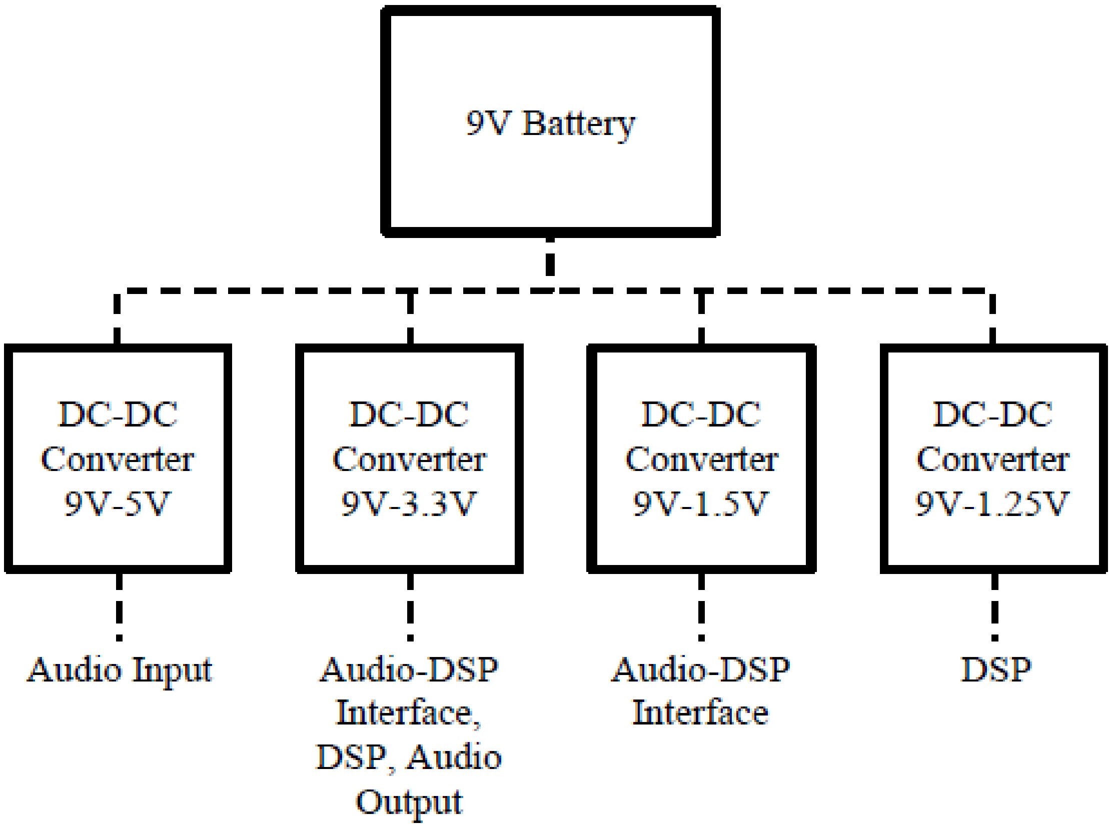
Original Design: Audio Output

- Outputs inverted signal
- N = 8



Original Design: Power Supply

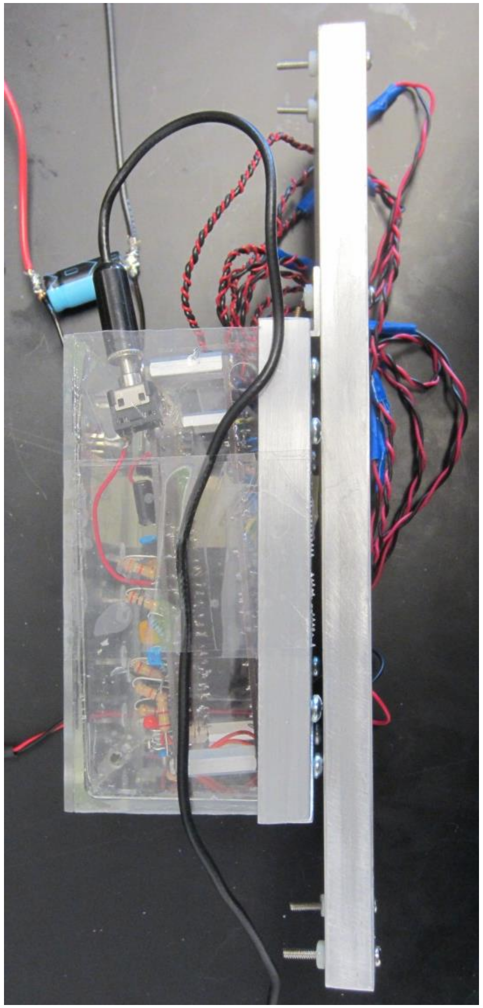
- Powers other modules
- Powered by 9V
- Uses Four Voltage Regulators

