Acoustical Response of Traditional Speakers to Single-Frequency Sine Waves

Jordan Leek

Senior Research Final Report

Mentor: Dr. Mark Hinders
Department of Applied Science

Department of Physics
College of William and Mary
Email: jordantleek@gmail.com

I. Introduction

In both the animal and the human world, the ability to hear and communicate is vital for survival. Vocal cues are used for various purposes, including reproductive, defensive, and informational, regarding food or otherwise. These cues are necessary to ensure the species' survival in their chosen area. Noise present in the same area and within the same range of communication of animals has been shown to affect their mating habits, eating habits, and even their presence altogether [1].

In many locations, such as airports, farms, and places of business, birds cause physical and financial damage. From 1990 to 2012, $639 million in costs due to wildlife strikes were reported to the Federal Aviation Administration (FAA) [2]. In 2013, $189 million in bird damage to agriculture was recorded for apple, blueberry, cherry, and wine-grape growers in California, Michigan, New York, Oregon, and Washington [3]. The financial costs due to wildlife are climbing, and traditionally used methods to deter wildlife have little to no effect over time [4].

Birds communicate within a characteristic frequency band dependent upon their species. When noise at the same frequency band is present, birds tend to leave due to the noise's impact upon on their communication. By isolating the frequencies that a species of bird communicates in, and playing noise at this frequency band over a specific area, birds are effectively displaced, both in the short- and long-term [5, 6, 7, 8, 9, 10]. The downside to projecting this noise over a large area is that it is within the human range of hearing and therefore it adds to noise pollution. For this method to be viable in highly populated areas, a means of spatial noise control must be developed.

In an effort to minimize the noise pollution of this signal in undesired areas, and better focus the pink noise signal, a multifrequency acoustic mirror will be designed, constructed, and tested with various speaker designs at the focal point. To better understand the amount of noise pollution that currently occurs and to optimize the design of the acoustic mirror, we will first characterize the acoustic spread of each of the four speakers using ten 500 Hz to 10kHz sine wave signals.

II. METHODS

Currently in use with the pink noise signal are five types of speakers: two traditional, which emit sound over a wide angle (like a floodlight); two parametric, which are highly focused sources of sound (like a laser or powerful flashlight); and one "cube" speaker with a traditional speaker on each face. The speakers we will be analyzing are the Technomad Berlin 60/40, the Pyle 9.4" Indoor/Outdoor 70 Volt PA Horn Speaker, and the Midstream Technologies cube speaker, as seen in Figure 1. To characterize the angular spread of the speakers, we first decided on a location. The location best suited for these tests would be an open field a mile or so from any residential area due to the audible frequencies of the signal; however, gaining access to such an area is difficult and there are multiple locations on the College of William and Mary campus that suit the needs of the tests. Lake Matoaka Amphitheatre, an outdoor venue that can fit 2,000

a) b) c)

Figure 1: The three speakers used. a) Technomad 60/40, b) Pyle 9.4" Indoor/Outdoor 70 Volt PA Horn Speaker, and c) Midstream Technologies cube speaker.

Figure 2: Experimental setup including the first speaker model, the Technomad Berlin 60/40, with Recorder 2 and the laptop from which the audio file was played.

people, was chosen as the test site for its large amount of space (200 feet long), isolation from industrial noise (approx. 800 feet to the nearest street, with dense foliage in between), and high angle between the top of the Amphitheatre to the bottom (20 foot change in elevation from top to bottom of amphitheater) [11]. By placing the speaker at the center of the top of the Amphitheatre, facing towards the lake, and angled slightly downwards, we limit the amount of reflective surfaces that the signal coming out of the speakers encounters. The setup at the top of the Amphitheatre can be seen in Figure 2.

In order to characterize the acoustic spread of each speaker, three battery-powered Zoom H2 audio recorders were placed at incrementally increasing distances from and angles to the speaker. The placement of the speaker and each audio recorder for the tests conducted on the Technomad 60/40 speaker can be seen in Figure 3, which overlays the placement of the speaker and each recorder on a satellite image of the Amphitheatre. The center recorder, labeled Recorder 2, was aligned to the speaker's central axis using a laser pointer attached at a 90° angle to the speaker face. The other two recorders, Recorders 1 and 3, were set to the same height as Recorder 2 post-alignment and then placed at their locations, which followed the inner and outer

