Project 21: Wall Of Sound

Beta Demo

Vectorized Acoustic Deterrence of Elephants Research

Team Members: Arpad Voros, Greyson Fitts, Hunter G. Cook, Morgan Pyrtle, Nwaf Alamro

Sponsors: Army Research Office: Paul Reid, Stephen Lee

Mentors: Dr. Pitts, Dr. Gupta, Dr. Scheifele

Project Background

- Create a passive deterrence system which inhibits elephants from trespassing on farmland, reducing the number of casualties of humans and elephants.

- To broadcast 10Hz - 15kHz (range of elephant hearing).
- Not cause any physical or psychological harm to any organisms.

- Have to accomodate for terrain, vegetation, weather patterns, and animal interference.

Project Timeline

CDR February 12th	Post CDR Feb. 12 - 29	Alpha Prep Mar. 1 - 18	Alpha March 18th	Beta Prep Mar. 18 - Apr. 16	Beta April 16th
- Measure characteristics of transducers using LCR meter - Begin designing amplifying circuit with transducer as load - Begin PCB layout of mixing circuit	- Complete amplifying circuit, test and debug - Begin PCB layout of amplifying circuit - Complete and purchase mixing PCB - Begin microphone circuit (no PCB)	- Complete and purchase amplifying PCB - Debug and verify mixing PCB - Begin 3D modeling of hexagonal encasing (LCD billboard) - Ensure encasing includes leads which connects commons, audio source, and power	- Using PCBs, perform audio test - If works, use microphone to begin radiation pattern - If does not work, test and debug, redesign appropriate PCB(s) - Begin software/MCU SD card reading of audio	- Polish PCB(s) and reorder appropriately - Begin 3D printing hexagonal casing, fit flush with hexagonal PCB - Radiation pattern, if not done already - Complete MCU audio reading, start on a simple UI	- Present working product with at least 2 hexagonal faces linked together - Play various audio files from MCU - Simple UI for selecting audio, stopping and playing of sound

Circuit Design | PCB Layout | Purchase PCB | Debug PCB | PCB encasing | Radiation Pattern | MCU/Software | Final Prototype

Beta Plan Changes

Sound Board	- Interface uSD card reader with the rest of the system - UTF screen to allow user to play, stop, and shuffle through various files - Work with Dr. Scheifele to select sounds appropriate to the system's purpose & application
Mixer	- Either add DC offset to TRS connector or apply this DC value to an offset pin on the AD633ARZ - Incorporate stand-alone frequency generator IC - Power connector - Add variable amplifier to output (and/or every port) - Replaced with standalone DAC
Amp / Transducer	- Remove caps for each amp, and have one larger cap to reduce complexity - Larger traces to match other board. - Possible changes to fix nonlinearity effect → Input via standalone DAC - Maybe add female housing pins for the transducer leads rather than soldering them in. In the case that some are faulty, they can be pulled out and placed in.
Modular Enclosure	- Create arrangement so pieces can easily branch from one to the other.
Microphone	- Create an array using multiple microphones to detect radiation patterns - Generate stand alone circuit (independent of AD2) which can be recognized as a microphone by a PC and capture information about analog signal through AUX/USB.
Misc	- Order more components to make multiple transducer boards - Power over ethernet to boards to reduce mess. - Connect PCB encasing to larger structure - Test array outside & anechoic chamber

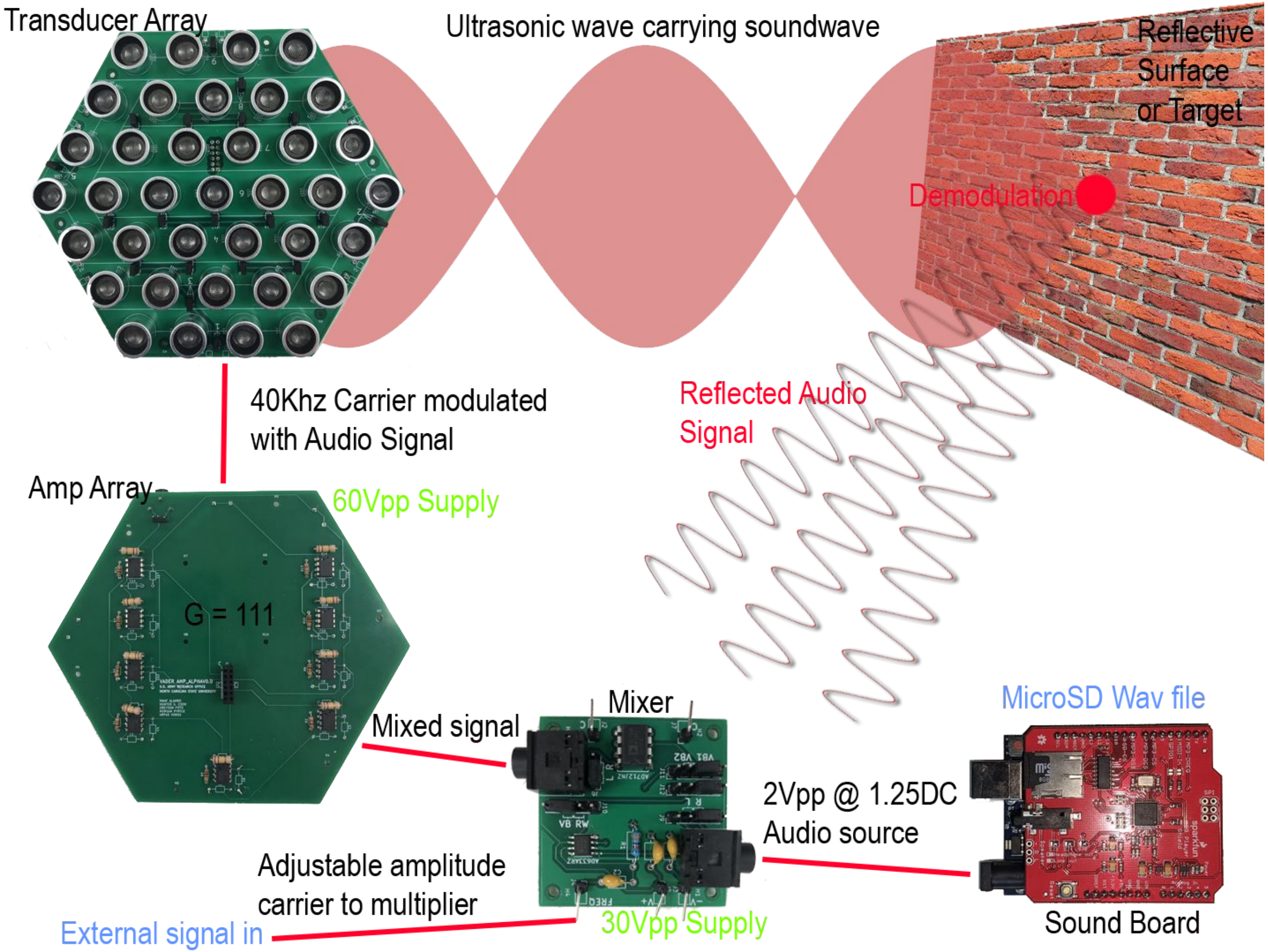
Legend
REMOVED
ADDITION
IN PROGRESS - GOOD STANDING
IN PROGRESS - POOR STANDING
COMPLETE

Member Accomplishments Since Alpha

- **Hunter**
  - **Talk** with Dr. Garner about low frequencies (it is a bust), **flesh** out hysteresis procedure with Dr. Pitts, **experiment** said procedure (testing different loads, components, and amp setups), **talk** with Skip about project possibilities, **manifest** MATLAB code for digital modulation and test with 96kHz DAC for proof, **research** new modulation techniques, **find** new DAC (192kHz), **setup** power adapter for ethernet interface, **drink** water
- **Greyson**
  - **Develop** microphone circuit (self powered via USB, detect/record audio), **fix** unusual amplitude oscillation from microphone, **submit** all orders for additional components in preparation for Beta demo
- **Nwaf**
  - **Tested** out multiple keypads for controlling the soundboard, **Found** a stackable keyboard shield to interface with both UTF screen and Arduino, **developed** Arduino C program for controlling keypad and LCD, **tested** the feasibility of stacking the Arduino UNO, uSD card reader, keypad, and UTF screen, **maintained** the primary contact between Dr. Skip Scheifele and Team VADER, **inquired** with Skip regarding elephants behaviour.
- **Arpad**
  - **Lead** meeting with Skip (conscribed most of the agenda w/ questions), **debugged** hysteresis issue to conclude mixer is at fault, ***conjured*** MATLAB GUI to output different modulated sound files, **designed** revised amplifier PCB to include modular leads on edge, variable amplifier on input port, optional filtering, **designed** revised multiplier PCB but thrown out due to hysteresis fix
- **Morgan**
  - **Developed** PCB enclosure design; **printed** the physical PCB enclosure; **built** the finished test bench stand; **interfaced** the enclosure with the larger stand

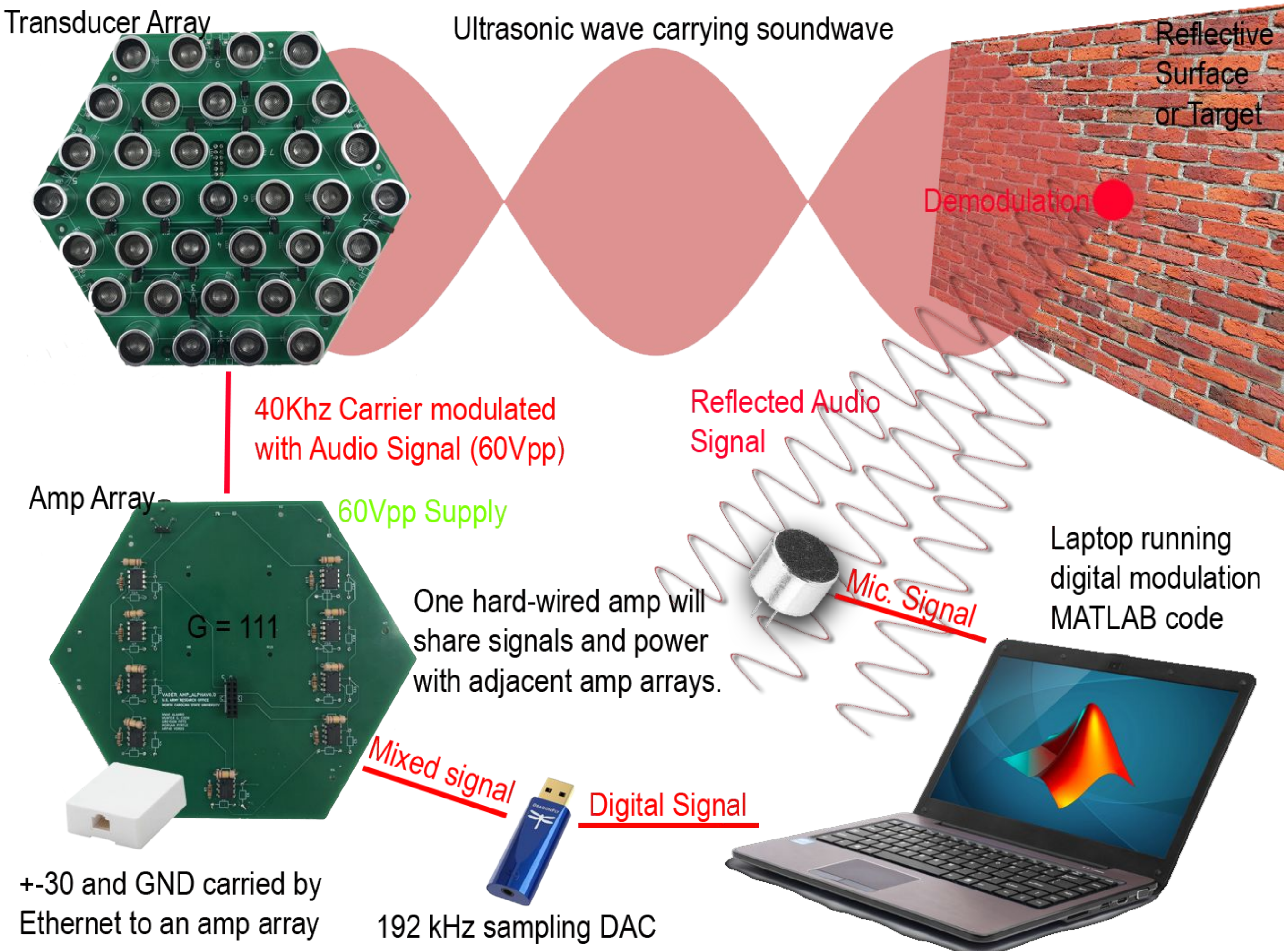
Old System

OLD SYSTEM


Transducer Array
Ultrasonic wave carrying soundwave
Reflective Surface or Target
Demodulation
40Khz Carrier modulated with Audio Signal
Reflected Audio Signal
Amp Array
60Vpp Supply
G = 111
Mixer
Mixed signal
MicroSD Wav file
2Vpp @ 1.25DC Audio source
Adjustable amplitude carrier to multiplier
External signal in
30Vpp Supply
Sound Board

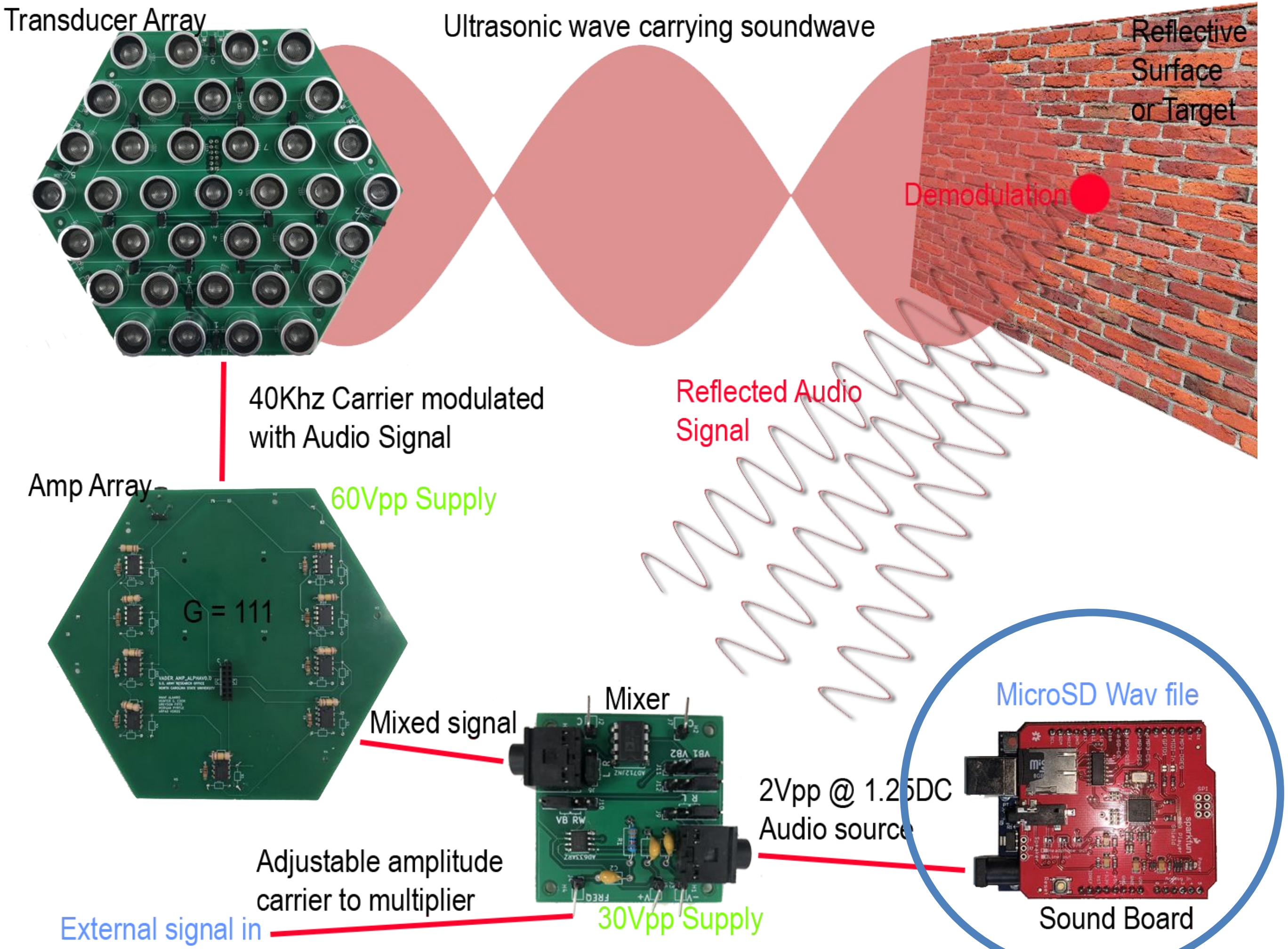
Revised System

REVISED SYSTEM



Subsystems

OLD SYSTEM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G = 111
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2Vpp @ 1.25DC Audio source
MicroSD Wav file
Sound Board
Adjustable amplitude carrier to multiplier
External signal in
30Vpp Supply