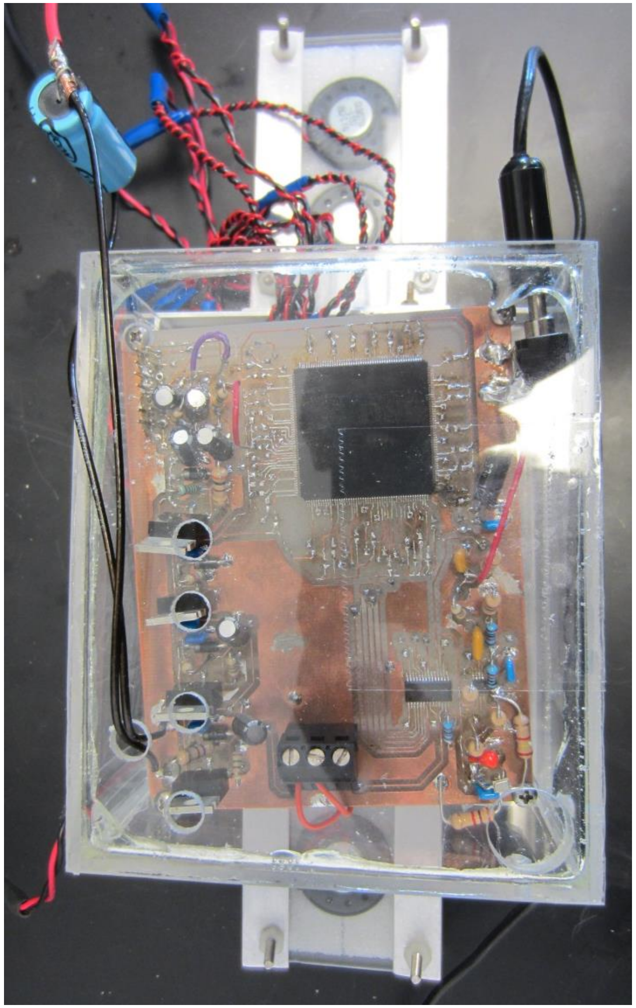


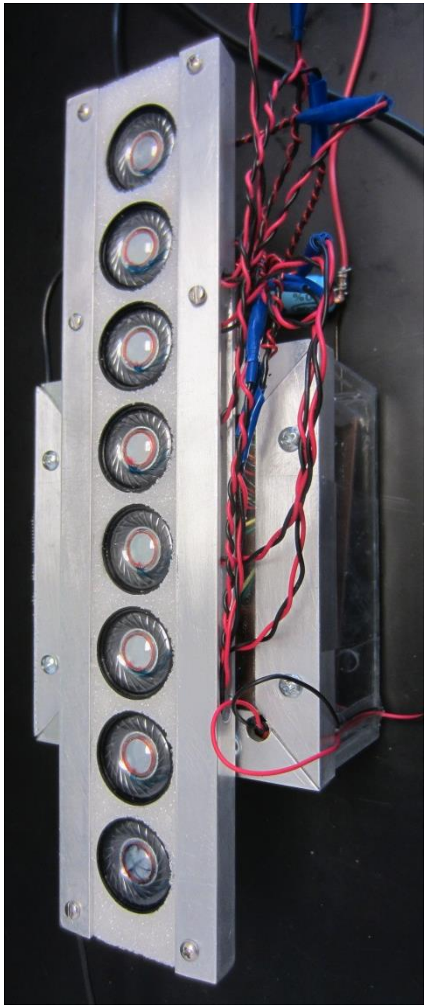
Original Design: Fabrication

- Two PCBs
  - one for output circuit speaker array
  - one for everything else
- Fabricated in Electronics Service Shop

Original Design: Fabrication







Modifications to Original Design

- Couldn’t program DSP
  - JTAG emulator cost $1,000
  - Used DSK (demonstration kit) instead
- 9V battery could not supply necessary current
  - Used power supply instead

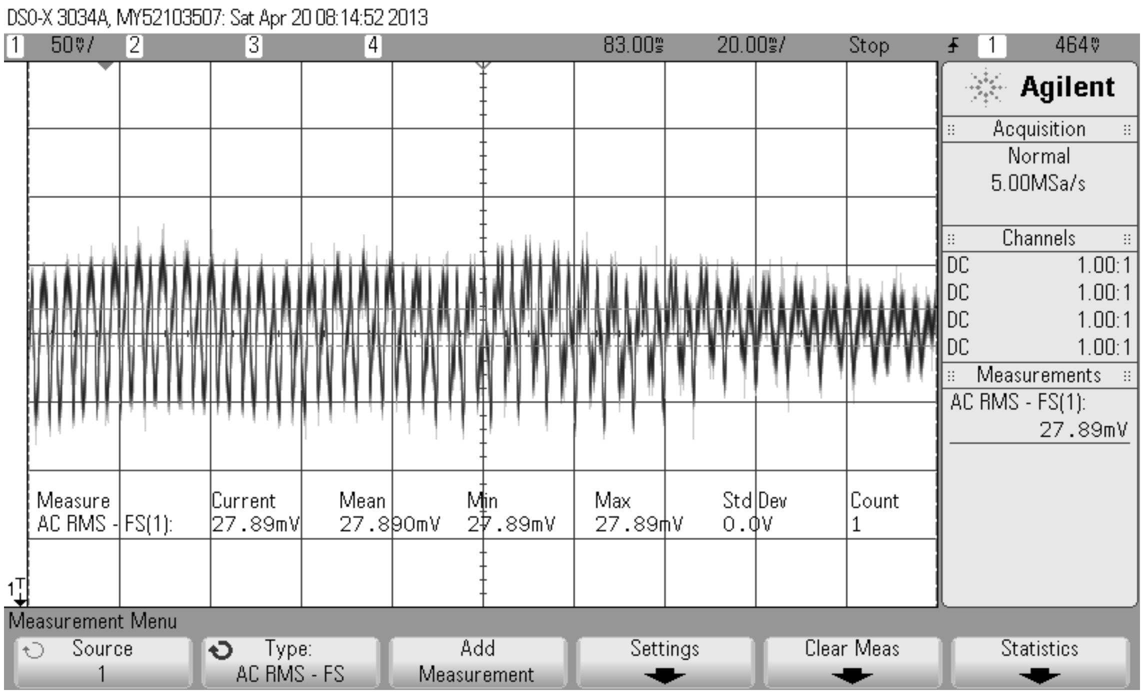
Requirements and Verification

Component Block	Requirements	Verification Results
Audio Input	• Acquire sound signal • Amplify low level signal	Pass Pass
Digital Signal Processor	• Phase shift signal by 180° • Configure CODEC • Power-up sequencing	Fail Pass Pass
Audio-DSP Interface	• Sample at correct frequency • Perform low-pass filtering	Pass Pass
Audio Output	• Amplify shifted signal • Emit phase shifted signal	Pass Pass
Power Supply	• Source enough current • Supply voltages	Pass Pass

Requirements and Verification
Audio Input

- Mic output within 0.5224 mV to 896.9 mV

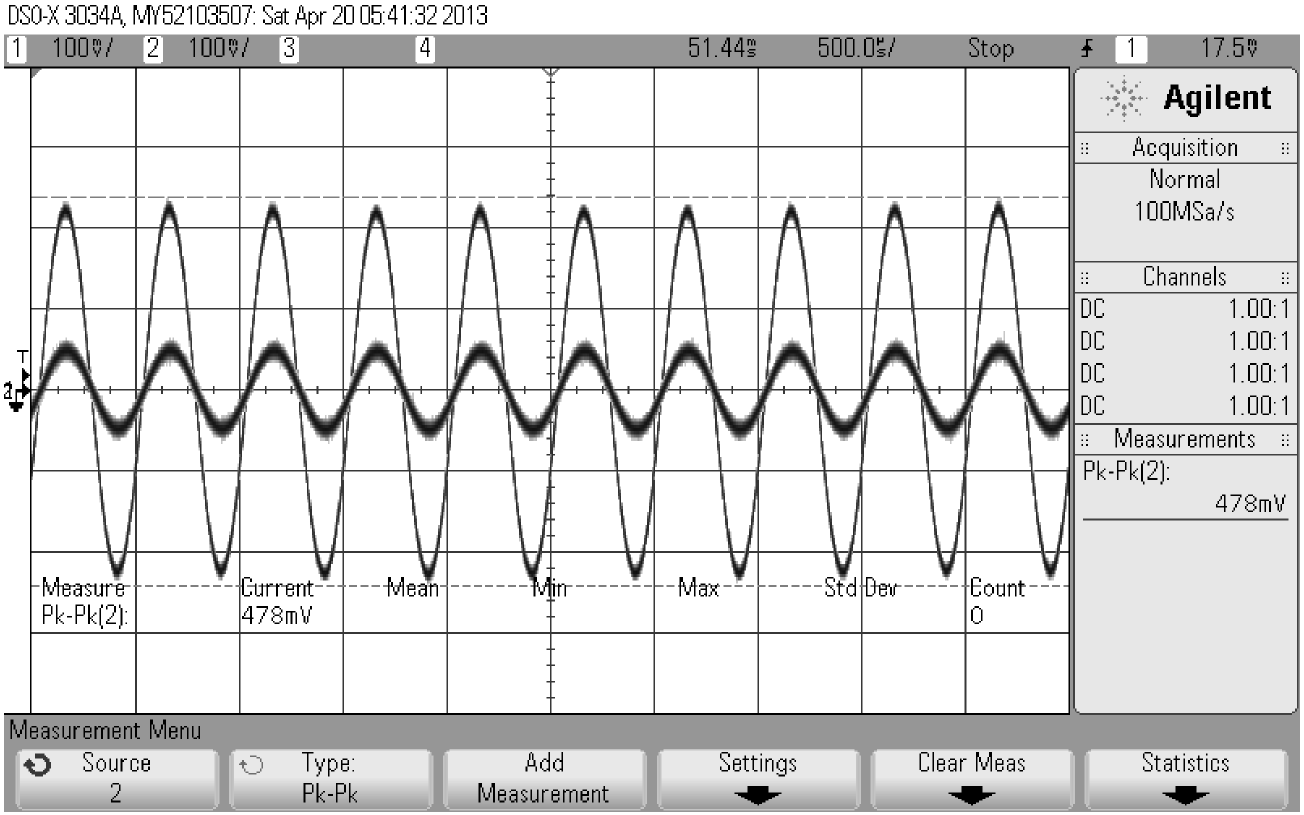
  27.89 mV from 8 cm under normal conditions