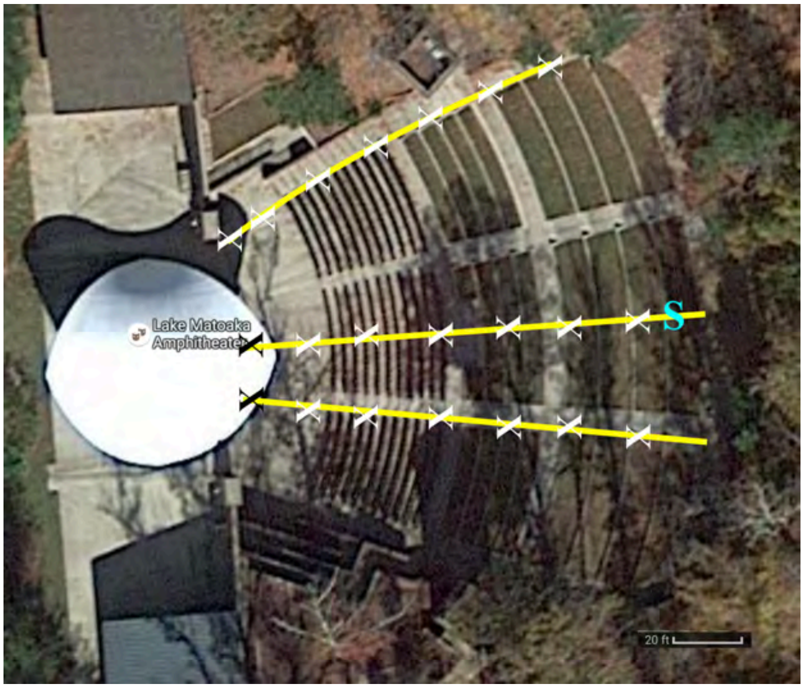

Figure 3: Speaker placement, noted by the blue "S", and audio recorder placement for the Technomad Berlin 60/40 testing. The lowest line of recorder placements were denoted "Recorder 1" in audio recordings, the middle row "Recorder 2", and the upper row “Recorder 3”. Image courtesy of Google maps.

aisle ways of the Amphitheatre. This was originally done to create a standard spacing; however, there is a large curvature of the seating in the Amphitheatre that was not accounted for during the testing of the first speaker model. For the remaining three speaker models, a grid overlaid onto satellite imagery of the Amphitheatre wherein Recorder 2 was placed along the central axis with the speaker, and Recorders 1 and 3 were equidistant from Recorder 2 on opposite sides (1 to the South and 3 to the North), as seen in Figure 4. The spacing of each vertical gridline corresponds to 20’-0” as measured by Google Maps, and will create a more symmetric measurement scheme for the speakers. Each recorder was set to the “Rear 120°” setting so that it only recorded from the speaker side over a 120° spread. This setting takes up less space on the memory card of the recorders as well as filters out any background noise (people on the lake) behind the recorder.

Figure 4: Measurement scheme for the remaining three speaker types, continuing with the same labeling of Recorders 1, 2, and 3.

To isolate the individual frequencies of the pink noise signal discussed in Swaddle et al. (2 kHz to 10 kHz) for the Technomad Speaker, an audio file was created in Audacity using ten ten-second sine wave signals of 500 Hz increments from 500 Hz to 10 kHz constructed in Matlab, as seen in Appendix A [5]. Between each of the ten signals, a five second silence was inserted to more easily distinguish the differences in signals in the spectrogram reading from the audio recorders. Specific gain levels were set on each frequency, so as not to overload the amplifier or the speaker, that were consistent throughout testing. Three trials were conducted per angle, and three angles at 0°, 20° to the right, and 40° to the right, marked on the speaker using a protractor, were tested. For the 9.4" Indoor/Outdoor 70 Volt PA Horn Speaker, the length of the audio was kept the same, but the frequencies of 200 Hz, 300 Hz, and 400 Hz were added in to account for the limits of the speaker. Likewise, the angles tested were 0°, 30°, 60°, 90°, 120°, 150°, and 180°, all to the right. Because the horn speaker has a large inherent acoustic spread (its purpose is to cover as large of an area as possible), we wanted to account for the sound in all directions, thus measuring over a complete semi-circle. Acoustic waves propagate spherically, so we only did half of a circle, as it should be symmetric for the other side of the speaker. Since the amphitheater is roughly 45 meters long and sound travels at 340 m/s, the audio files were shortened significantly for the cube speaker to assist in shortening the time required to be at the amphitheater gathering data. The signals were shortened to 0.5 seconds; with a 0.5 second gap in between. This allowed for the sound waves to propagate through the amphitheater and reach the recorders, but not waste time, and also allotted a short time for the sound to die out before the next signal began. This audio file was repeated three times (three trials per angle) to account for any wind or extraneous noise that could interfere with the recording. For the cube speaker, it was important to measure the area between the speakers on each face, as well map out the overall spread of the cube. To do this, the cube was measured at 0°, 30°, 60°, and 90° for distances 1, 2, 3, 5, 6, and 0°-330° in 30° increments for distance 4. This was done as a sanity check for the speaker, to ensure that there was accurate symmetry surrounding the speaker.

The largest factor to consider when testing was wind speed, because too much wind would interfere with the data being recorded. Wind causes a noise signal in the lower frequency ranges (< 500 Hz) that becomes higher pitched and louder as wind speed increases [12]. This effect can be seen on some of the cube speaker data, where there is a large amount of noise in the lower frequencies (<300Hz), and affected the lower-frequency data. In an attempt to combat this, testing was conducted on days with wind speed less than ten miles an hour. However, frequent planes passing overhead and large gusts of wind (on relatively non-windy days) still affected the recordings. Another factor that affected the recordings was students and other Williamsburg residents that kept walking by and having their own conversations. To try and reduce the effect this had on the recordings, anyone walking by was asked to keep quiet until they were out of the amphitheater. This did not have a large effect on data, but when combined with any wind gusts or other outside noise, amounted to enough noise that affected low-frequency measurements.