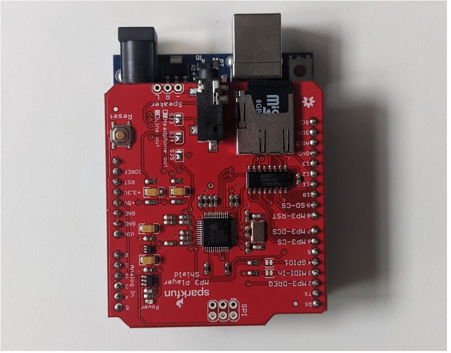
Subsystem - Sound Board

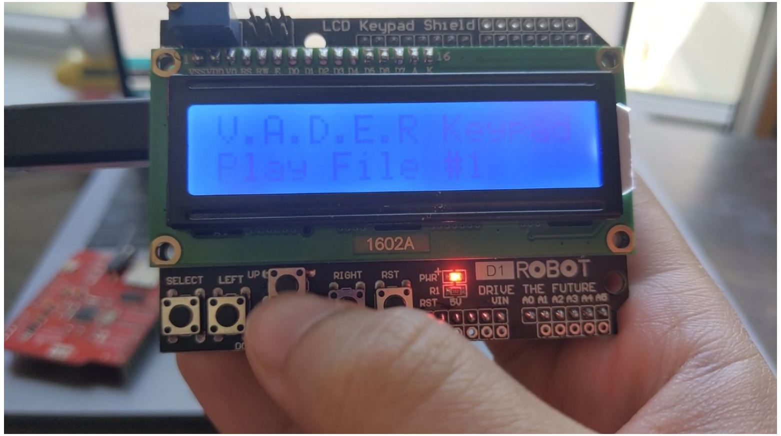
Sound Shield

- Uses VS1053 IC on an Arduino Uno
- Sound files are uploaded on a 16GB micro SD card.
- Files can be in WAV, MP3, VMA, AAC etc..
- Using the Arduino IDE, you can play, stop, and change the volume of the tracks.
- VS1053 DAC max output is 50kHz (unfortunately).

Keypad and LCD Control

- Capability of playing different tracks
- Shows filename on LCD
- No connection to a laptop





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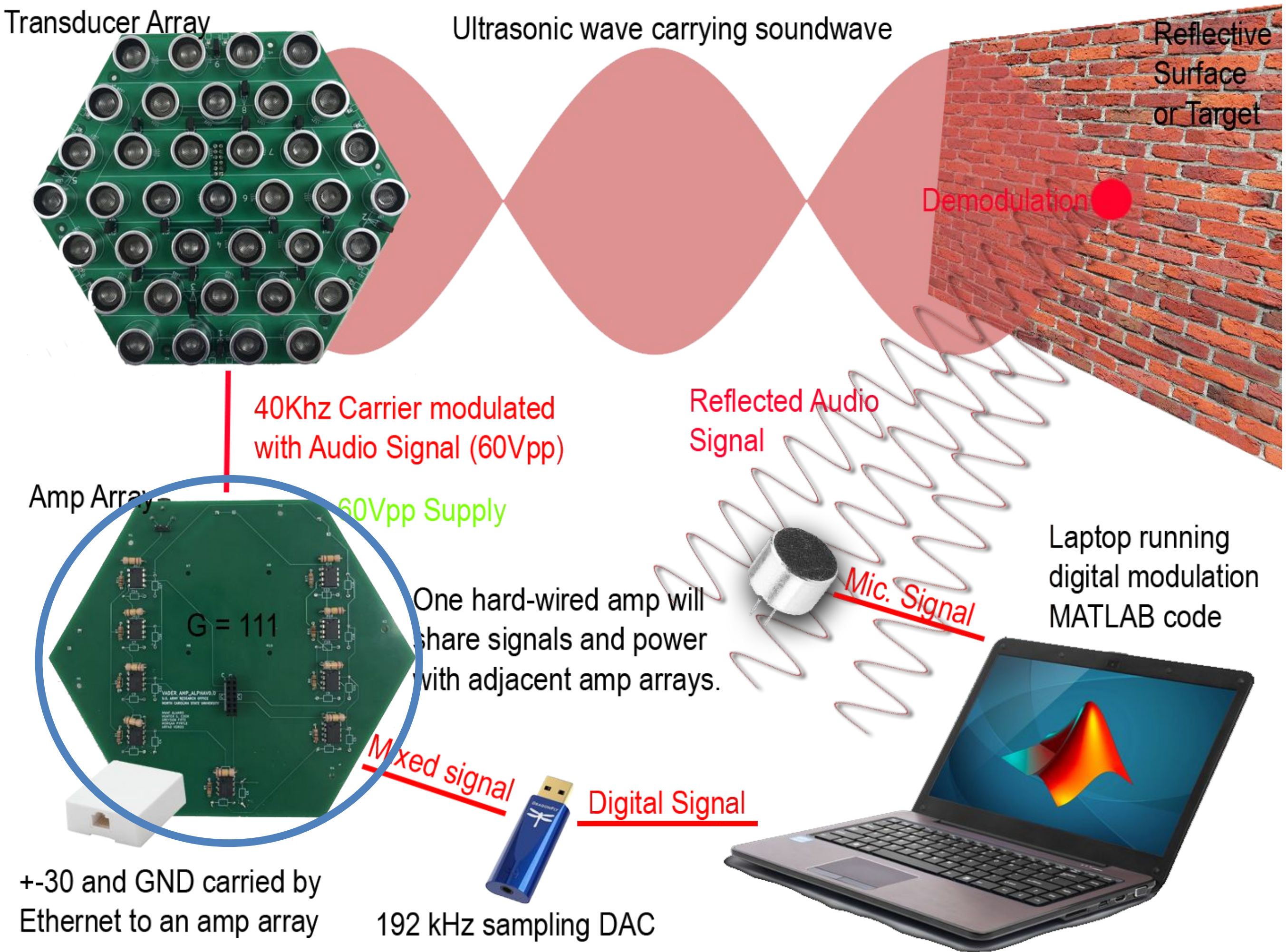
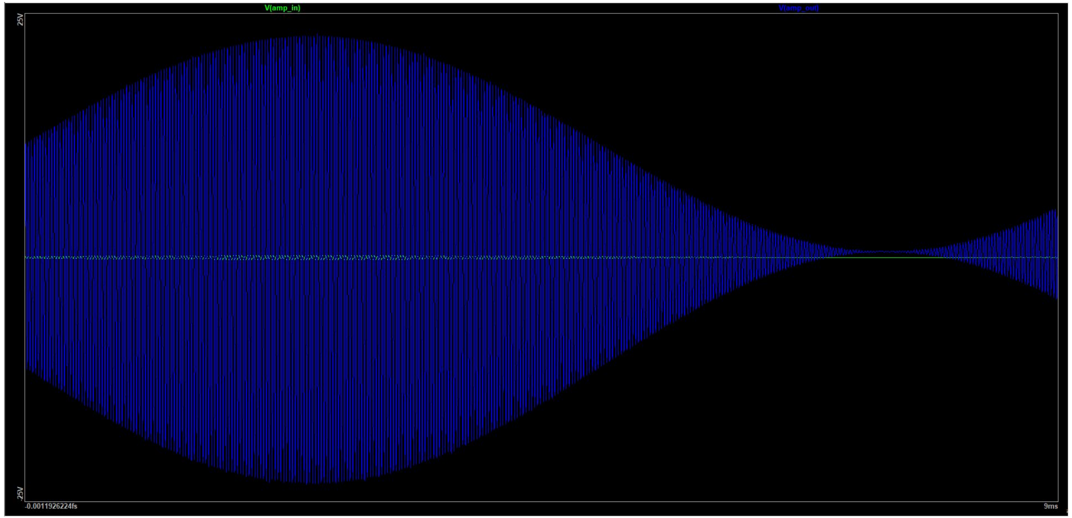
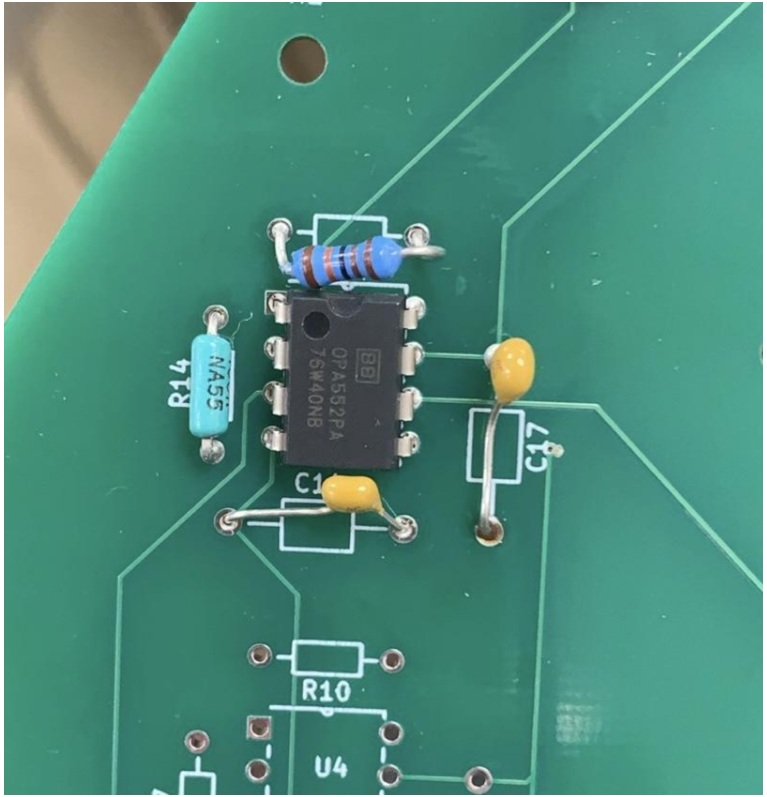


Subsystem - Amplifier Circuit

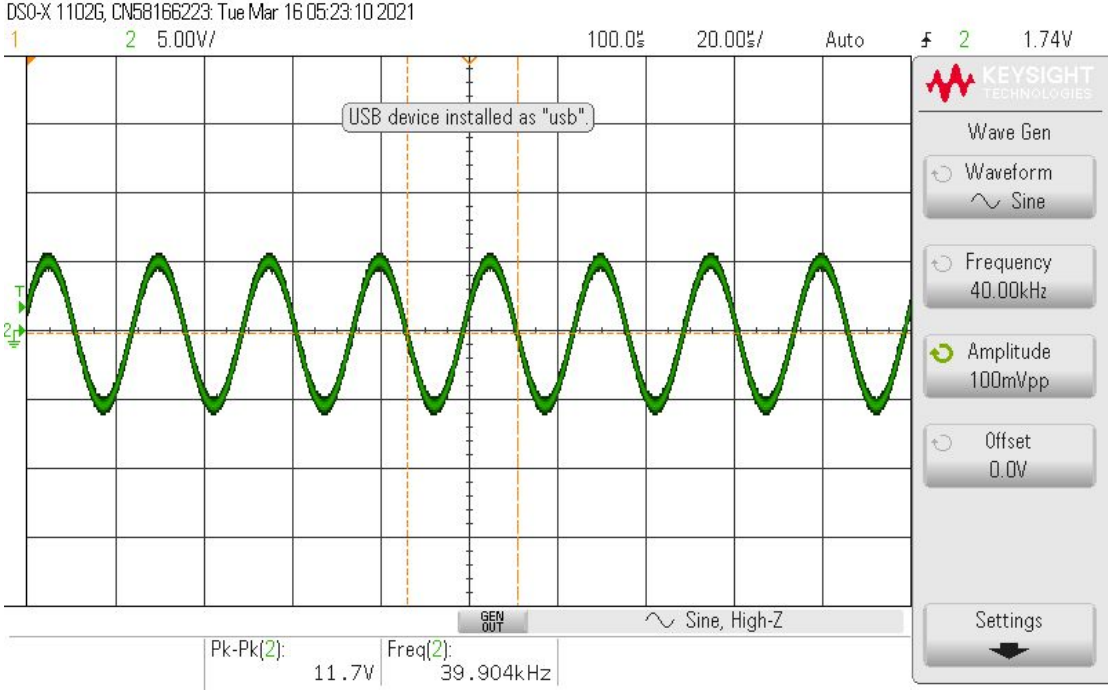
- Trying to achieve 60Vpp (±30V)
- Linear gain across audible spectrum. Simulated to be 111 to achieve with conventional DAC
- Was the largest perceived issue with Alpha Demo, so it was addressed heavily. Thus revealing revisions needed





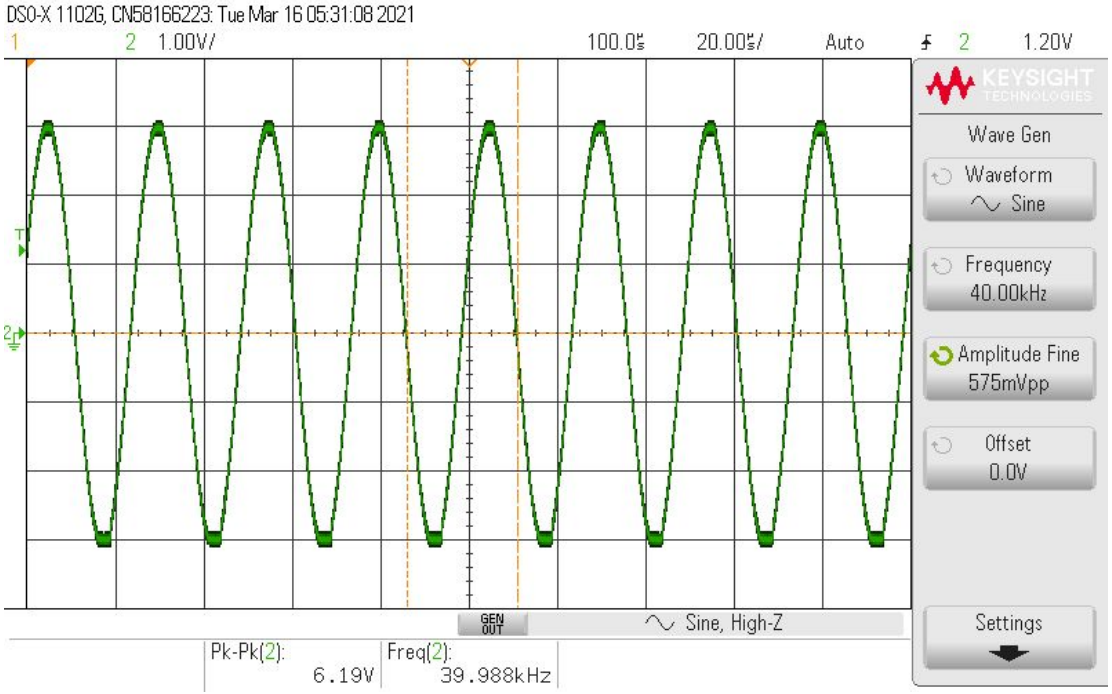
Amplifier Functionality - General

- Single 40kHz sinusoidal input of 100mVpp. Not mixed, Unloaded
- Output of 11.7Vpp, Gain of over 111. This is due to using a 113kΩ resistor instead of 110k so there is a slightly higher gain here
- Frequency bounces between 39.9kHz and 40.1kHz
- Test case was verified with this as presented in CDR.