Requirements and Verification

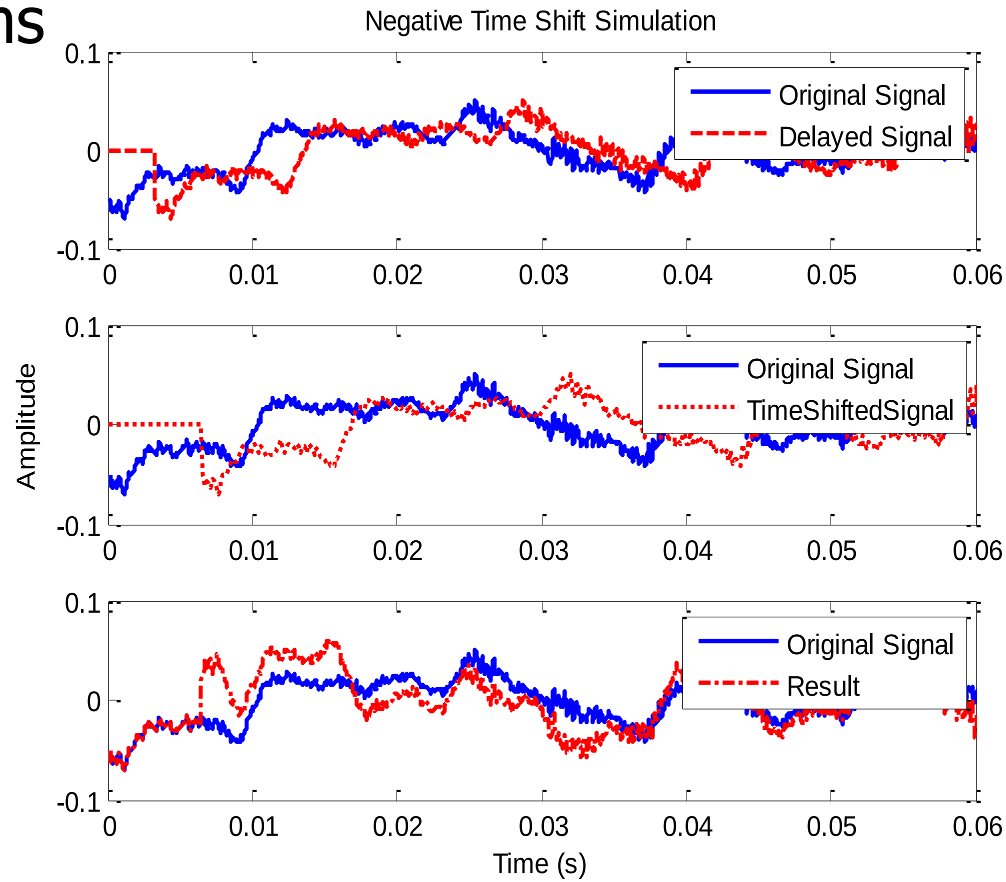
Audio Input

- Microphone amplifier voltage gain is 4.78 V/V
- Specified as 5V/V±0.25V/V



Requirements and Verification
Digital Signal Processor

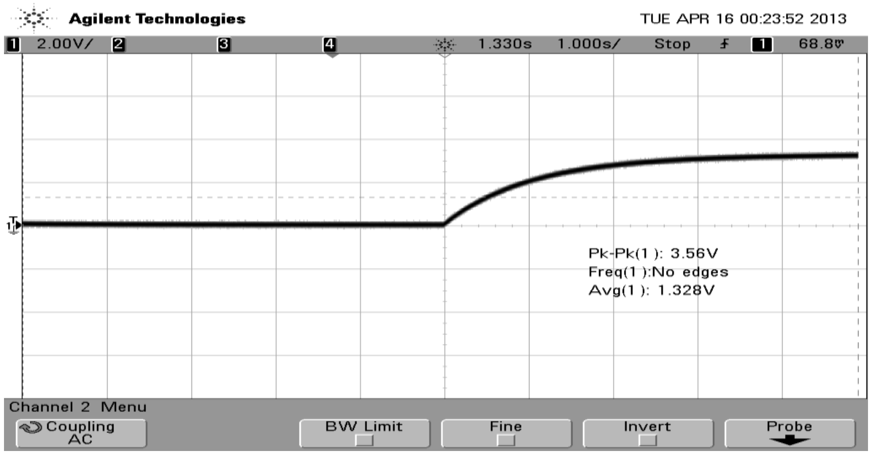
- 180° ±18° phase shift could not be attained
- DSP latency ~ 3.26 ms
  - Destructive interference every n(153 Hz)
  - Constructive interference every n(153 Hz) + 76.5 Hz
- Negative time shift worsened problem