In order to characterize the acoustic spread of each speaker, a Fast Fourier Transform was conducted on the output signal (that signal that we sent through the speaker) and the input signal (that which we recorded) using Matlab, as seen in Appendix B. As defined by Raichel, a Fast Fourier Transform analysis “executes an efficient transformation of the signal from the time domain to the frequency domain”[13] This returns a plot that, theoretically, contains spikes at specific frequencies by which we see and can measure the amplitude of each frequency; reducing a large time-domain audio file to a short, easy-to-read spectra of frequencies, as seen in Figure 5 below [14]. By measuring and comparing the amplitudes, we can extrapolate the amplitude of the various frequencies based on the angle and distance from the speaker in order to create a map of the acoustic spread. While modern technology has made conducting Fast Fourier Transforms possible, there are still limitations. Each recording contains multiple angles and in the case of the Technomad speaker, trials, resulting in an audio file that must be manually split into the separate

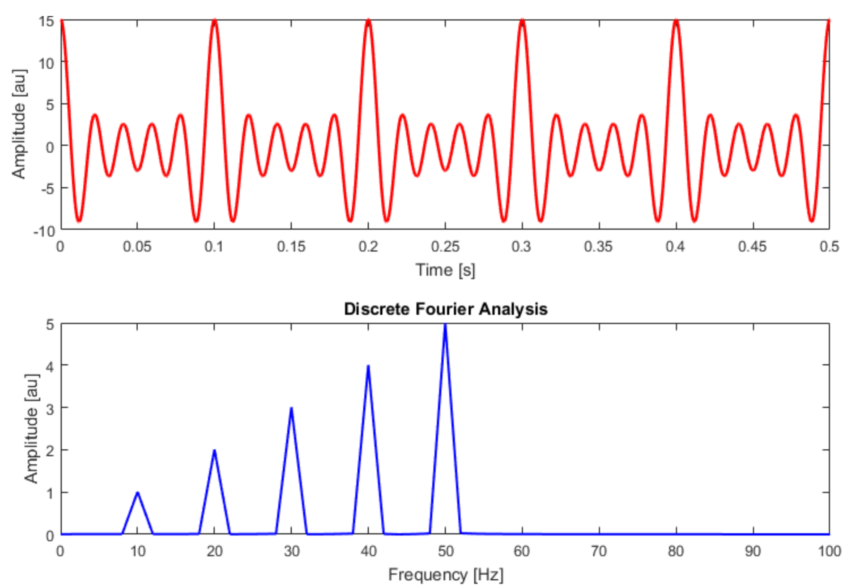

Figure 5: An example of a Fast Fourier Transform, as conducted on a cosine summation function [10].

angles and trials. Even with cutting the twenty minute-plus audio files down to six minutes, the six-minute files still take a very long time and a significant amount of computer memory to process. Since computer memory is limited, in both a laptop and the William and Mary School computers, the initial attempt for data analysis consisted of using the BG servers (almost ½ TB of memory, compared to the MacBook Pro 8GB memory) in the Computer Science Department. While significantly faster and able to process the data in a reasonable amount of time, this format required remote server access and did not allow for the graphical interface of Matlab, which was necessary to get the data values for these particular sets of data.

At the time of writing, only the Technomad 60/40, the 9.4" horn speaker, and the custom cube speaker have been measured. The first Technomad speaker that we used started smoking from various spots during the second hour of testing, and had to be replaced with a new speaker. The first speaker had previously been used extensively in another test, and as a result had lost most of the function of the tweeters, which produce the higher frequencies (2 kHz to 20 kHz) [15]. By playing the audio file through the damaged speaker, the components were overloaded and caught fire, as indicated by the smoke. Because we did not know how extensive the damage was before testing began, all testing that was previously conducted on the first speaker was redone on the second speaker. Gain levels set for this first speaker were readjusted for the second, and the volume of the laptop was kept consistent throughout testing. For the horn speaker and cube speaker, the gain was set to zero as it created a more equal experimental design and did not put an artificial system error into the design.

II. Data

More than 400 trials were conducted for the three speakers, each one labeled with a specific Recorder, Angle, and Trial number. Audacity, the primary application for audio files, has the ability to show a recording as a spectrogram, as seen in Figure 6 (below); however, it does not give any numerical values to the data. Therefore, the .wav files must be imported into Matlab, and then put through a Fast Fourier Transform to give a frequency versus amplitude plot. This was done to each trial, and then the average response of each Recorder at a specific angle and distance, based on the three trials, is plotted as a function of said distance and angle.