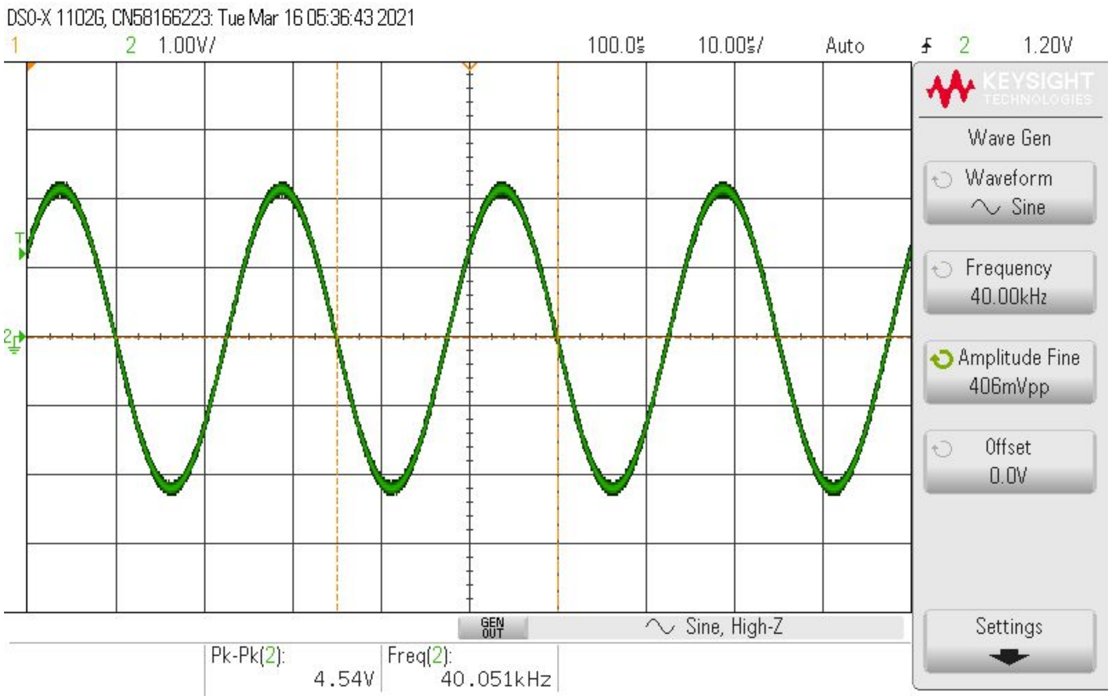
Amplifier Functionality - Clipping

- Single 40kHz sinusoidal input of 575mVpp. Not mixed, Unloaded
- Output of 61.9Vpp (Probe is 1:10)
- This was a test for clipping. We would expect it to occur at 60Vpp due to power supply limitation.
- We have decided to drive our transducers at this maximum level, since they are able to handle a max of 150Vpp



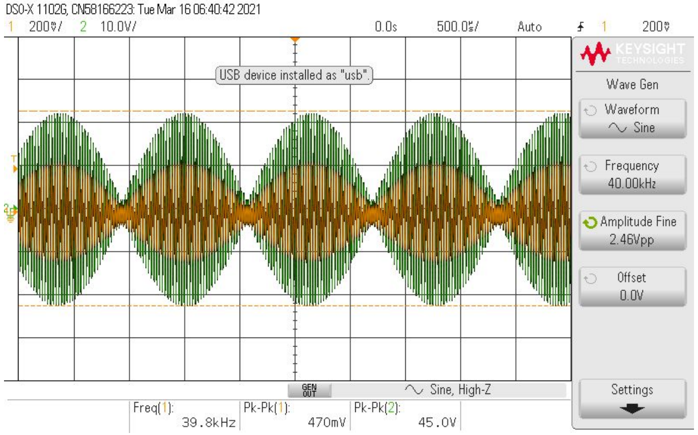
Amplifier Functionality - Loaded

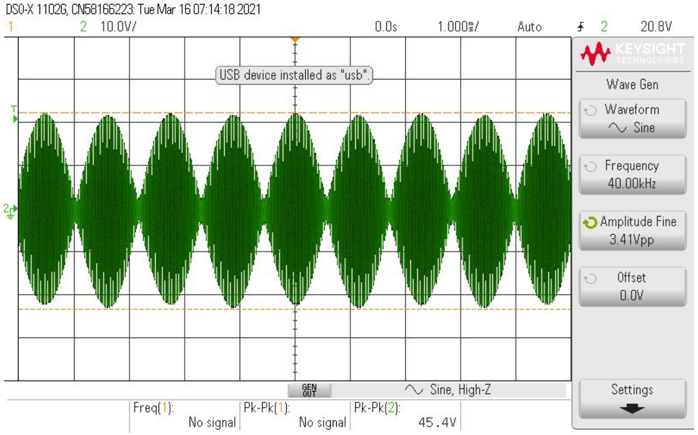
- Single 40kHz sinusoidal input of 406mVpp. Not mixed, Loaded
- Output of 45.4Vpp (Probe is 1:10)
- With the load of 4 transducers we get the same gain and behavior from the amplifier with a single tone.



Amplifier Functionality - Loaded

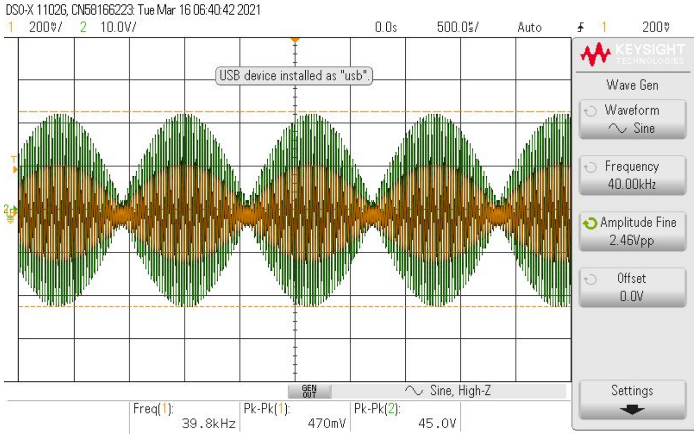
- Now mixed and loaded
- Same properties exhibited from the mixed signal, we have no issues amplifying the 40kHz carrier modulated with audio.
- No hysteresis in this subsystem



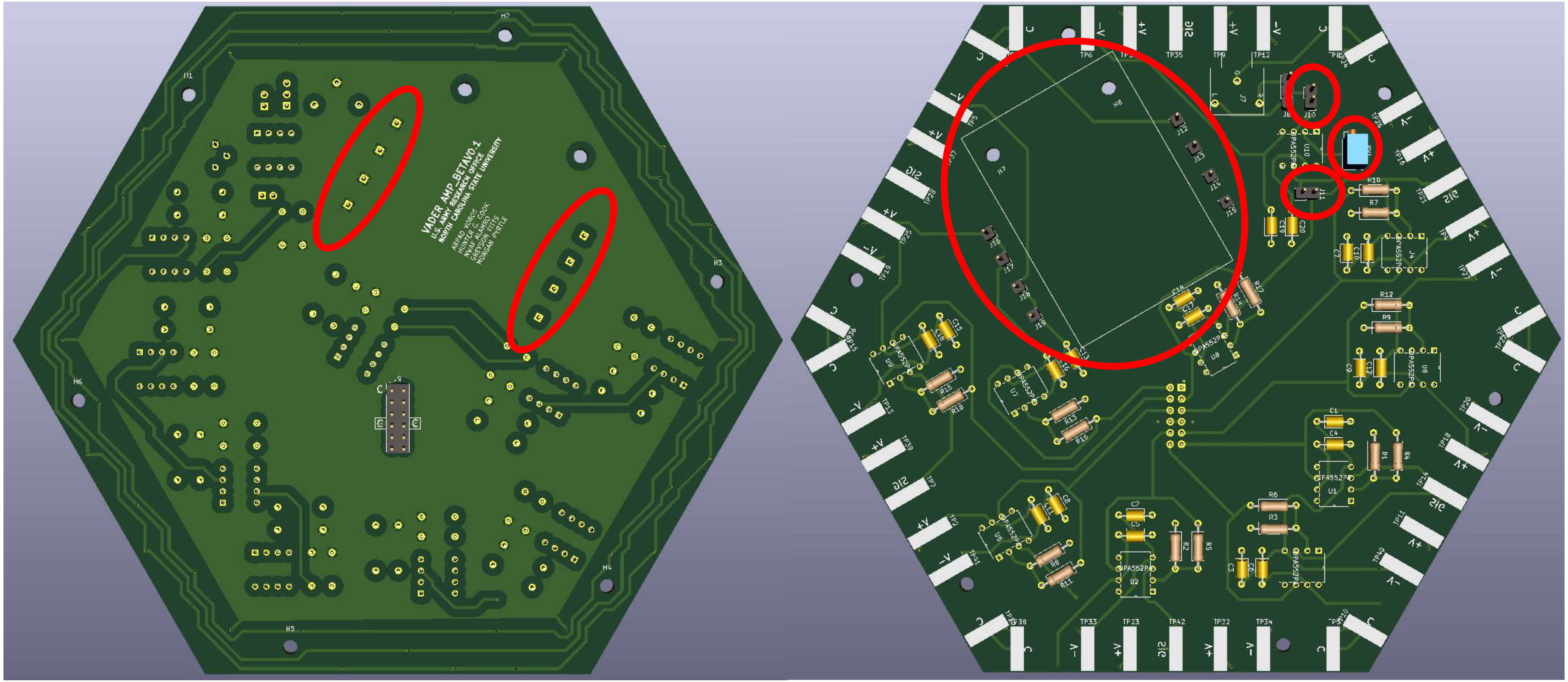


Amplifier Functionality - Hysteresis

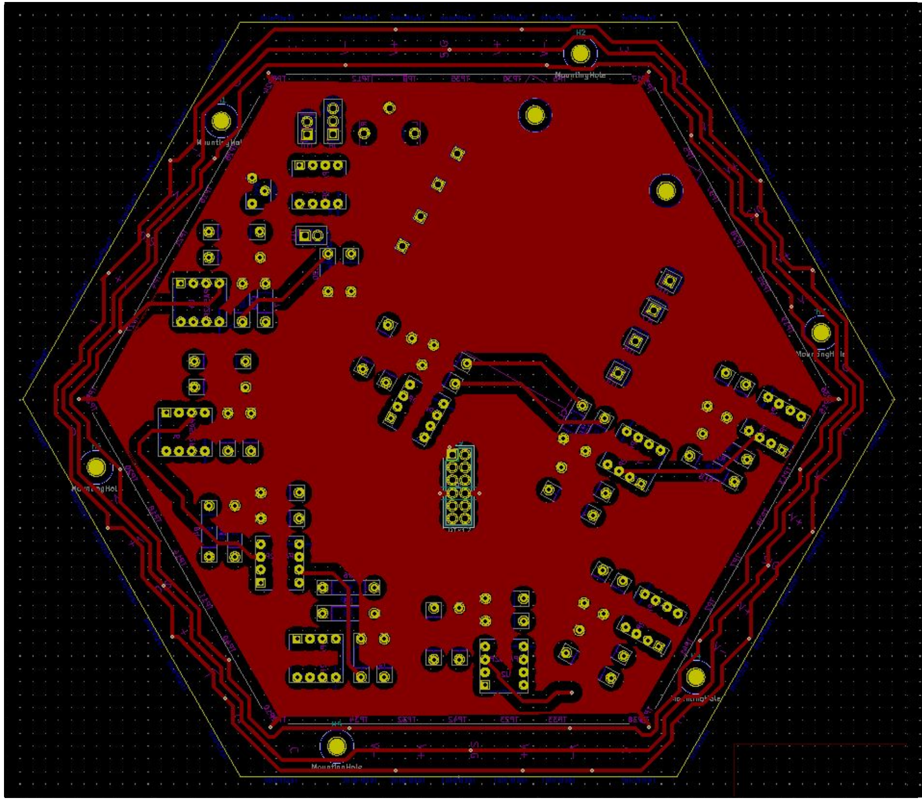
- At Alpha we noticed that there was hysteresis in the system.
- Removing the transducers had no effect, so I decided to test a voltage buffer, different biasing configs, and different loads as the signal source the input to the amps (the mixer).
- Hysteresis was occurring at the output of the mixer.
- Due to time limitations of finding a new mixer and making a new circuit, It was decided that we would modulate the audio digitally (in MATLAB) and input that directly into the amp.
- Eliminating the need for a mixer board.
- More on that after discussing the amp PCB