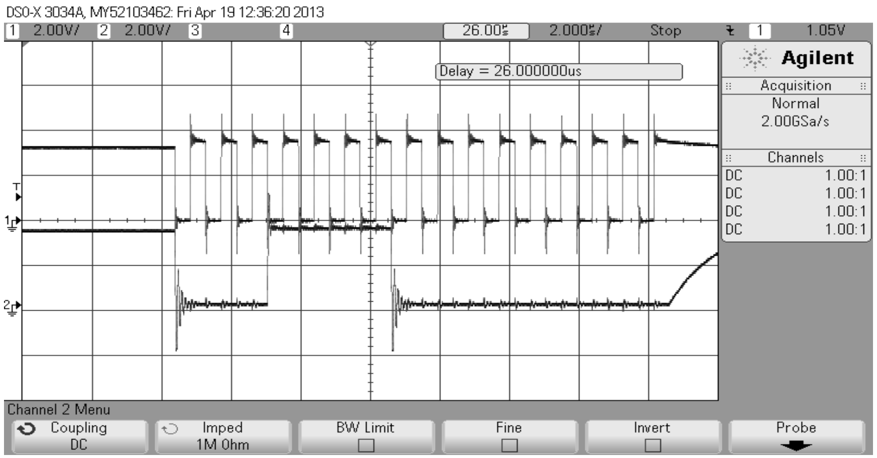
Requirements and Verification

Digital Signal Processor

- Reset pin on DSP is held low for 1 ± 0.5 seconds



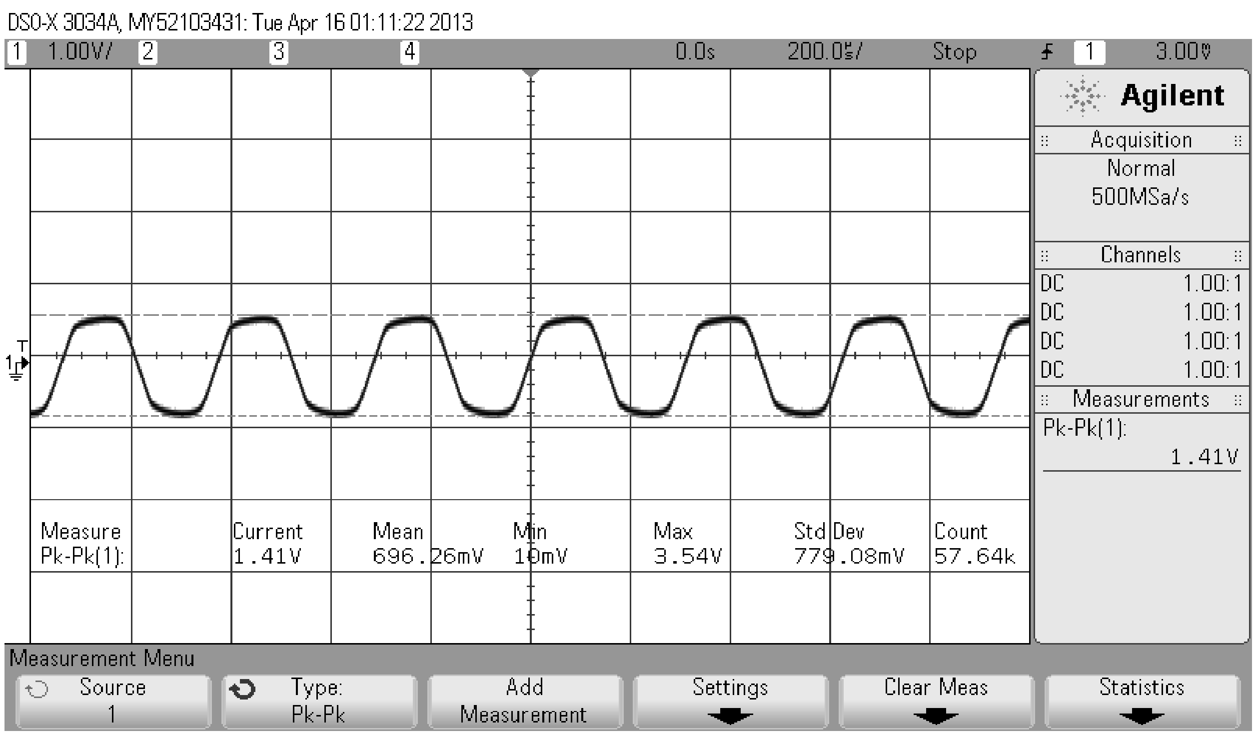
- Audio CODEC is configured correctly



Requirements and Verification

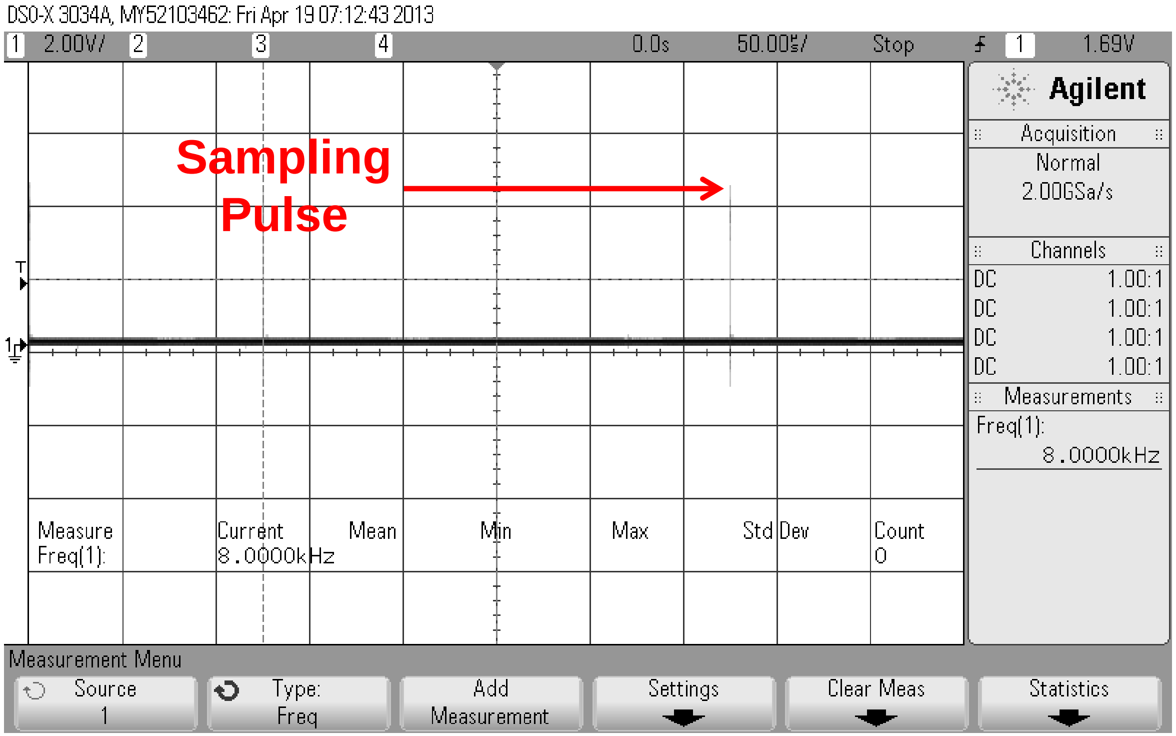
Audio-DSP Interface

- The low pass anti-aliasing filter has a 3 dB cutoff frequency at 3 ± 0.3 kHz
- Actual f_{3dB} = 3.2 kHz with 2.01V Pk-Pk input wave