Currently, there are many recordings that took place during continual testing, resulting in

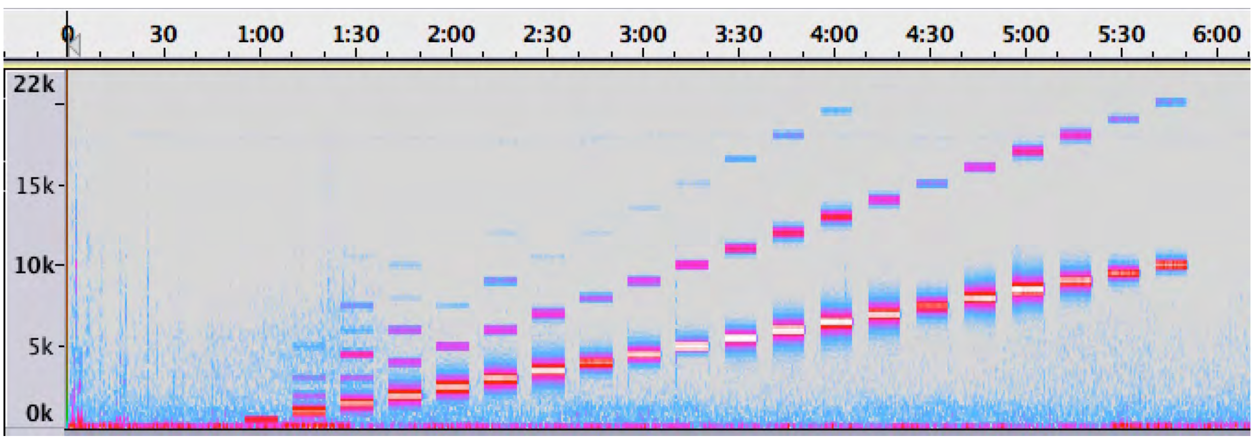

Figure 6: The spectrogram readout of Technomad Recorder 2 at 20° twenty feet from the speaker. As seen, it only provides a qualitative view of the frequency response over time.

multiple trials per recording. These audio files are being broken into separate files, one for each trial. Once the recordings are cleaned up (by filtering out background noise), they will be put through the previously discussed data analysis to determine the amplitude of the response based on the angle of the recorder to the speaker. Because the audio files were so long and the priority was placed on getting them all split into their specific files and coded correctly ("recorder #" – "distance" – "angle"), there was only time left to go through the central recorder's data on the cube speaker. The audio files for the cube speaker were only a minute and a half long after splitting and coding, which took about 20 seconds processing time in Matlab. Then, the data cursor was used to select the maximum value of the transformation results, and this information was recorded, taking up about 20 minutes (longer than the recording itself). Figure 7, right, is an example of the output of the Matlab code, specifically for the center recorder at distance 1 (approximately 20 ft.) at an angle of 0°. As seen in the figure, there are very sharp spikes at each of the tested frequencies, as well as a small amount of noise at the lower frequencies. This noise did not affect the measurement of the amplitude at the lower frequencies. In Figure 8, right, is the same recorder at the same angle, but at distance 4 (80 feet). There is a noticeable difference in amplitude between the two samples, something that can be examined across all angles and distances to map out the spread of the speaker. To do so, it is useful to plot the data for each distance and angle, and then fit this with a curve to discover the best-fit relationship between the variables.

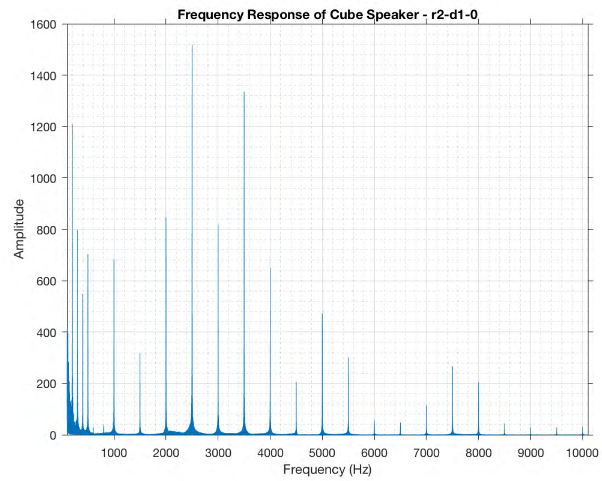

Figure 7: Frequency Response of Cube Speaker from Recorder 2 at Distance 1 and Angle 0.

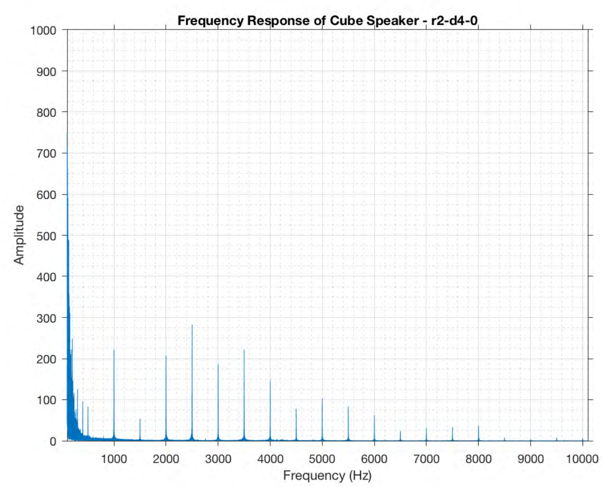

Figure 8: Frequency Response of Cube Speaker from Recorder 2 at Distance 4 and Angle 0.