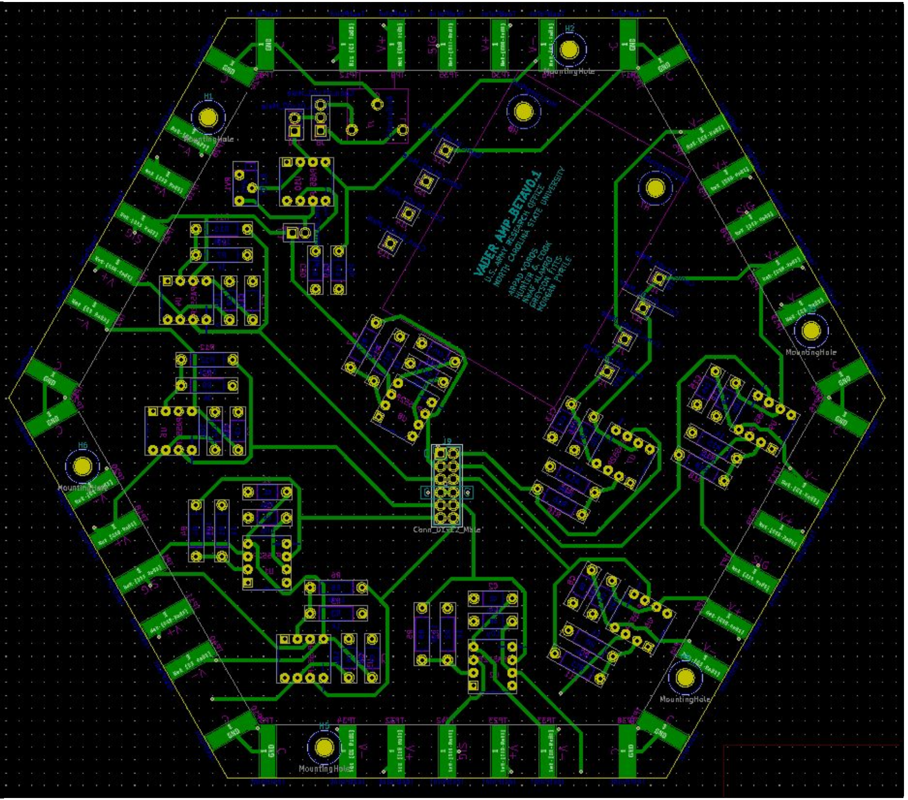


Amplifier Functionality - Revised Amp PCB



Amplifier Functionality - Revised Amp PCB





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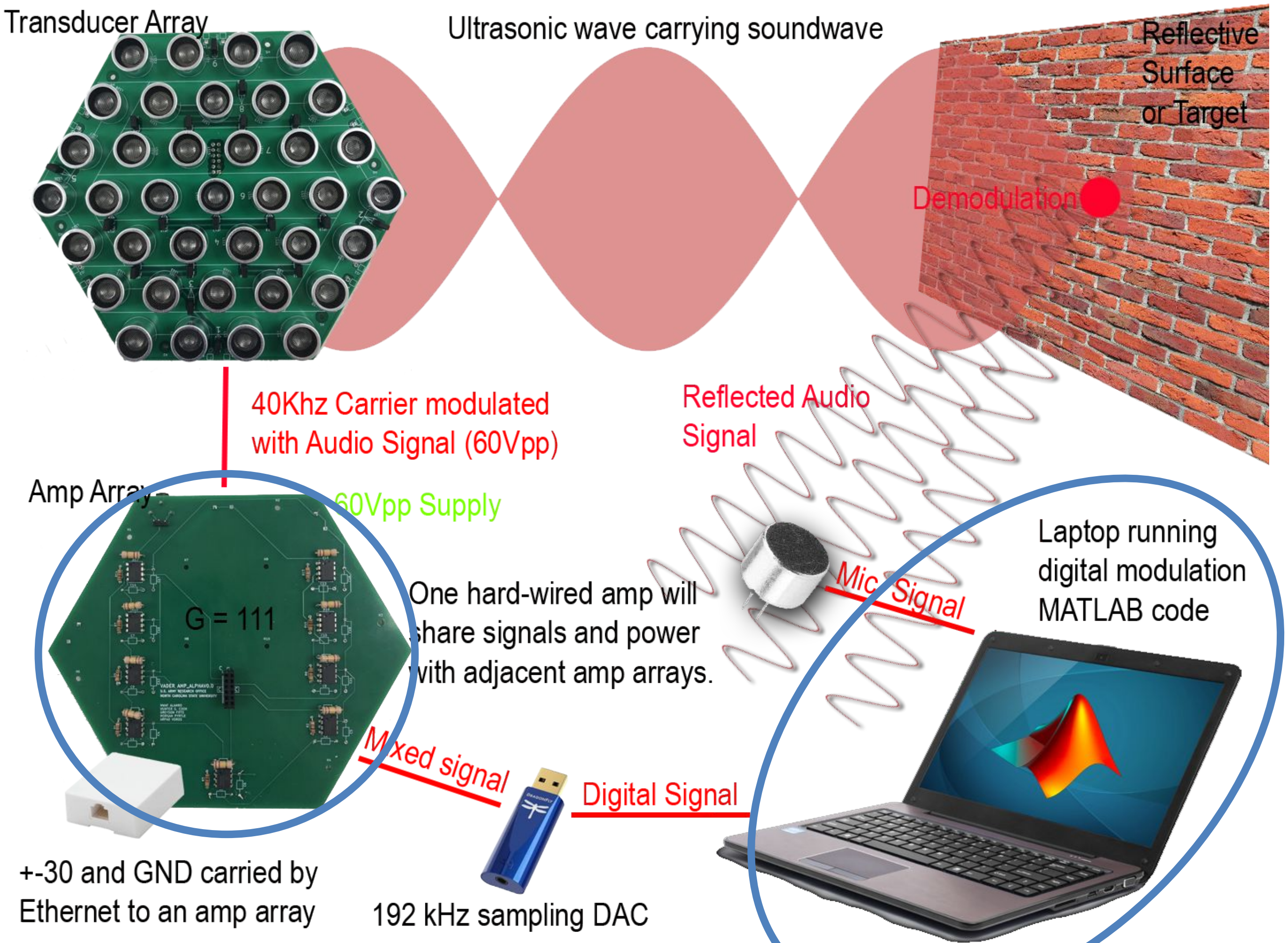
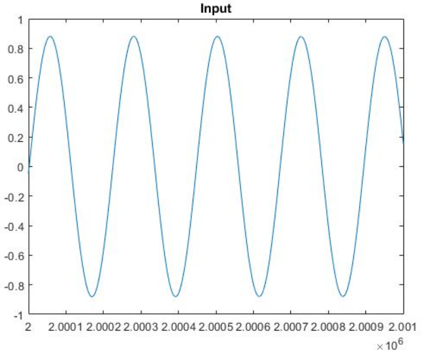
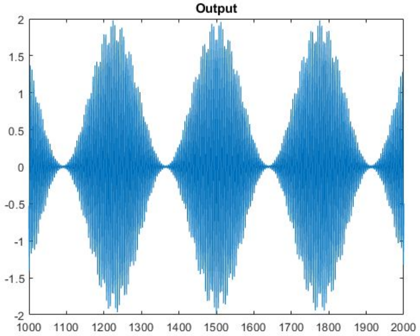
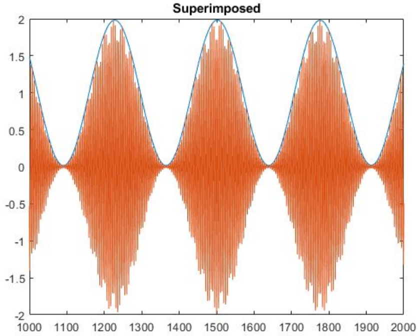


Subsystem - Digital Modulation

- Using MATLAB we are able to modulate a WAV file (containing audio) onto a 40kHz carrier and export it or play it out of a connected DAC.
- Since we need a 40kHz carrier it is required to sample at or above the nyquist rate. Sampling at 96kHz gives us 8kHz of range (96k/2 - 40k) (This was verified with a 96kHz DAC), where 192kHz would give us as much range as needed for audio. Thus we need a DAC that has a sample rate of 192kHz.
- Simply put, in MATLAB we resample the input audio at 192kHz, modulate the carrier with the signal, then output at 192kHz.
- Standard amplitude modulation below (with mod. index = 1), Arpad will discuss different techniques we found and how this change is going to be useful.







Digital Modulation Techniques

Distortion Analysis and Reduction for the Parametric Array

Ee-Leng Tan[1], Woon-Seng Gan[1], PeiFeng Ji[2] and Jun Yang[2]

[1] Nanyang Technological University, School of Electrical & Electronic Engineering, 639798, Singapore
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[2] Institute of Acoustic, Chinese Academy of Sciences, Beijing, China
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2008

Article

Experimental Evaluation of Distortion in Amplitude Modulation Techniques for Parametric Loudspeakers

Ricardo San Martín [1,*], Pablo Tello [1], Ana Valencia [1] and Asier Marzo [2]

1 Acoustics Group, Institute for Advanced Materials and Mathematics—INAMAT, Universidad Pública de Navarra, 31006 Pamplona, Spain; tello.106735@e.unavarra.es (P.T.); ana.valencia@unavarra.es (A.V.)

2 UpnaLab, Institute of Smart Cities—ISC, Universidad Pública de Navarra, 31006 Pamplona, Spain; asier.marzo@unavarra.es

2020

Modified Amplitude Modulation
(using Orthogonal Carrier)

$$P_1(t) = P_0 e^{-\alpha x} \left\{ \begin{array}{l} \left[1 + m g(t)\right] \sin \omega_c t \\ + \left[1 - \frac{1}{2} m^2 g^2(t) - \frac{1}{8} m^4 g^4(t)\right] \cos \omega_c t \end{array} \right\}.$$

$$p_{MAM1}(t) = [1 + ms(t)] sin(\omega_c t) + \left[1 - \frac{1}{2} m^2 s^2(t)\right] cos(\omega_c t),$$

$$p_{MAM2}(t) = [1 + ms(t)] sin(\omega_c t) + \left[1 - \frac{1}{2} m^2 s^2(t) - \frac{1}{8} m^4 s^4(t)\right] cos(\omega_c t).$$

Digital Modulation Techniques - Cont.

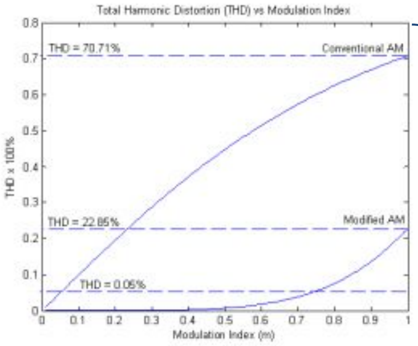


Figure 6 Comparison of THD between improved AM and conventional AM

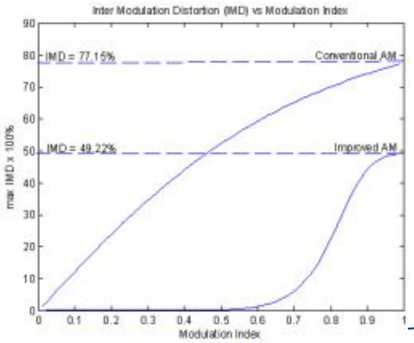
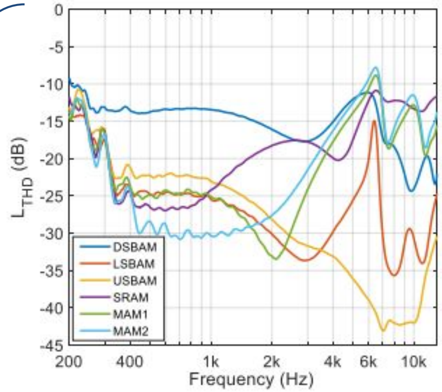
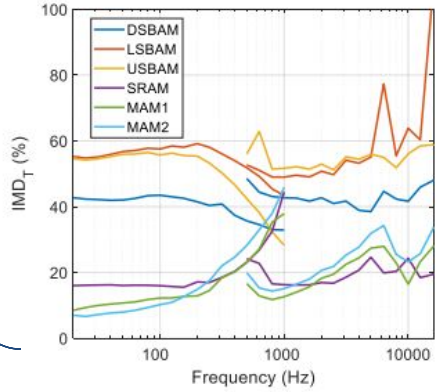