Requirements and Verification
Audio-DSP Interface

- Audio CODEC sampling rate verified as 8 kHz

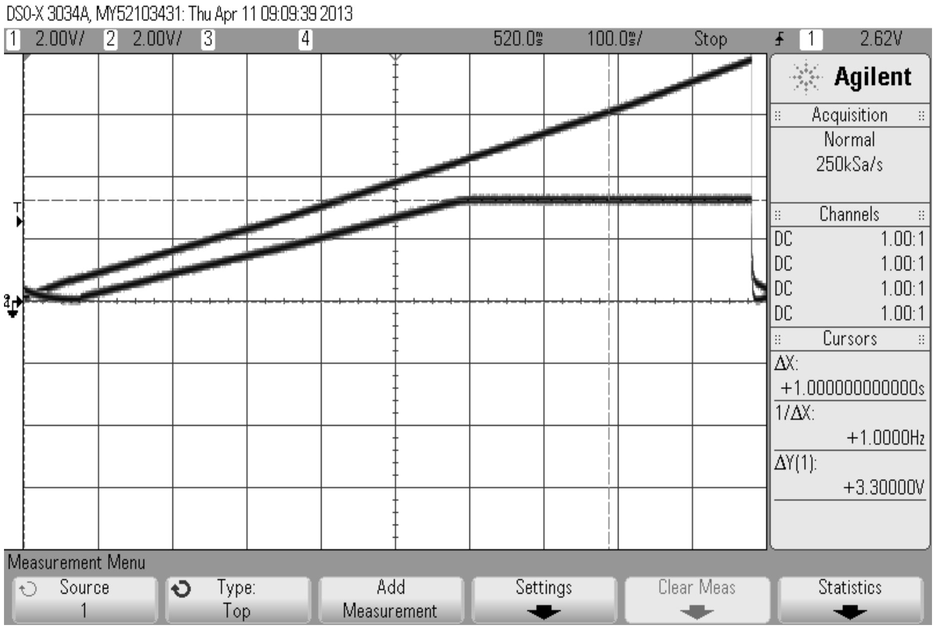


Requirements and Verification

Power Supply

- DC-DC Converters tested with 9V source
- Minimum supply voltage tested using voltage sweep
- Must operate with input voltage ≥ 6.25V

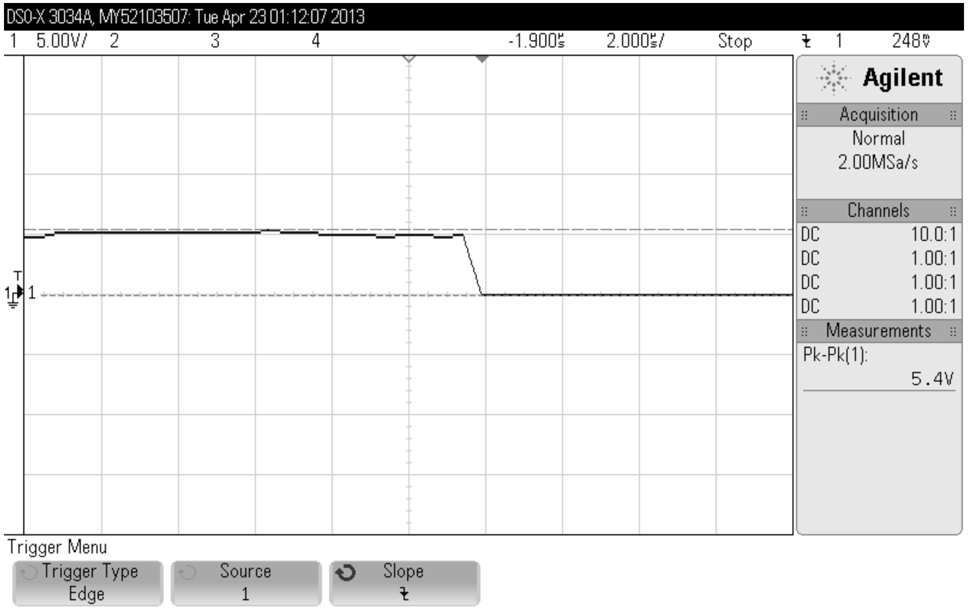
Spec	Actual	$V_{in,min}$
5 V ±0.05 V	**4.97 V**	**6.25 V**
3.3 V ±0.033V	**3.29 V**	**~**
1.5 ±0.015	**1.51 V**	**~**
1.25 ±0.0125	**1.25 V**	**~**



Requirements and Verification

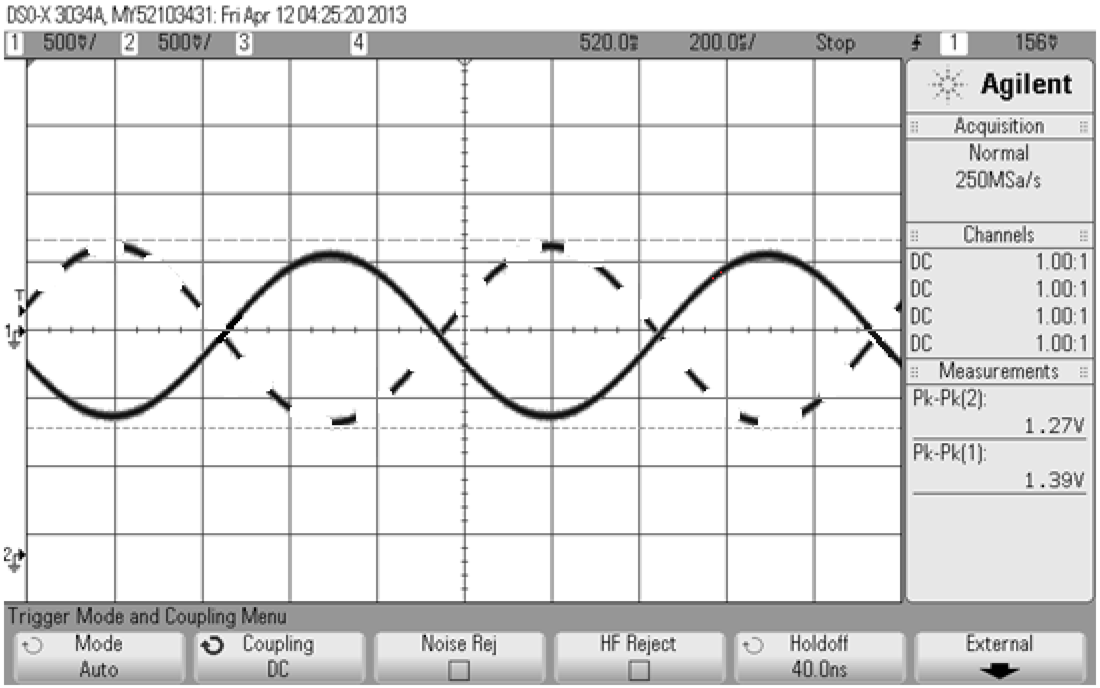
Power Supply

- Maximum power draw of circuit ~ 1.5 A
  - 9V battery could not supply this current
  - DC Power Supply used instead
- 9V battery output voltage across 6 ohm load