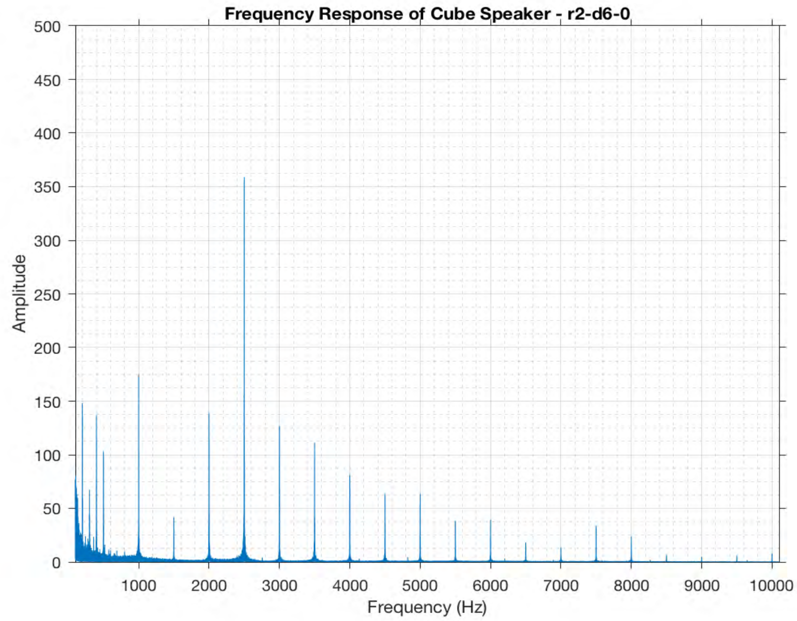

Figure 9: Frequency Response of the Cube Speaker at distance 3, 0°. Note how the 2500 Hz response is almost three times larger than any other response.

As seen in the plots in Appendix C, the 2500 Hz frequency has the greatest strength for the Cube Speaker. Figure 9, above, shows the frequency response of the speaker at distance 6, 0°. It is very clear that the 2500 Hz signal is strongest, with the surrounding frequencies being quite strong as well, which indicates that this speaker is suited more towards frequencies on the 1-4kHz range. Because the 2500 Hz signal is strongest across each trial, it is worthwhile to look specifically at how this signal behaves over both distance and angle.

III. Analysis

To take a closer look at this speaker, we can analyze how it behaves over all six distances, as well as how it behaves angularly at a singular distance. First, by plotting the amplitude of the signal across all six distances, we can see a fairly obvious best fit line, as seen in Figure 10. While it is obvious that there is a general trend that intensity of audio decreases with distance, there are not enough data points (or trials per data point) to get a truly reliable measure of the relationship. The variations seen in the plot below can be due to any number of causes, including wind or simply the audio not hitting the same spot on the recorder at the different distances. The last of these options is the most likely, as the changes in vertical height in the amphitheater are inconsistent and could cause the center of the acoustic wave to miss the recorder. Likewise, measuring the audio at more incremental distances, such as every 10 feet

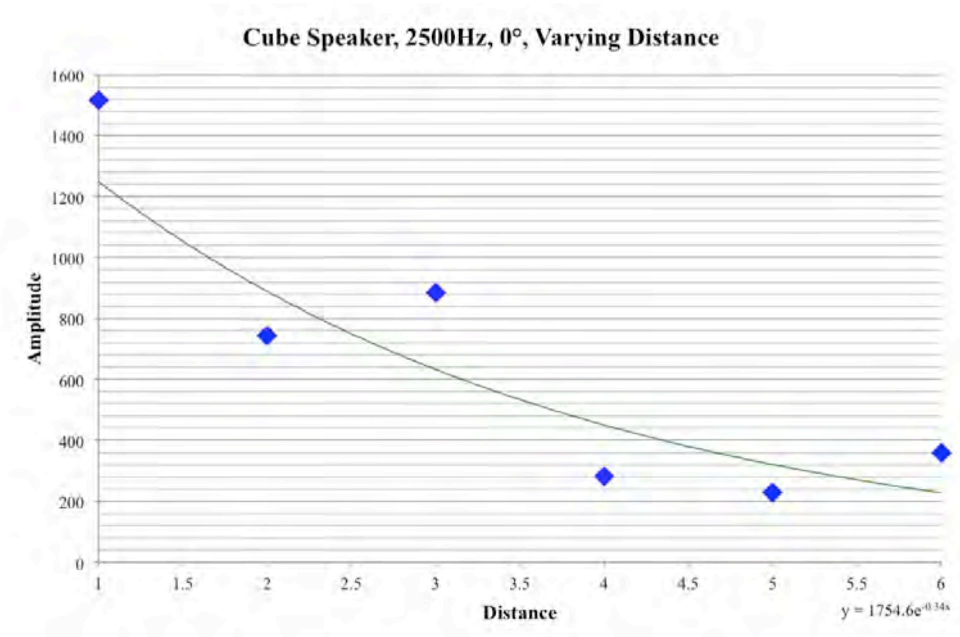

Figure 10: Cube Speaker behavior at the same angle (along the central axis of the acoustic wave) over varying distance.

instead of every 20 feet, as well as measuring along further distances could improve upon this data. This would only be made possible with a larger space than the amphitheater, as Distance 6 hits the edge of the amphitheater.