Figure 7 Comparison of IMD between improved AM and conventional AM

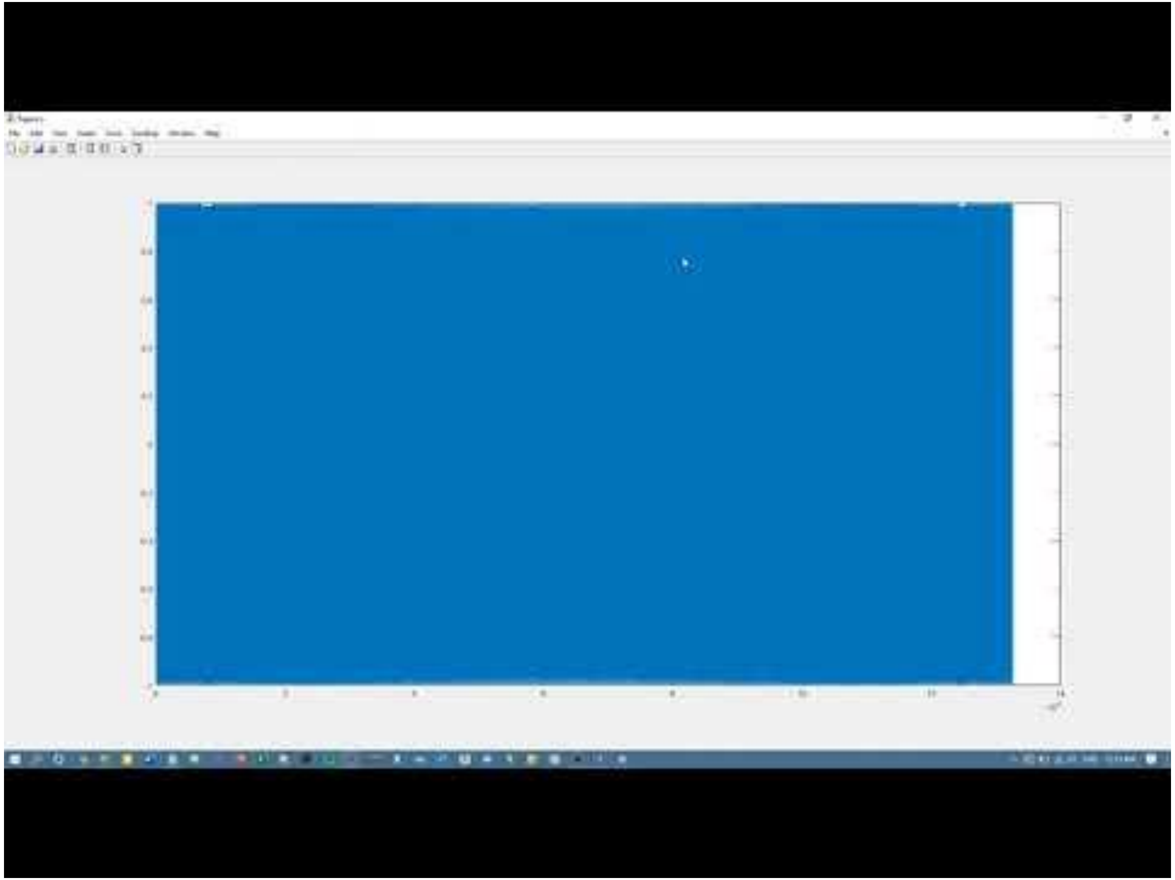
average from 0→ 20kHz sweep

m = 1





Digital Modulation GUI



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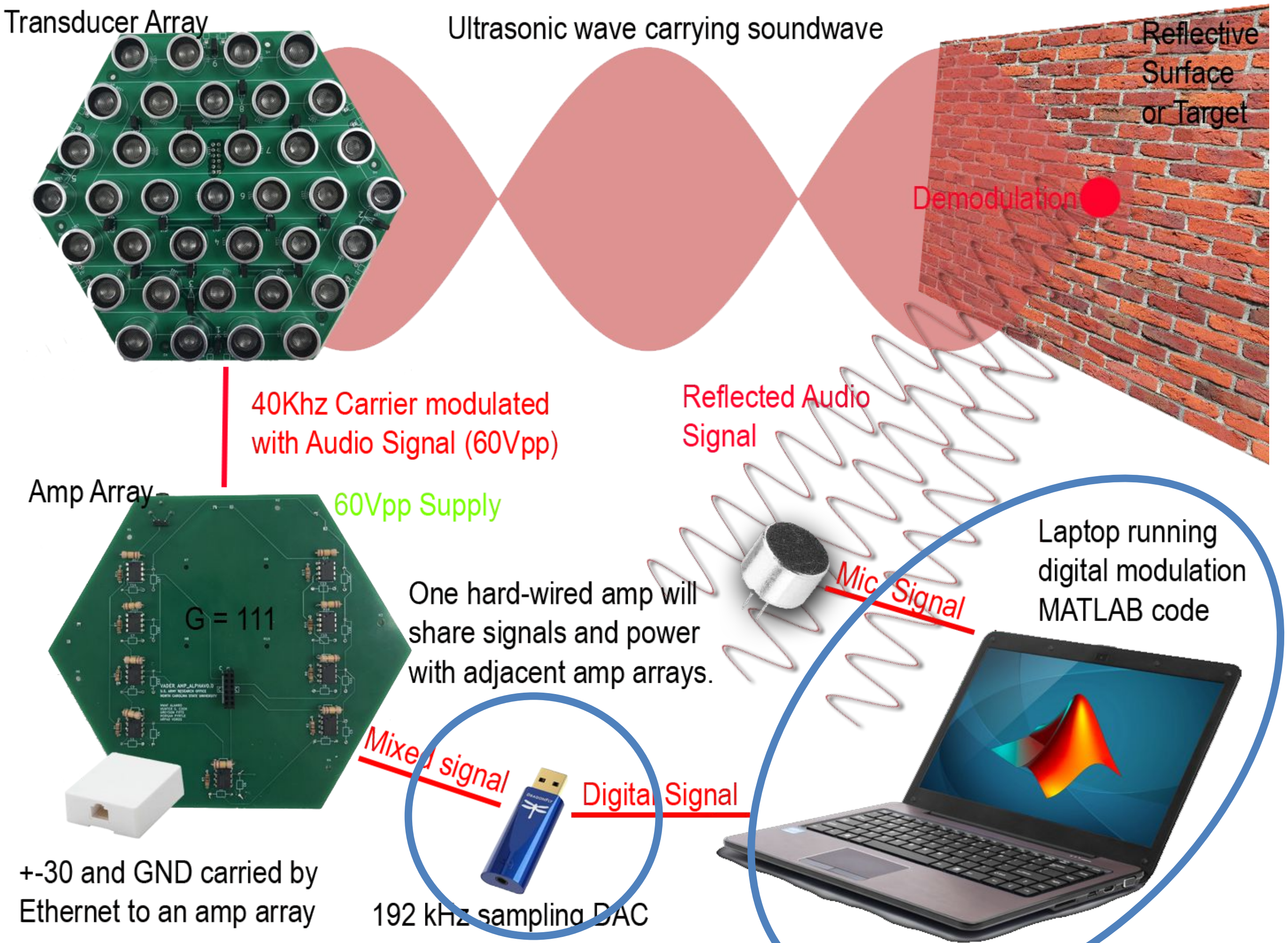
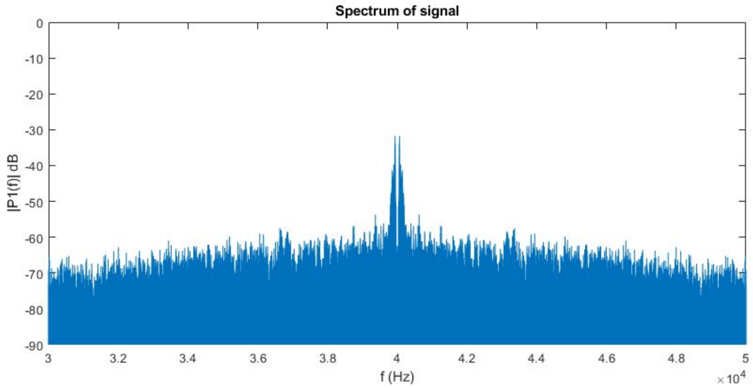
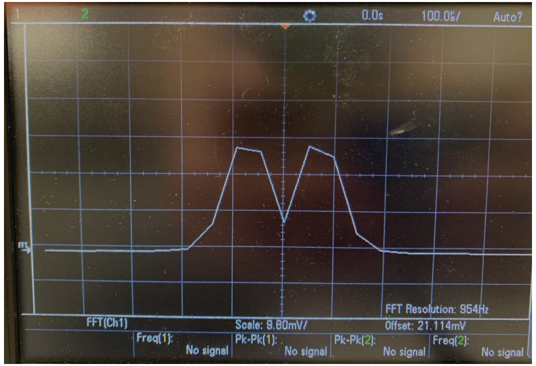


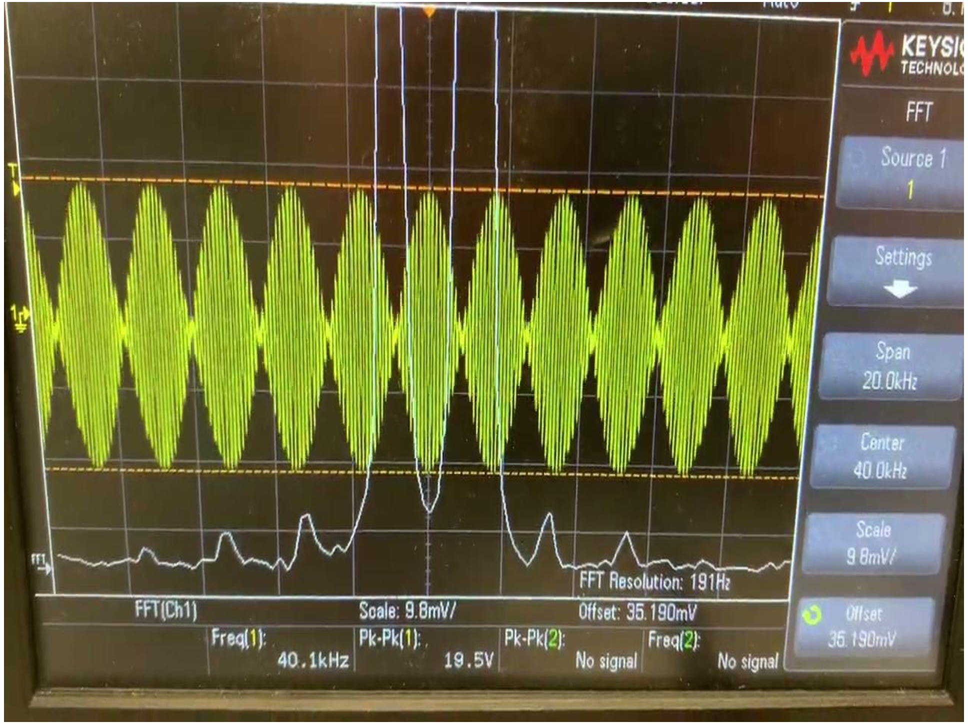
Subsystem - New Dac (w/ No Sound Board)

- Arpad and I reviewed sound board documentation to determine it can only do 50kHz
- As previously stated we needed a DAC that could sample at 192kHZ, A USB solution and a laptop was the fastest and easiest way to accomplish this.
- FFT of entire signal in MATLAB vs actual DAC instantaneous spectrum:





Amp + DAC = No Hysteresis (Level Gain Freq. Sweep)



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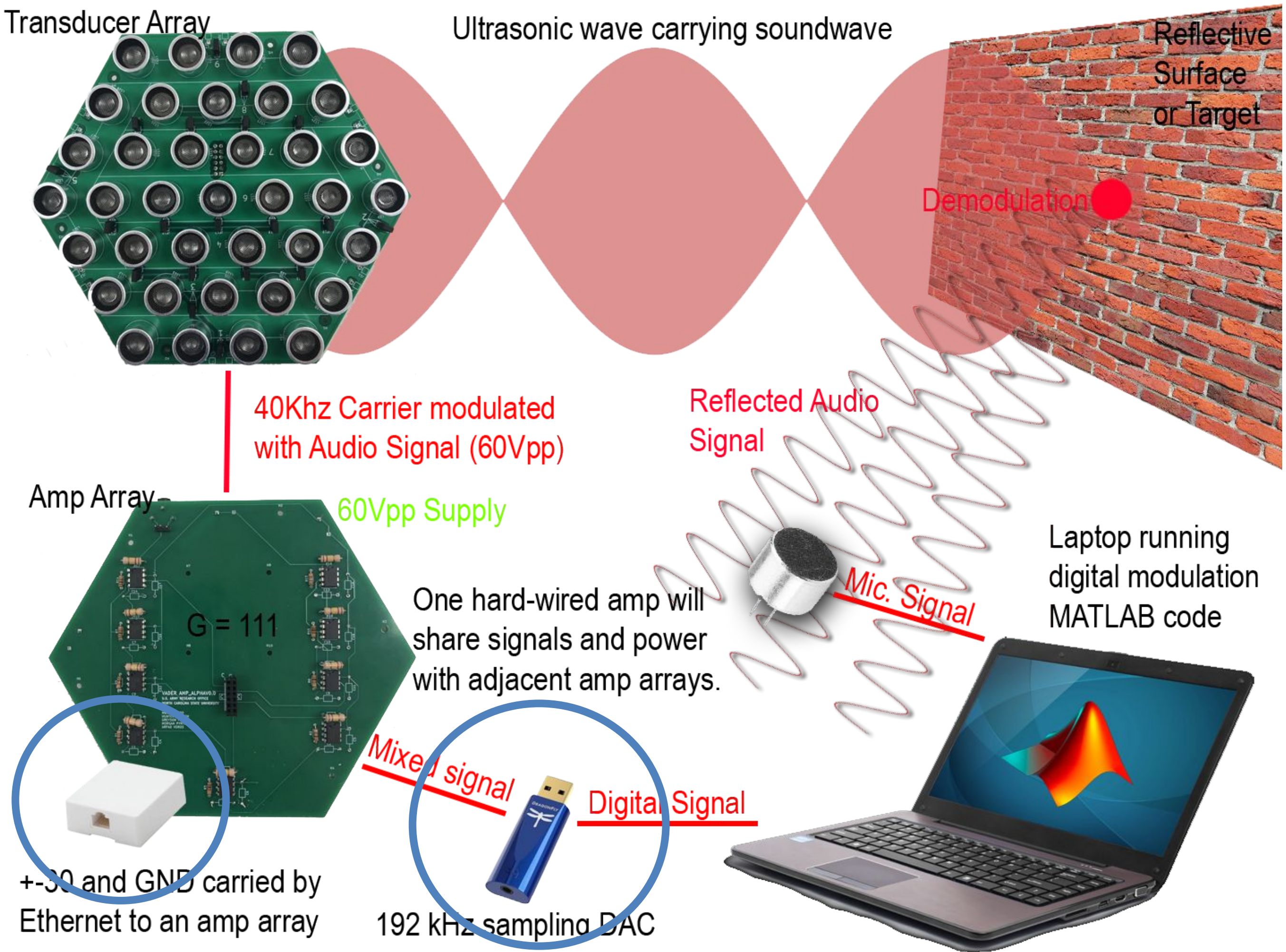
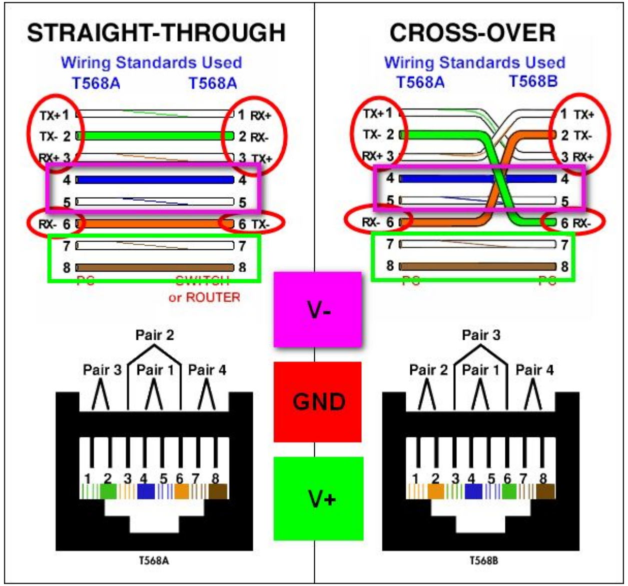
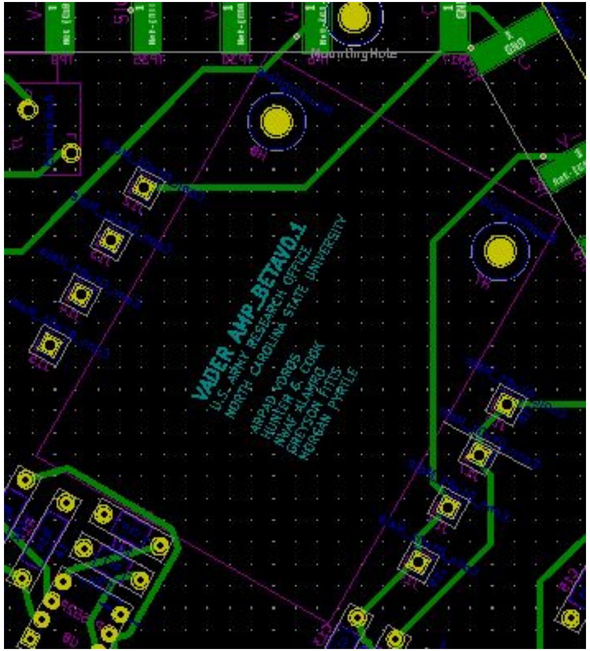


Subsystem - Ethernet Power (On Amp Board)