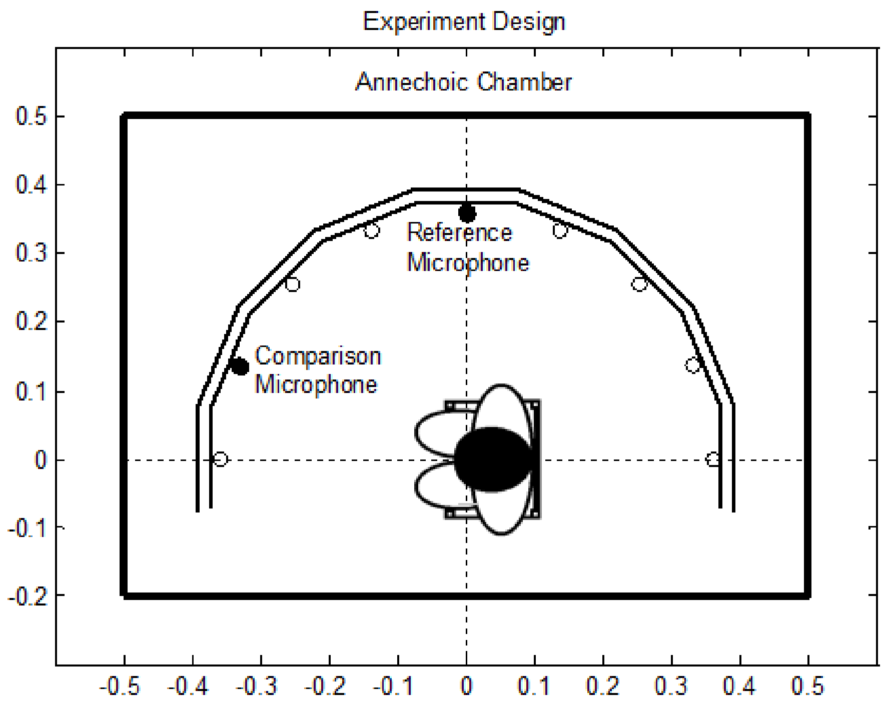
Requirements and Verification
Audio Output

- Speaker amplifier gain verified at 0.91 V/V
  - Tolerance is 0.89±0.05 V/V



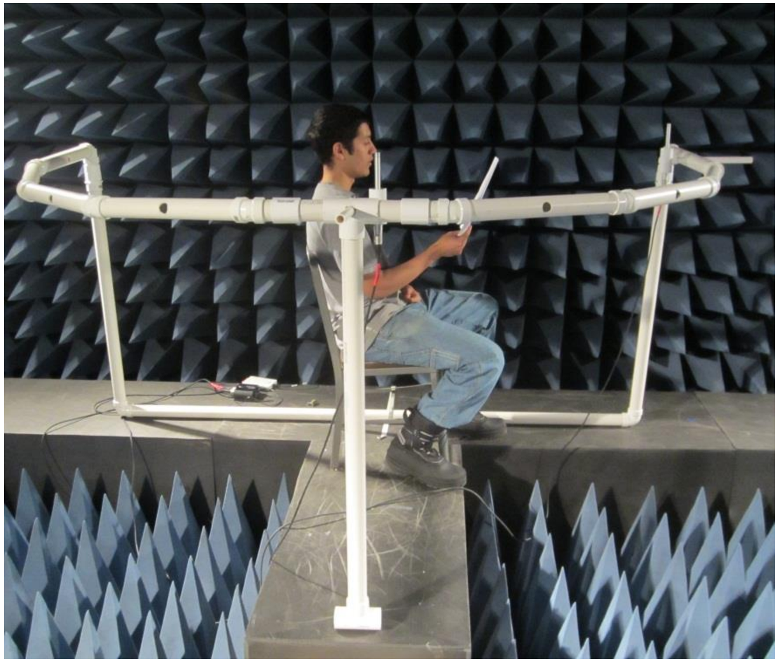
Voice Characterization
Experiment Design

- Measure acoustic radiation pattern of human voice
- Measure typical frequency content
- One reference microphone and one comparison microphone



Voice Characterization
Experiment Design

- Speech sample containing many English "consonants, vowels, and clusters" read at 9 microphone positions