When looking at the relationship between the angle of the speaker and the response, there is no decipherable pattern, as seen in Figure 11. The 2500Hz signal is strongest every 60°, with a dip in this strength at 240°. However, looking at this as purely single-speaker strength is problematic due to the fact that there are four speakers, all playing the same signal, on the Cube Speaker. This means, then, that there should be a pattern that repeats every 90°; i.e., the 0°, 90°,

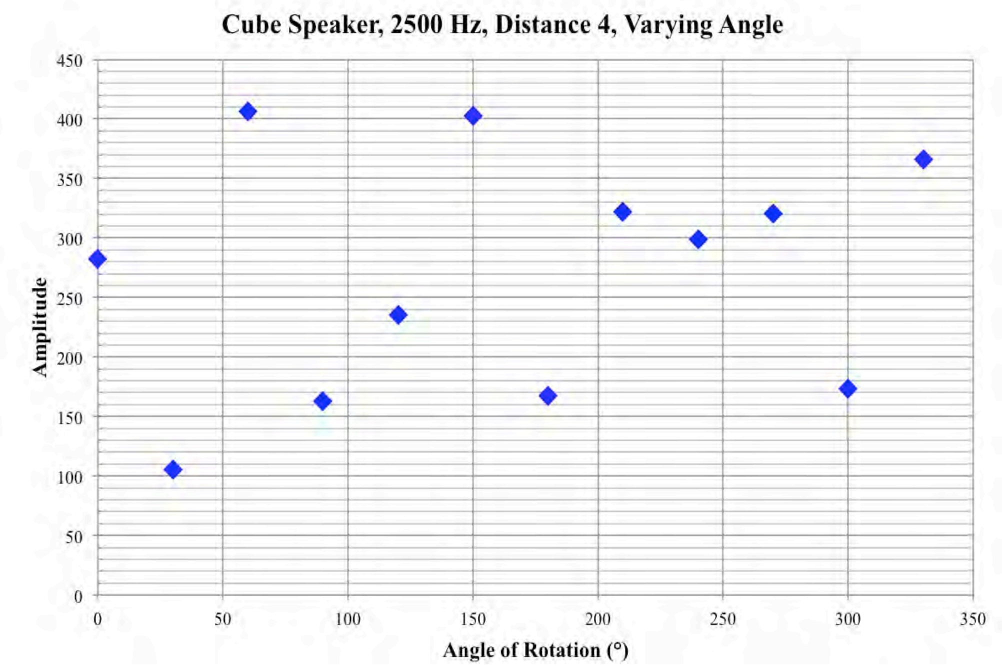

Figure 11: Cube Speaker behavior at the same distance (just past the halfway distance) over a varying angle with 30-degree increments.

180°, and 270° amplitudes should all be the same; the 30°, 120°, 210°, and 300° amplitudes should all be the same; and the 60°, 150°, 240°, and 330° amplitudes should all be the same. While it is true that the angles that fall within each of these three groups are similar, they are not equal as expected. As before, this could be due to changes in the air between the speaker and the recorder, as sound is a physical wave that can be affected by changes in air pressure, as well as could be due to inaccurate measurement of the angle of the speaker. While the markings on the speaker were created with a protractor, there is always the human error component that could affect this; especially when a small error at the speaker end could result in a larger error at the recorder end.

Lastly, it is important to analyze the response of the speaker to each of the frequencies. 2500 Hz is the most responsive frequency for the cube speaker, and 10000Hz is the least responsive, but the comparison of all the frequencies to one another is vital to understand so that the appropriate speaker can be chosen for the appropriate species of bird in the commercial applications of the product. Figure 12, below, shows the comparison of the 500-9500 Hz frequencies in 500 Hz increments. As seen in this plot, the 2500 Hz is, in fact, the “loudest” of the signals, with 3500Hz also measuring much “louder” than the other signals. The rest of the frequencies all have a similar best-fit curve; and for almost all of the frequencies, the higher the frequency, the “quieter” it is. This plot suffers from the same problems as the previous equal angle / varying distance plot: the data points do not follow a logical curve, but a best fit curve with a small amount of error can be applied. Unfortunately, all this does is confirms what we know about sound: that it decays exponentially. Fortunately, if more data were to be taken between the distances, or more data were taken at each distance, a much more reliable plot might be created that can be significantly more helpful in terms of layout design. Overall, this gives valuable insight as to the behavior of the Cube Speaker; and when combined with the data of the other two speakers will aid in the design and implementation of the commercial application of the Sonic Nets technology.