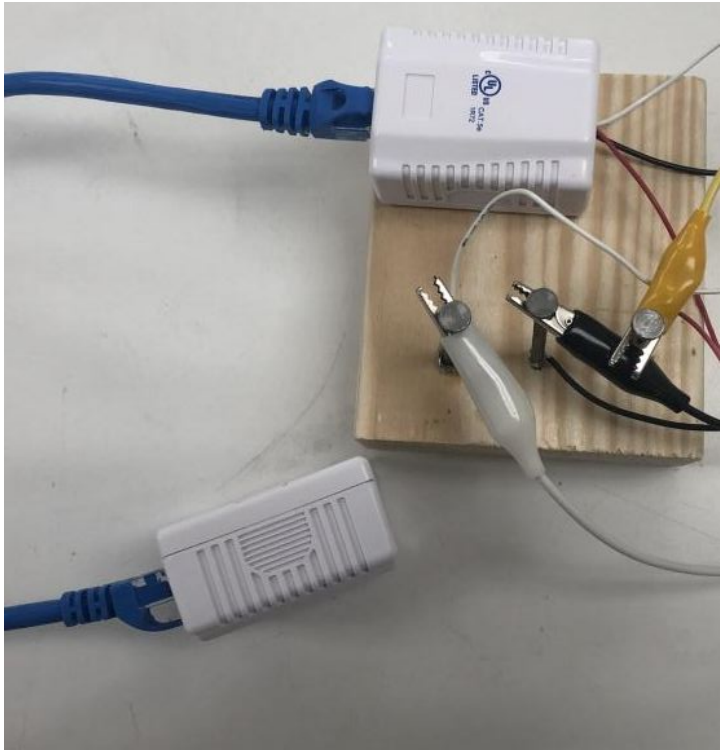
- Benefits of including ethernet cable is to power is to simply extend the range of our PSU. We have not tested the power draw & limitations of this system yet
- PCBs have NOT arrived yet, so cannot show it on the revised amplifier board





Subsystem - Ethernet Power (From PSU)

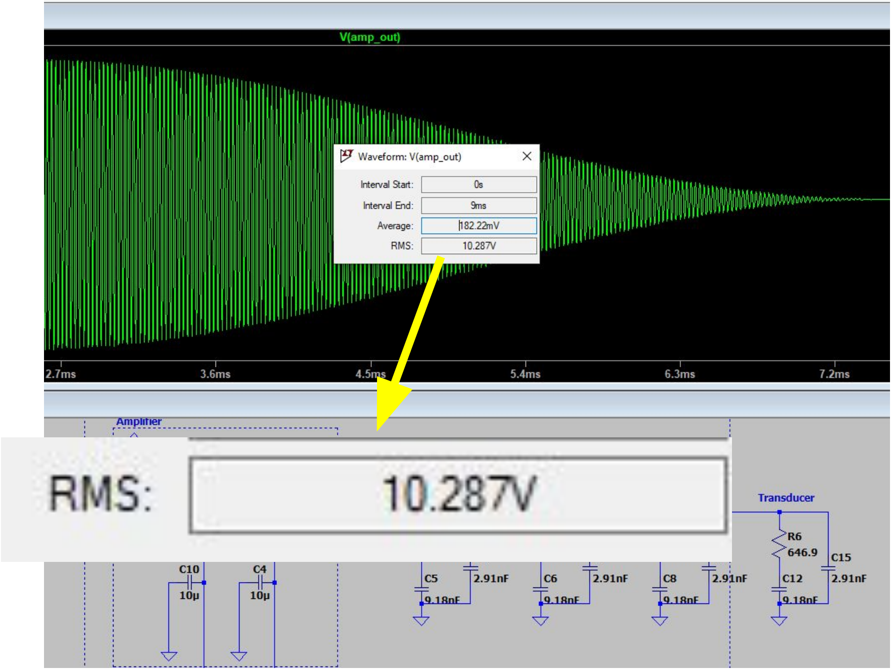
- Created a 3 terminal board that a bench-top power supply can easily hook too and routes it directly to an Ethernet port.
- This gives us much more ease of use in our system, allowed the power end to stay setup and for the amp board to simply plug in and move around.
- Not elegant but effective. Ideally this would be its own little PCB that sits next to our power solution.



Driving Voltage

- As mentioned before, trying to achieve 45Vpp (±22.5V)
- This is due to the fact that our transducers were said to be nominal at 10Vrms. Calculation shown in CDR that given modulated signal, we need 45Vpp to achieve 10Vrms
- Is confirmed in screenshot to the right

- But is 10Vrms really nominal? After looking at datasheet, we are unsure
- Maximum input can go as high as 150Vpp
- Anything greater than 60Vpp is loud/non-directional, but this statement could be biased because we tested in an enclosed space with lots of reflections
    - Test outside
    - Test in anechoic chamber
- Can and probably will go higher for Beta



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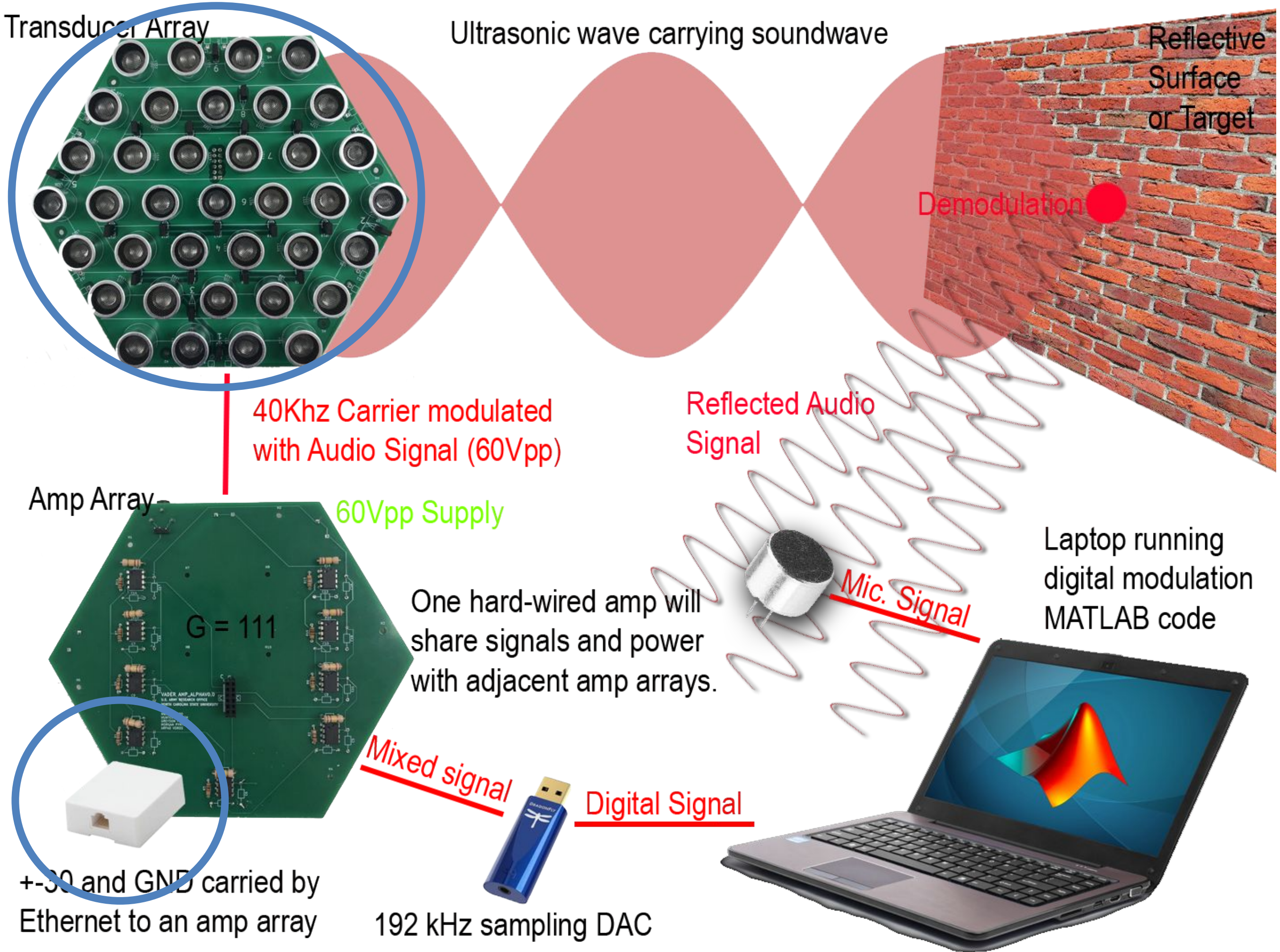
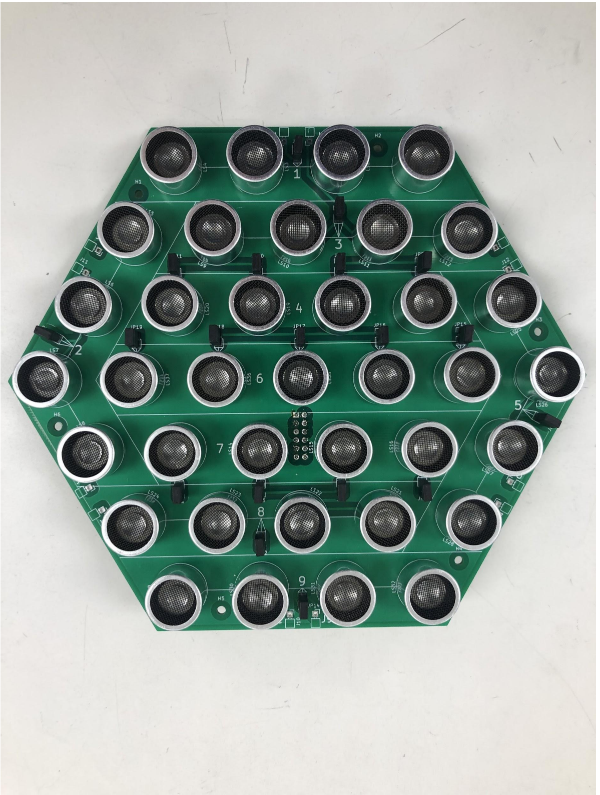
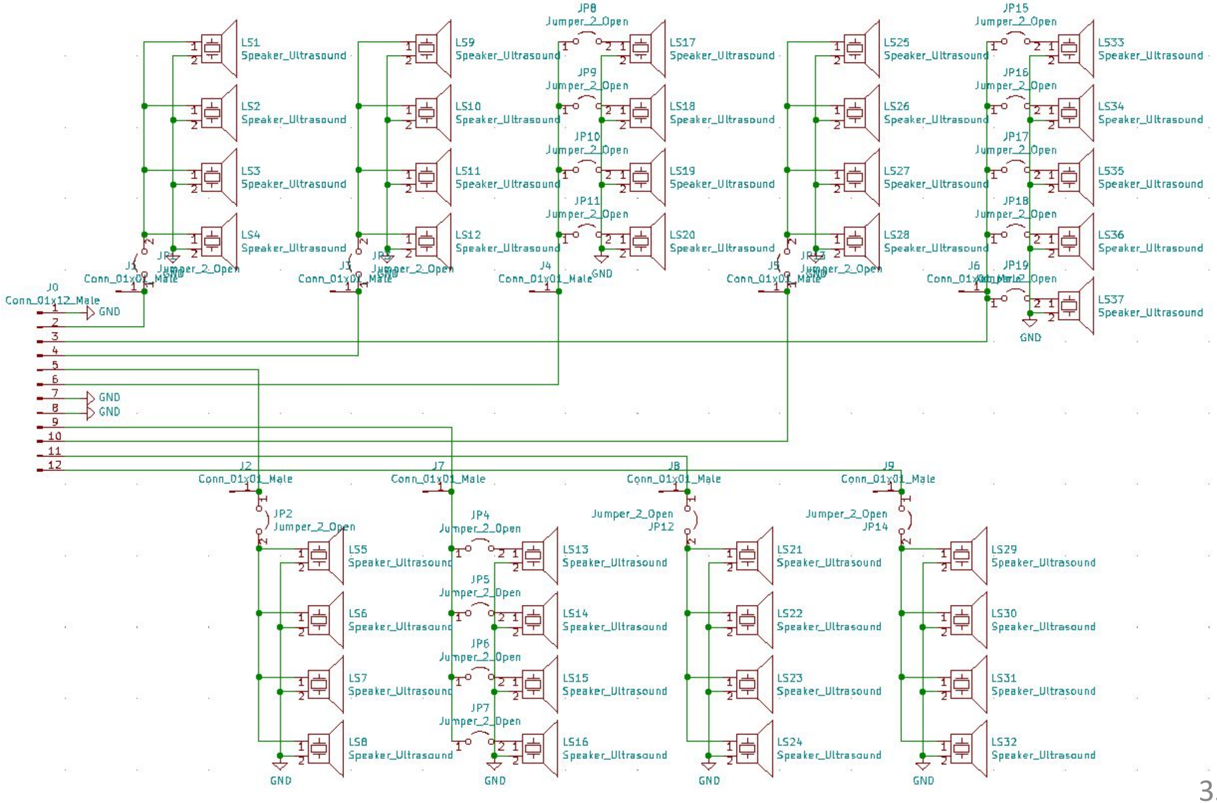