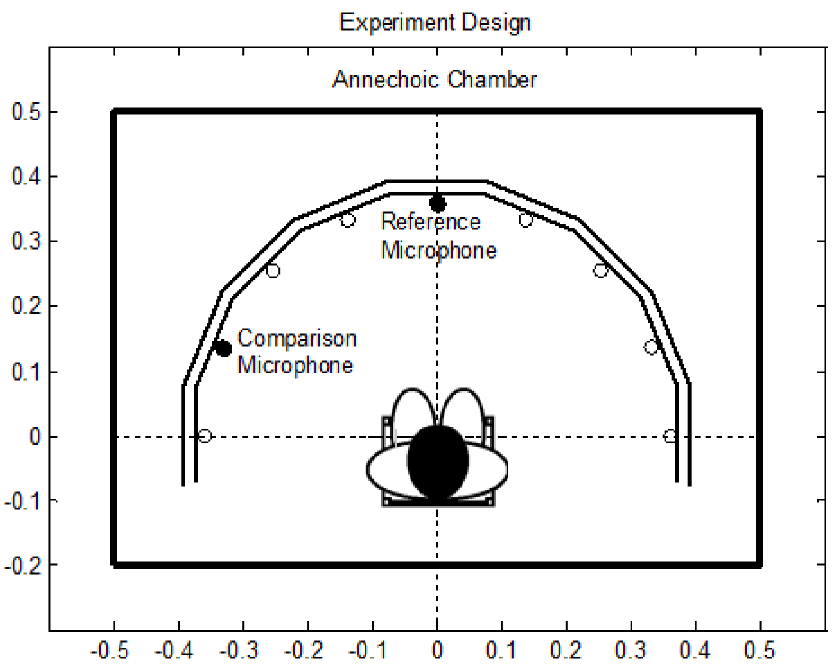


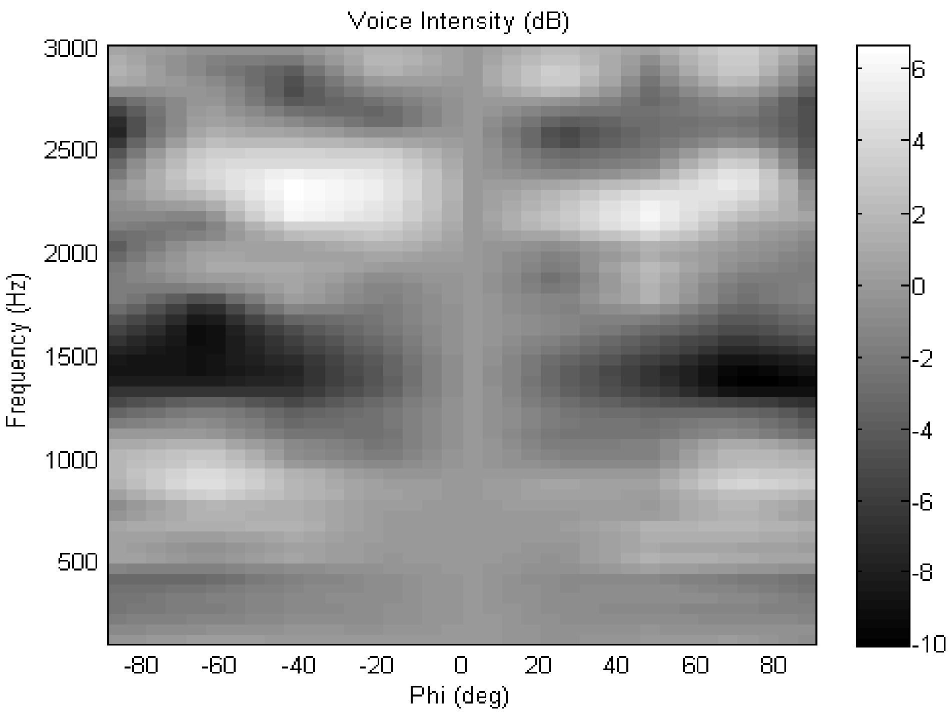
Please call Stella. Ask her to bring these things with her from the store: Six spoons of fresh snow peas, five thick slabs of blue cheese, and maybe a snack for her brother Bob. We also need a small plastic snake and a big toy frog for the kids. She can scoop these things into three red bags, and we will go meet her Wednesday at the train station.

Weinberger, Steven. (2013). *Speech Accent Archive*. George Mason University.

Voice Characterization
Experiment Verification

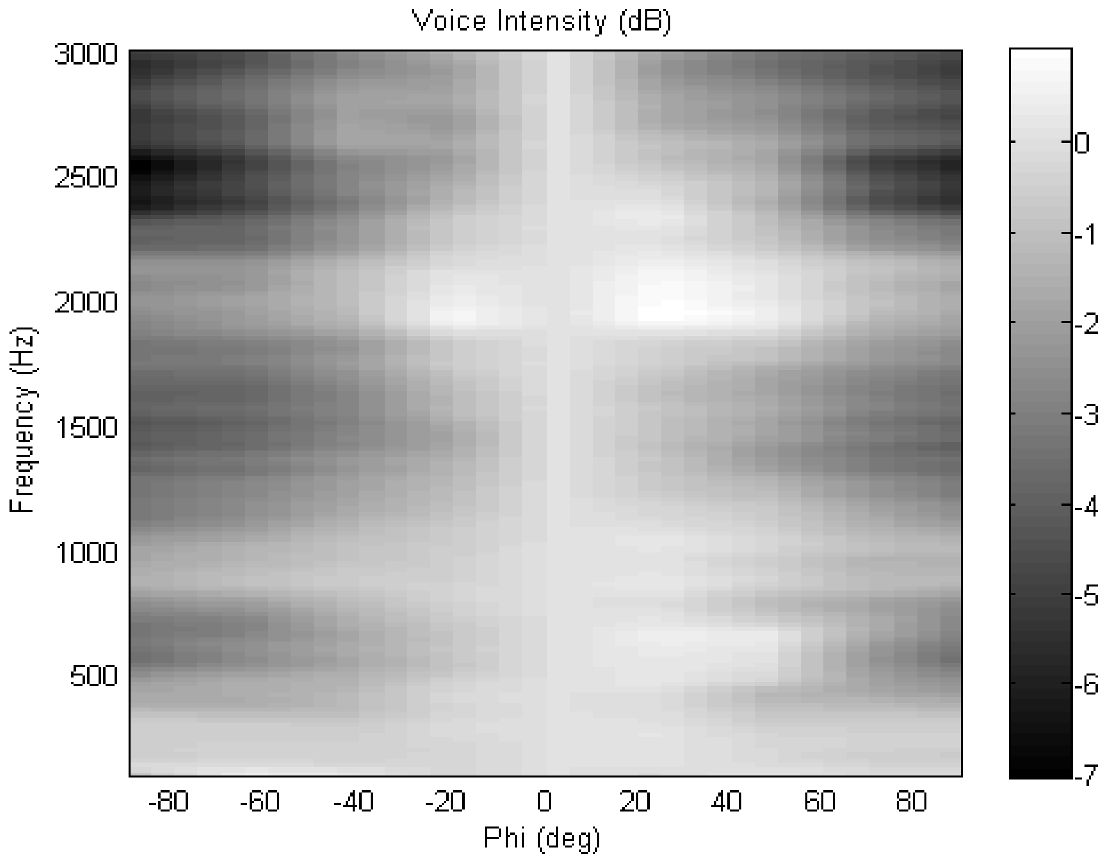
- Looking for symmetry (pattern of Joel's voice)





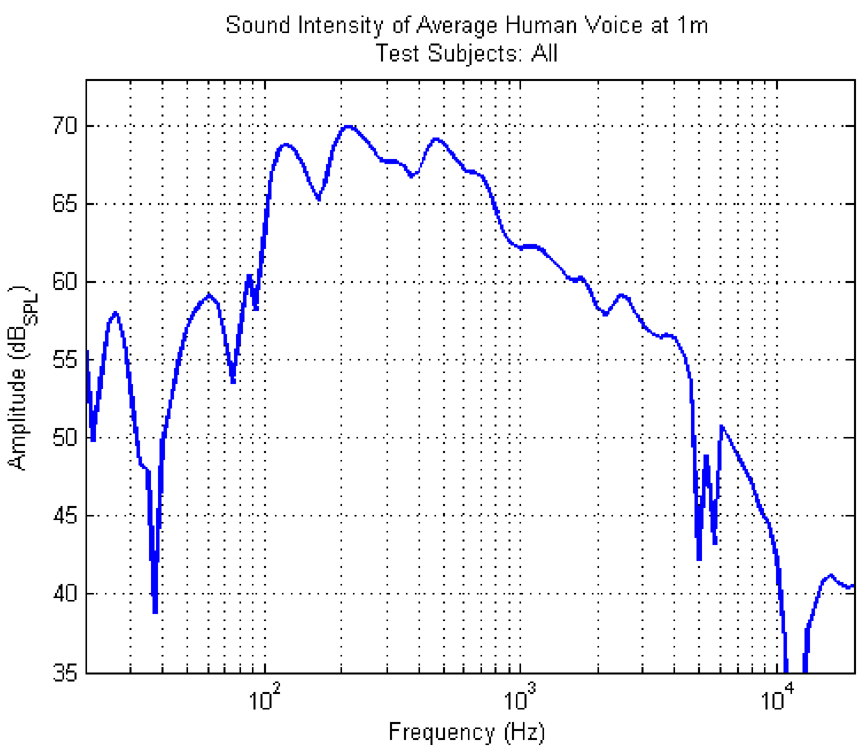
Voice Characterization
Experiment Verification