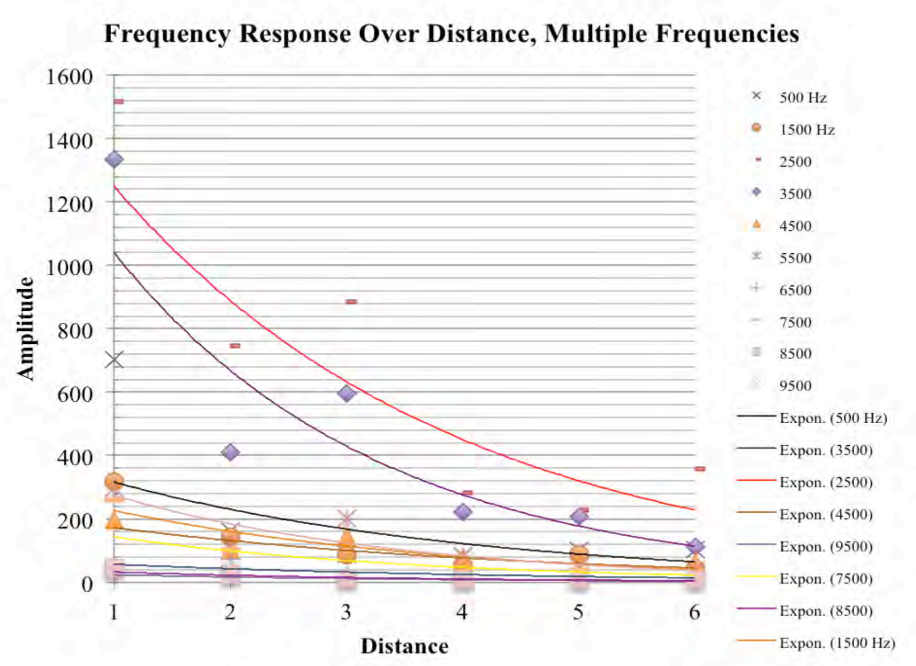

Figure 12: Response of all frequencies over all distances at a constant angle (0 degrees).

IV. Conclusions and Future Work

The data that was collected over the course of the year is all prepared and ready for being analyzed by Matlab, something that could hopefully be streamlined by a computer with greater memory and something written into the code that calculated the maximum at a certain x-value (Frequency). As seen in the plots from the Cube Speaker (Figures 9-11), there is a confirmed relationship between the distance from the recorder, the angle of the speaker, and the resulting amplitude of the sound, as fitted with a relationship via Matlab. This relationship can be used by researchers to improve design of the pink noise system, as it will allow them to maximize coverage with the minimum amount of equipment, as well as help design the acoustic shell.

At this point, almost all of the audio data has been cut, labeled, and sorted, and some of it has been translated into a graphical representation via Matlab. There are more than 25 hours of raw audio, all of which must be cut to each trial, then processed via Matlab, and then analyzed. Much of the time of this project was spent actually recording the audio, and then processing and cutting all of the audio files into their specific trials. This was done to make the files being put through analysis in Matlab small enough so that they do not use excessive memory of the computer; too large a file would cause the computer to freeze and crash. Moving forward, all of the audio is ready to be put through the Fast Fourier Transform algorithm, which will allow a future researcher to create more detailed acoustic spread diagrams.

Once the speakers' acoustic spreads have been mapped out, we can move forward with designing the acoustic mirror. To design the acoustic mirror, we will need to research the materials and densities best suited to absorb the various frequencies on one side of the speaker, as well as design and build the mold for the concrete. As a departure point, we can look at past models of acoustic mirrors, such as the large-scale ones used for airplane detection, to gain insight into the shaping of the cavity that the speaker will be placed in front of. While many models of acoustic mirrors are much too large to be practical in our case, they provide valuable information on determining the focal point of the mirror and the mirror’s effects. Various molds and test structures, based on the speaker data, will be made to test the performance of the mirror, which will then be used in commercial applications of the Sonic Nets.

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APPENDIX A: AUDIO CREATION CODE

4/21/17 2:10 PM /Users/jordanleek/D.../freqpick.m 1 of 1

```
function [ ] = freqpick( LT,fs,freq,filename)
%freqpick is used to make audiofiles of different length and frequencies
%for the eagle hearing tests
%Input Arguments
%   LT      lenth of time for signal in seconds
%   fs      sampling frequency generally 44100
%   freq    frequency of the signal in Hz


T=(0:1/fs:LT);
signal=sin(2*pi*freq*T);


audiowrite(filename,signal,fs)



end
```

This code was used to create each of the audio signals, including: 200 Hz, 300 Hz, 400 Hz, and 500-10000 Hz in 500 Hz increments. The length of the sample was adjusted as necessary, ranging from 0.5 seconds to 10 seconds between speaker models. Code courtesy of Elizabeth Skinner, researcher with Nondestructive Evaluation Group.