Subsystem - Transducers





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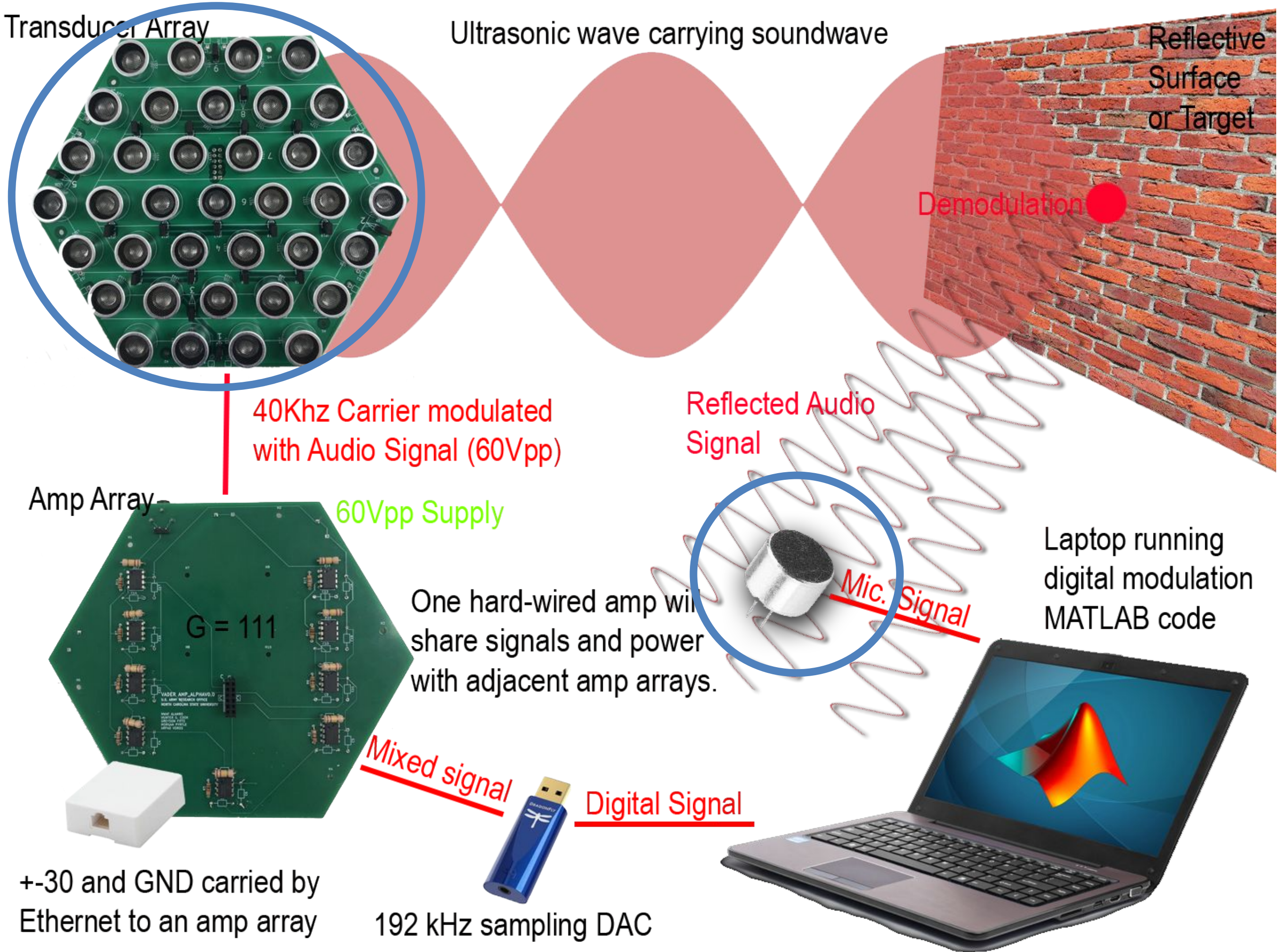
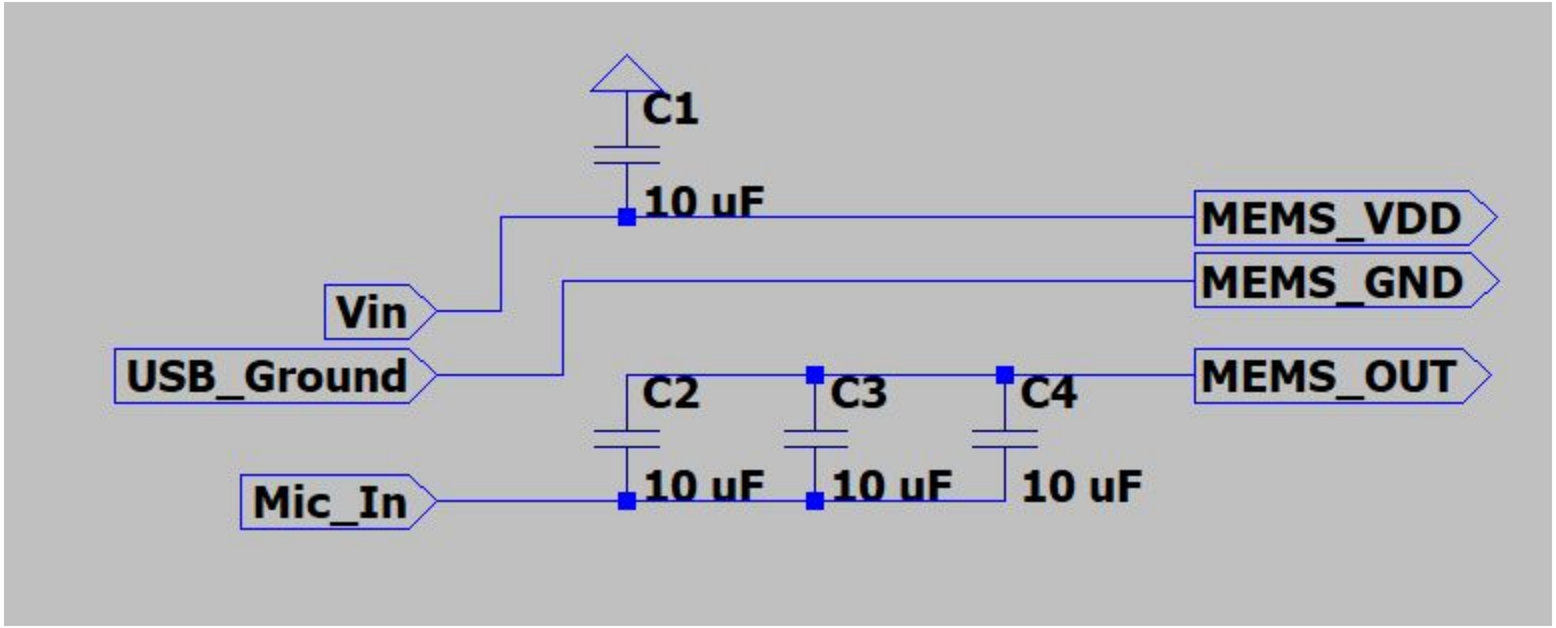
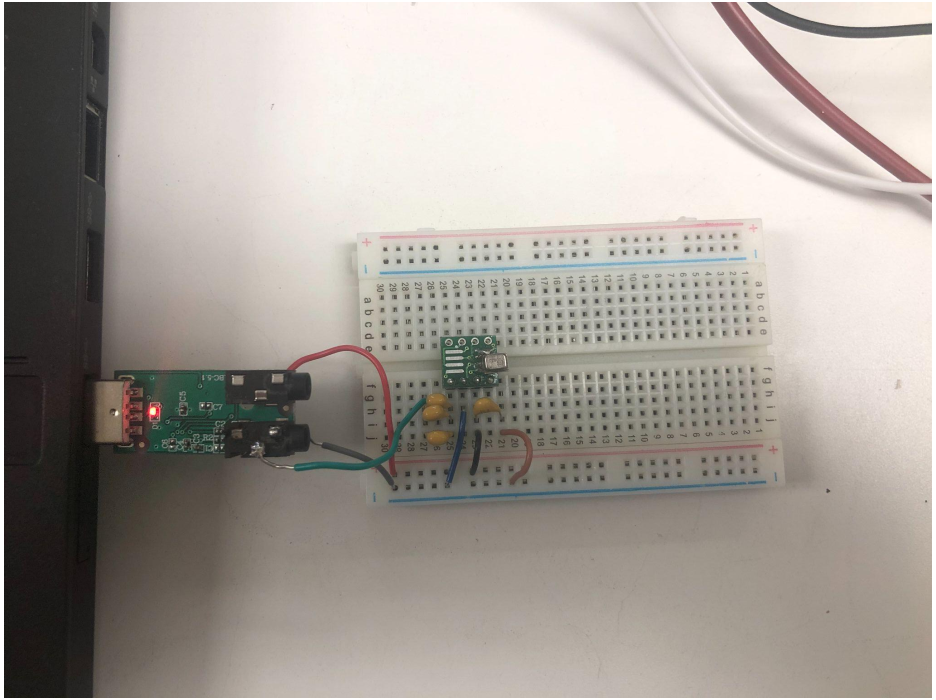


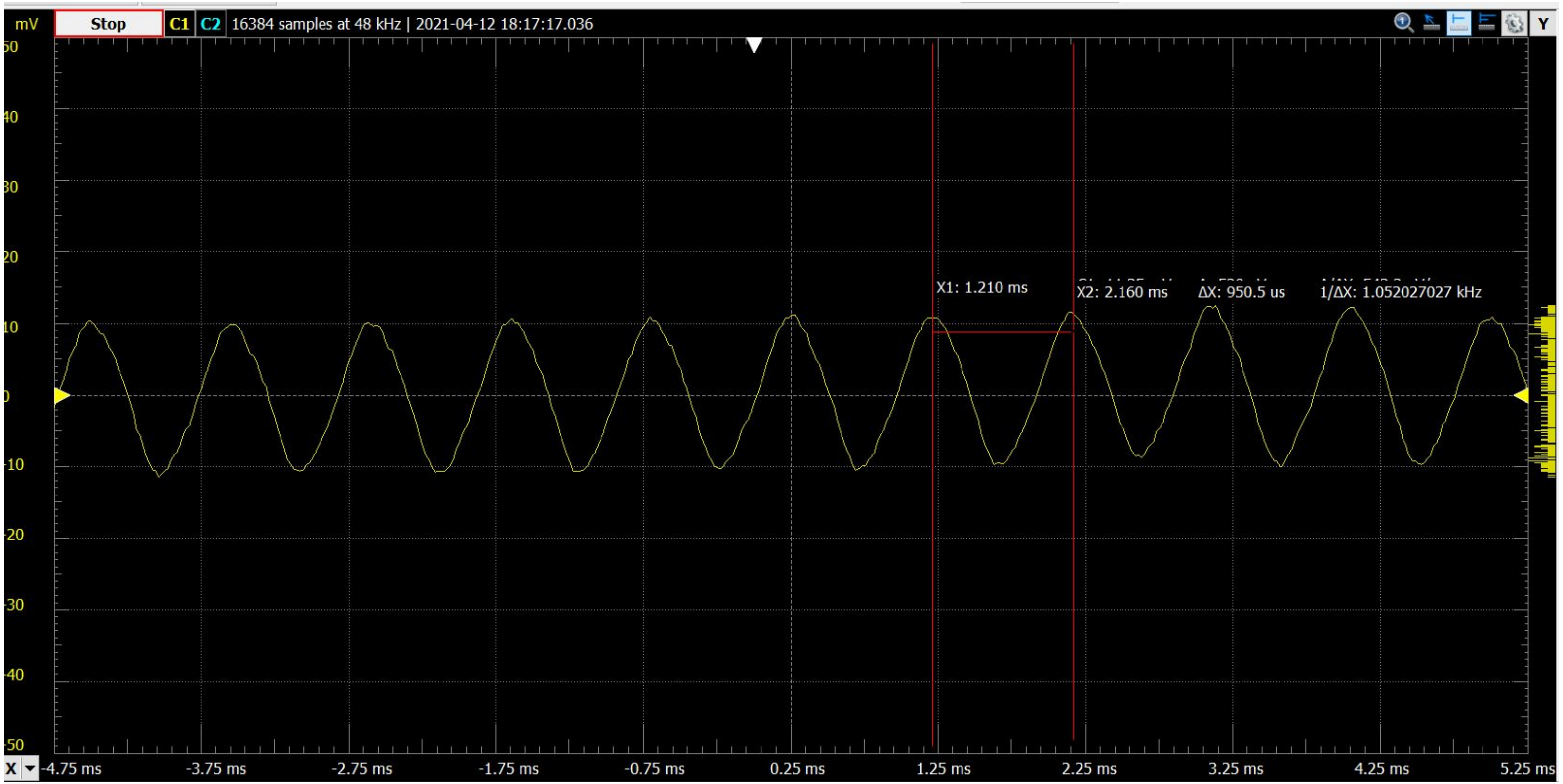
Subsystem - Audio Detection



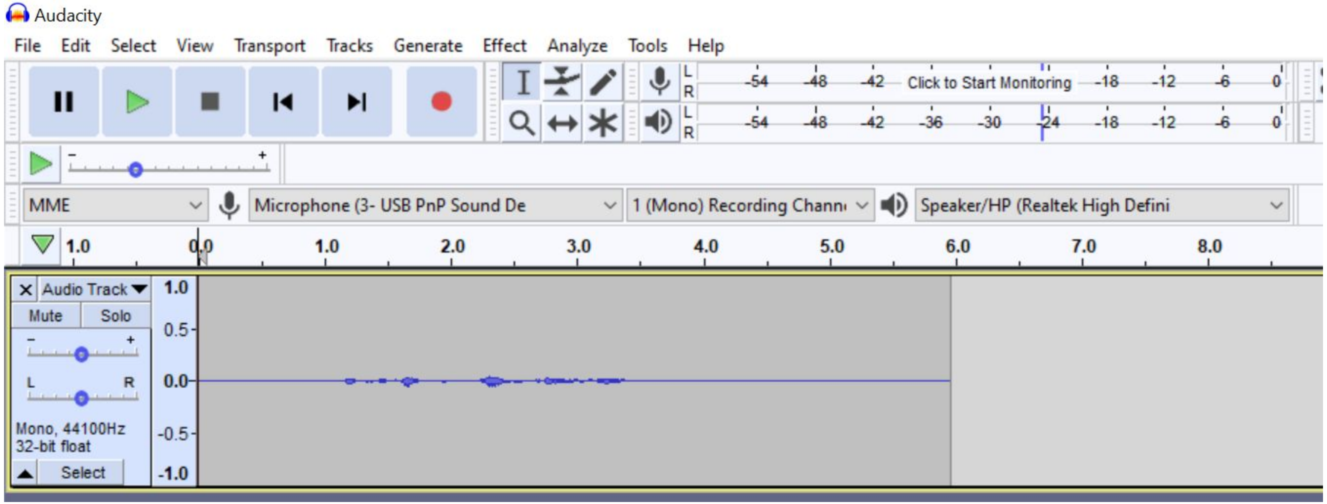
Subsystem - Audio Detection



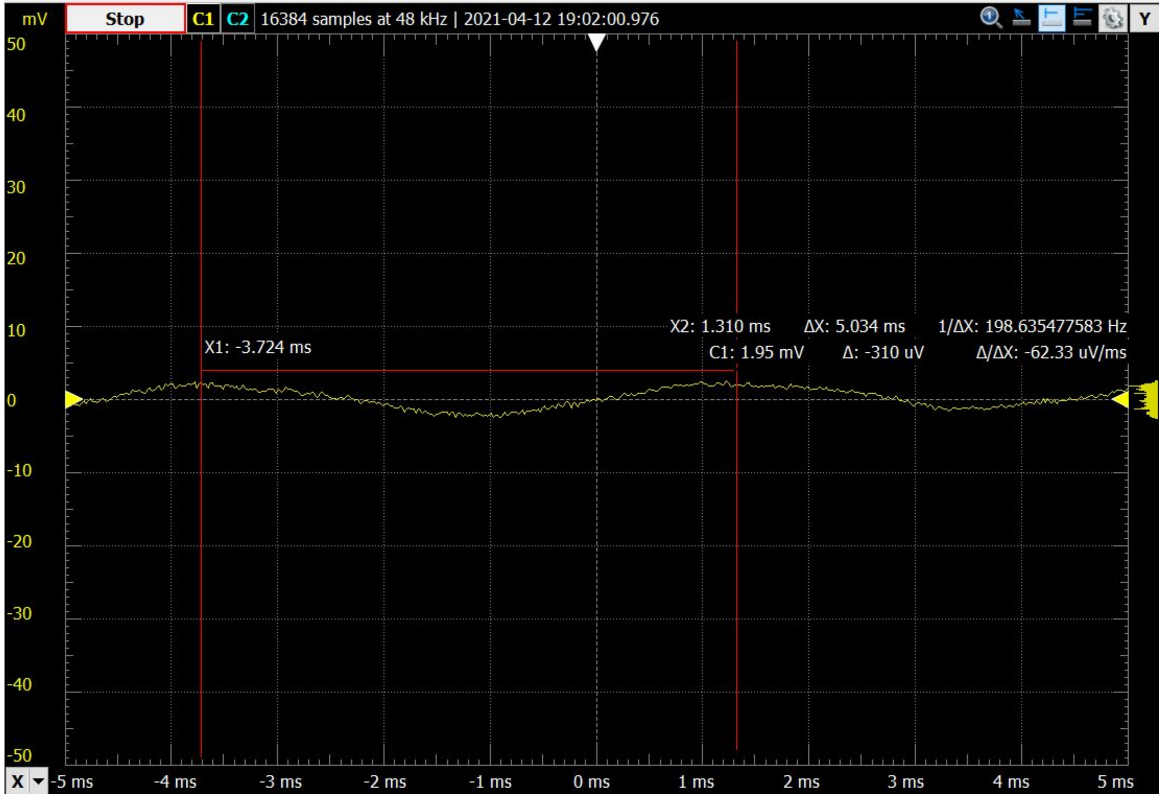
Microphone Functionality: Removal of Oscillations