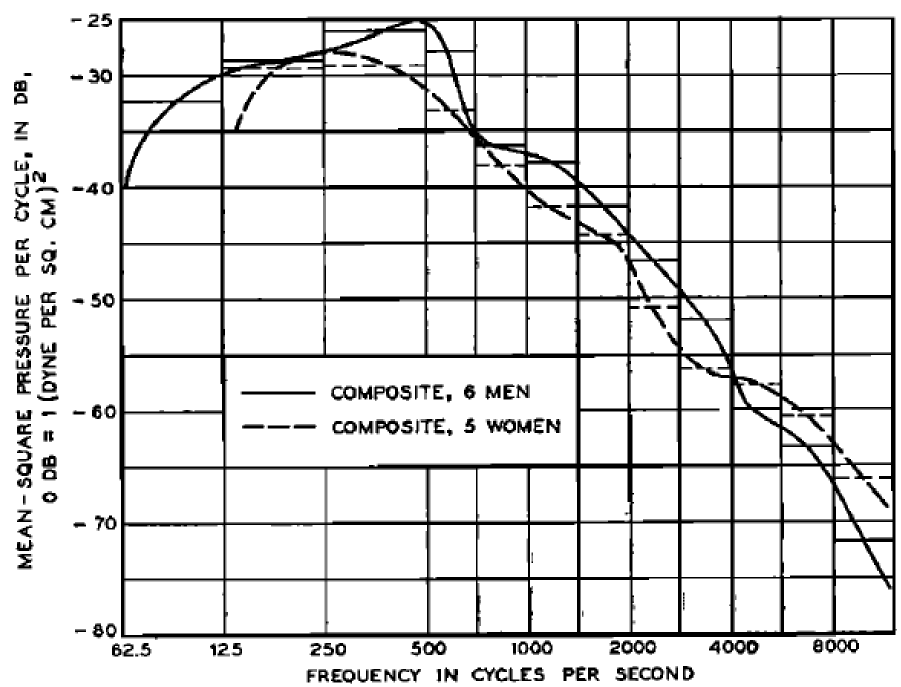
- CUI Speaker used in array



Voice Characterization
Experiment Verification

- Average frequency content (3 men, 2 women)

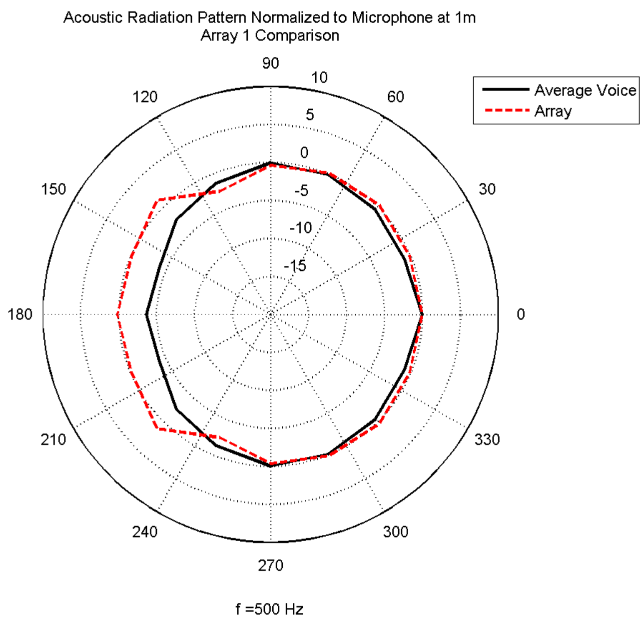


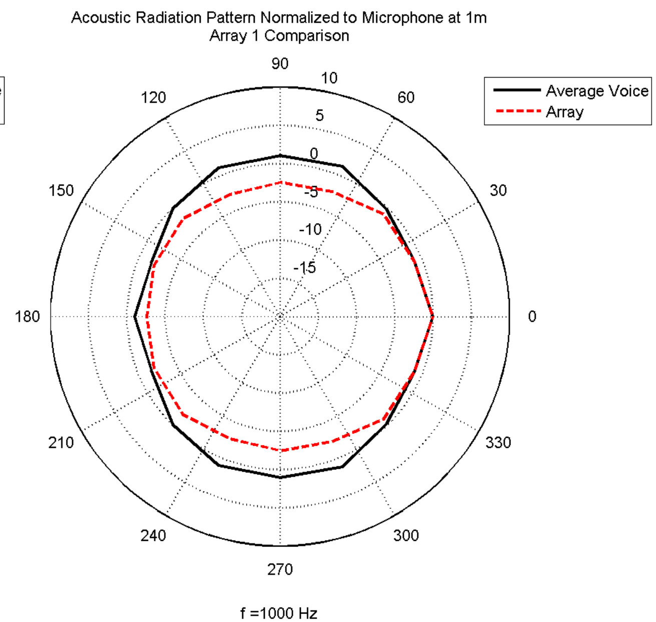


H. K. Dunn, S. D. White, "Statistical Measurements on Conversational Speech", J.A.S.A, vol. 11, pp. 278-288, Jan. 1940.

Speaker Array Evaluation

- Attempt to match acoustic radiation pattern of average human voice
- Several speaker arrays evaluated
- Single element pattern found to be best match





Further Testing

- Tested to see if we can cancel a single frequency at a given point in space
- Found that we can achieve partial cancellation
- Difficult to match phase and amplitude
  - Speaker frequency response is not flat

Summary and Conclusions

- We acquired the acoustic radiation pattern of human voice
- We built a circuit to read in a signal and output an inverted version (for certain frequencies)
- We tested multiple speaker arrays and chose the one with the response most similar to that of the human voice

Summary and Conclusions

- Our method will not work
  - Latency
  - Radiation patterns differ on individual basis
  - Must know sound source location to match amplitude
  - Small speakers have poor low frequency response

Possible Future Work

- Using feedback to control amplitude
  - Needs more microphones
  - Bulky and not portable
- Using inverting amplifiers
  - No latency
  - Gives up frequency scaling (not feasible with length 32 FFT either)
- Use of FPGA
  - Decrease latency

Credits

Professor Jennifer Bernhard

EM Laboratory students and staff

Professor Jont Allen

Parts Shop staff

Our TA Justine Fortier

ECE 445 entire staff

Thanks for Listening!