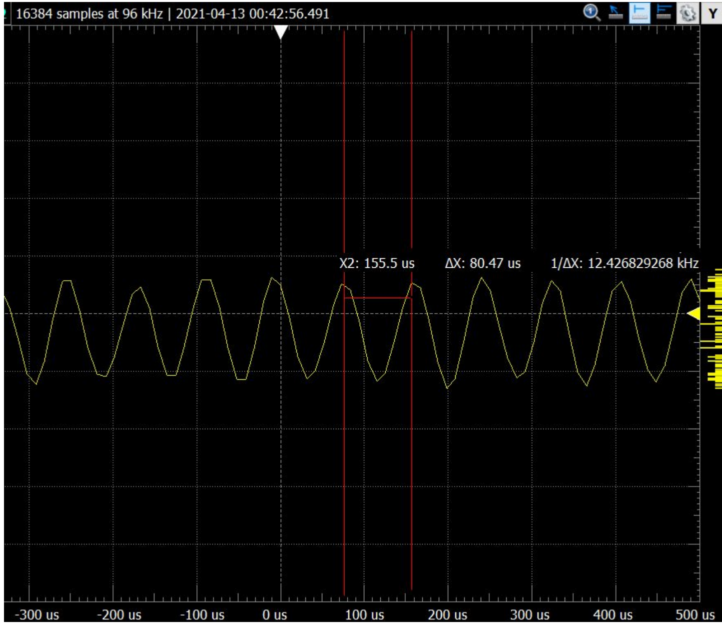


Microphone Recording via Audacity



Upper and Lower Frequencies





Microphone Functionality

Product Requirements:

For Alpha:

- Requirement 1.1 - (Detect) Length from Source
  - Pass: **Sound is present**
  - Fail: Sound is not present

For Beta:

- Requirement 1.2? - Multidirectionality using splitting sound waves
  - **Pass:** Microphones pick up sound at anticipated location
  - Fail: Microphones do not pick up sound at anticipated location

-Generate stand-alone circuit (independent of AD2) which can be recognized as a microphone by a PC and capture information about analog signal through AUX/USB **(Done)**

-Fix issue with wave oscillation **(Done)**

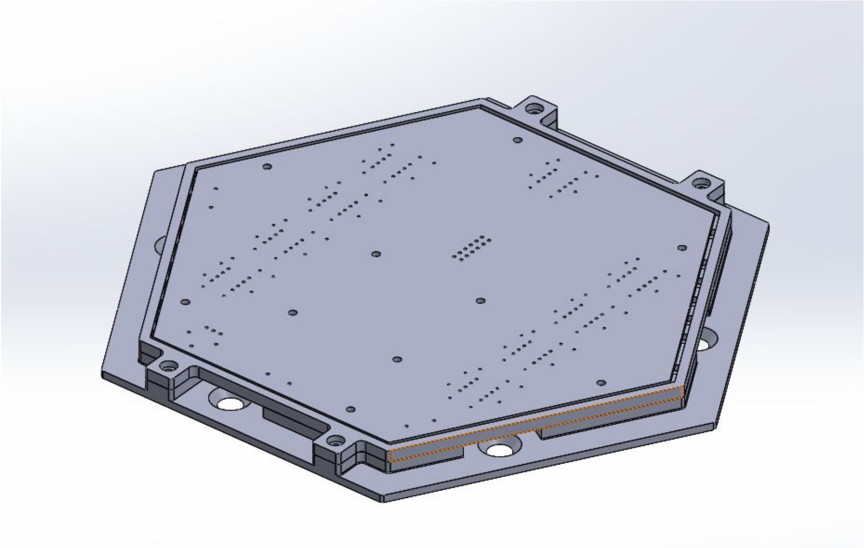
For Design Day:

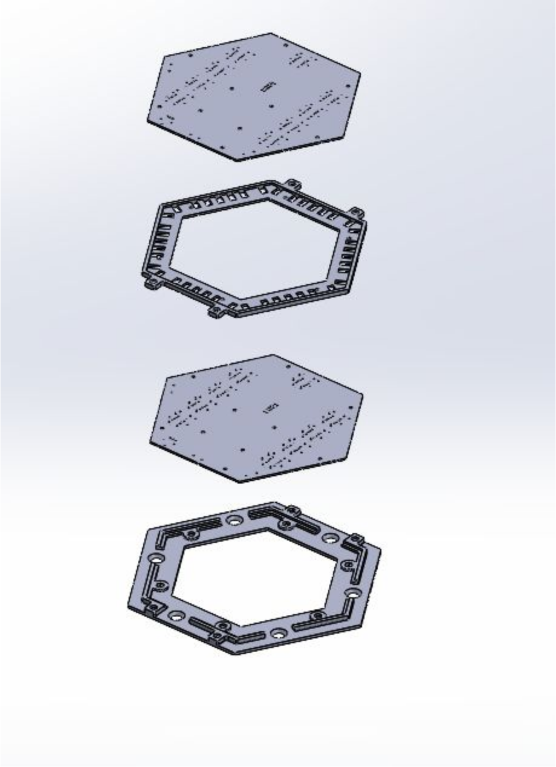
-Generate Radiation Pattern for final array (MATLAB code is ready)

-Redo soldering in an attempt to minimize noise.

- Make circuit more portable

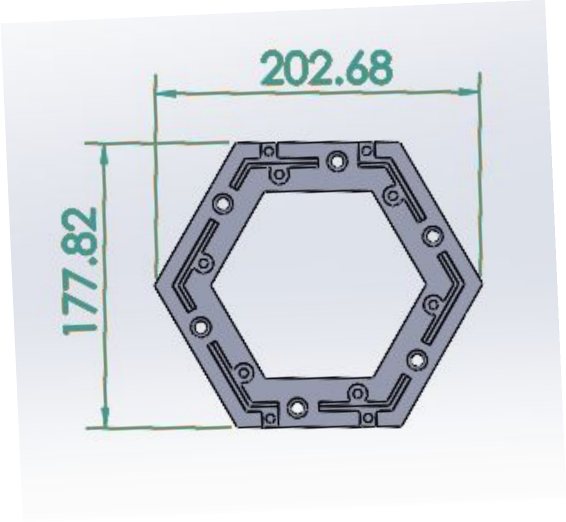
Subsystem - PCB Enclosure

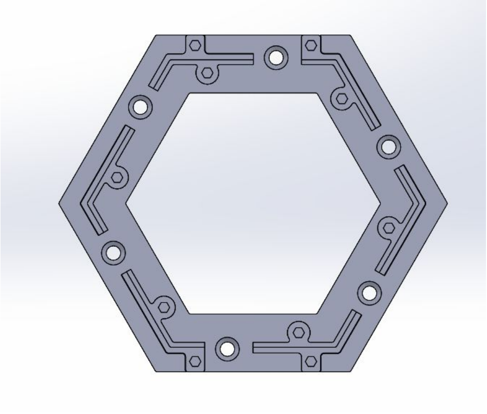


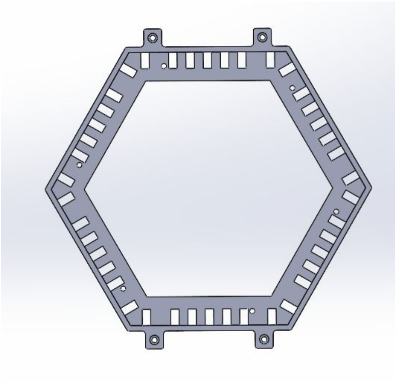


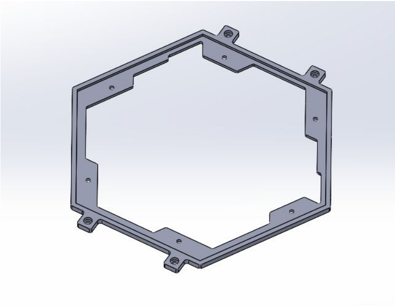
Subsystem - PCB Enclosure

Updated PCB Enclosure Model:

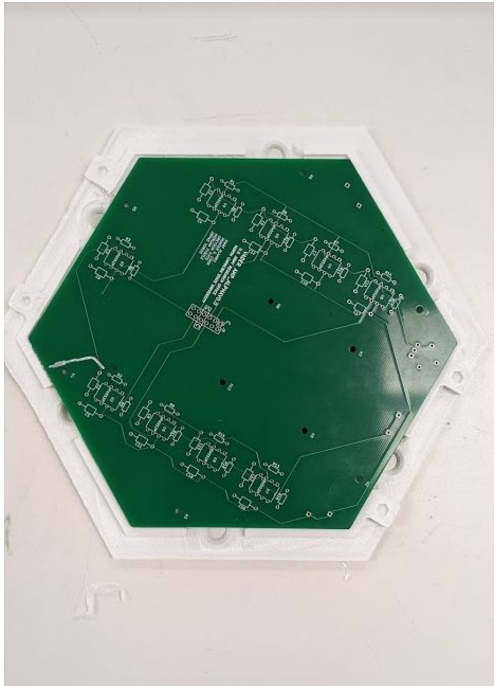


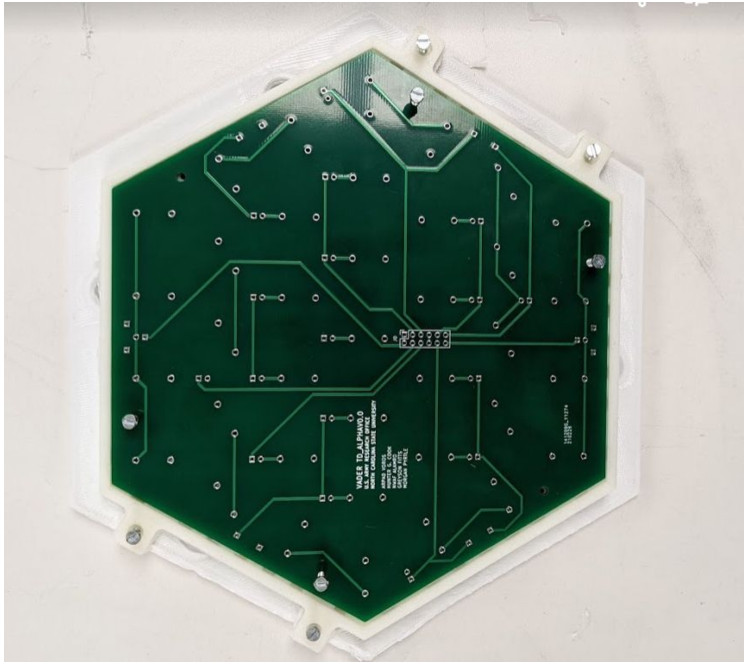




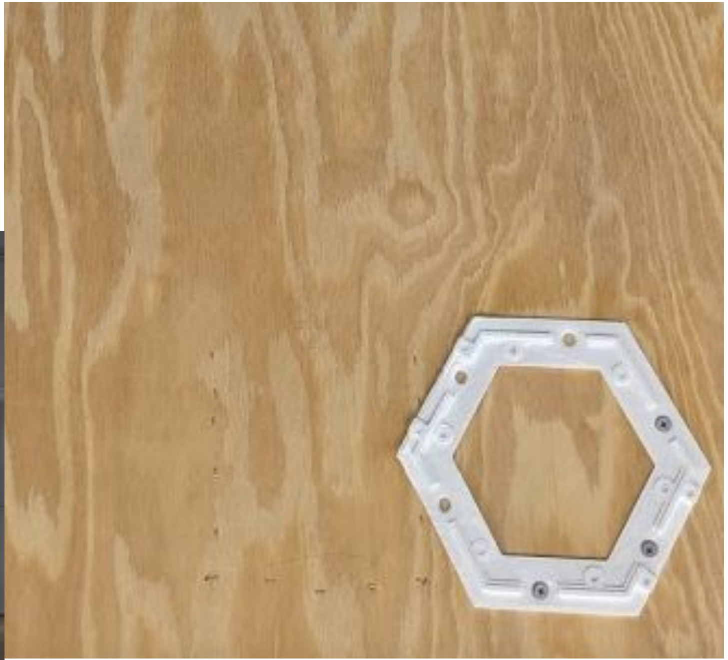


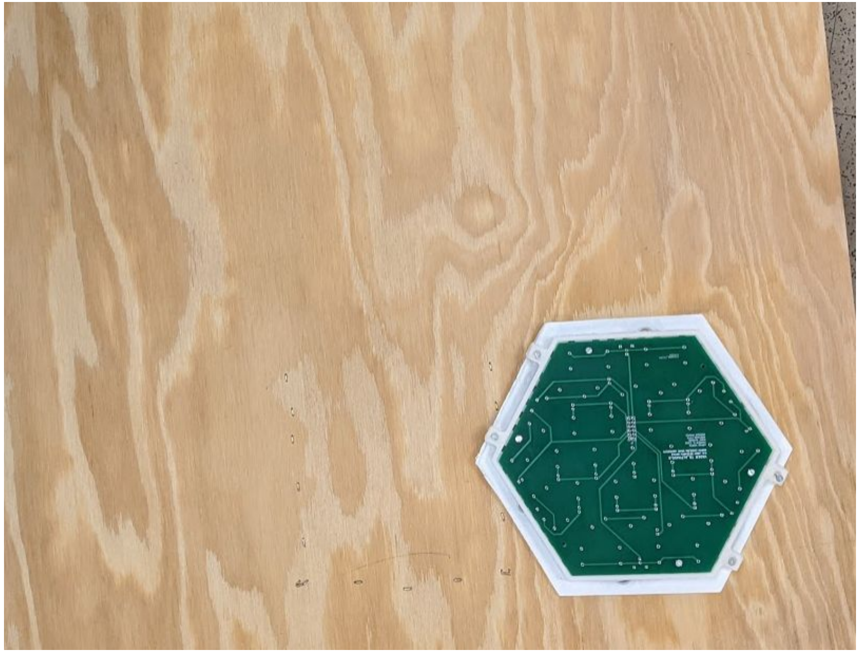
Subsystem - PCB Enclosure





Subsystem - Housing





Subsystem - Housing

Major Changes Since Alpha:

- Accommodates for putting the PCBs much closer together, which allows for better modular testing
- Using ABS as the filament for the PCB enclosure, which allows for more outdoor use and flexibility
- Wheels added to the test bench stand
- Integration of the PCB enclosure and the test bench stand

Modular Enclosure/Housing Functionality

Product Requirements Achieved:

- The PCBs do not fall out or break (3.3)
- Array housing is not brittle (4.3)
- The PCB enclosure successfully connects to larger structure.
- The larger structure is physically durable.
- The larger structure is easy to assemble; it requires less than 5 major steps for general assembly.
- The current layout of it makes it easier to set up and transport.

Design Day Plan for Housing:

- Interface an entire array pattern
- Make more modules to support array pattern

Design Day

- Gather data from system
  - Multiple modulation techniques + play around with parameters to see what sounds loud, clear, and not distorted
  - Once done and sound recorded, implement predistortion code into the GUI and test to see if sounds better
  - Complete radiation patterns in indoor setting, outdoor setting, and anechoic chamber
  - Carrier leakage
- Video of all full system function (Need those darn PCBs!)

that's all