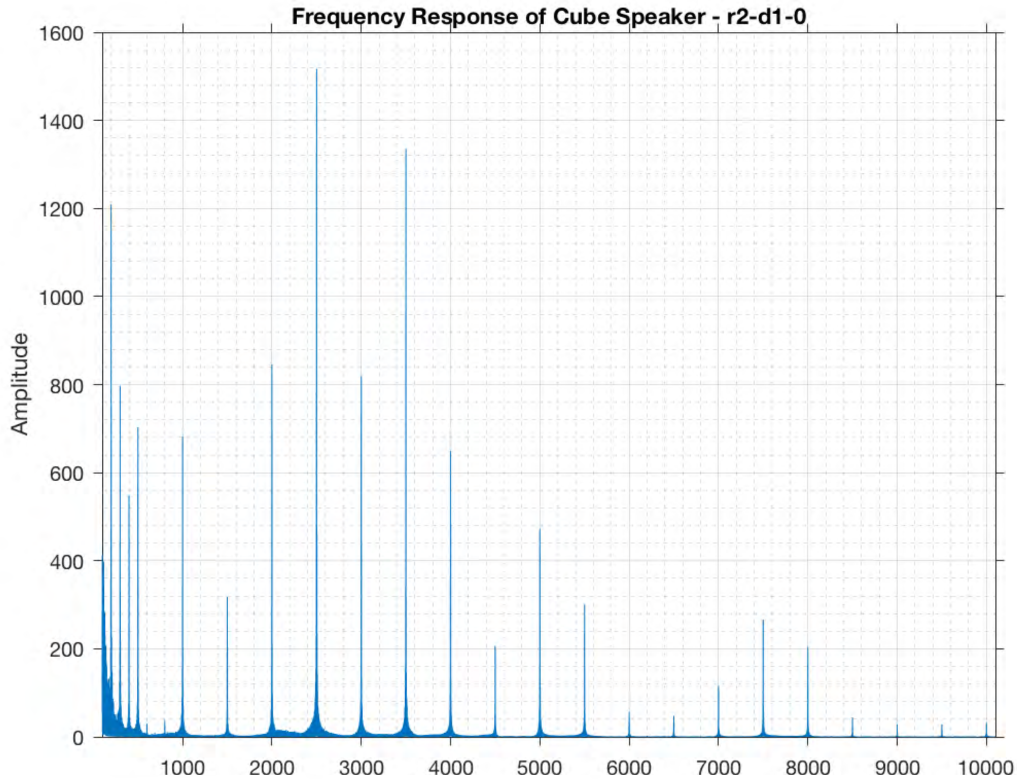
APPENDIX B: FAST FOURIER TRANSFORM & PLOTTER CODE

```
%Jordan Leek
%Senior Project
%Data Analysis

%Load File
[y,Fs] = audioread('/Users/jordanleek/Documents/School/2016-
2017/Research/Cube Speaker/Recorder 2/r2-d6-90.wav');

Nsamps = length(y);
t = (1/Fs)*(1:Nsamps);          %Prepare time data for plot

%Do Fourier Transform
y_fft = abs(fft(y));            %Retain Magnitude
y_fft = y_fft(1:Nsamps/2);      %Discard Half of Points
f = Fs*(0:Nsamps/2-1)/Nsamps;   %Prepare freq data for plot

%Plot in Frequency Domain
figure
plot(f, y_fft)
set(gca,'TickDir','out')
grid on
grid minor
xlim([100, 10100])
xlabel('Frequency (Hz)')
ylim([0, 500])
ylabel('Amplitude')
title('Frequency Response of Cube Speaker - r2-d6-90')
print('/Users/jordanleek/Documents/School/2016-2017/Research/Cube Speaker/r2-
d6-90', '-dpng', '-r0')
```

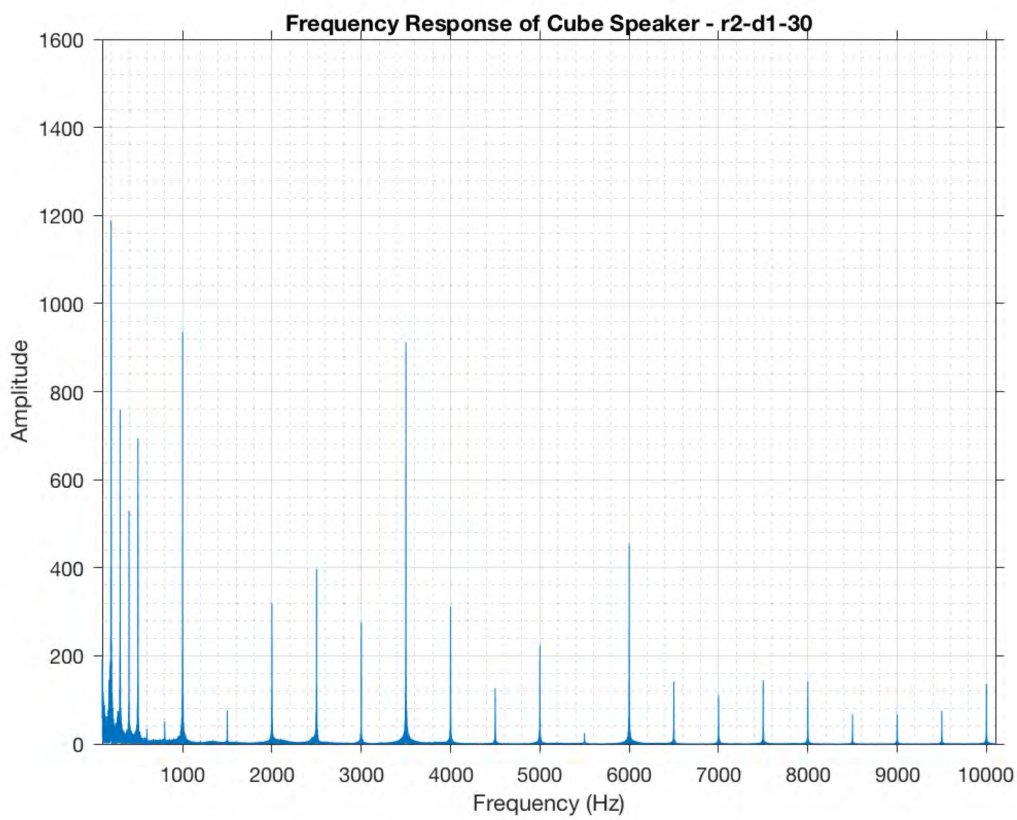
This code was used to analyze each recording after splitting it into its specific recorder/distance/angle file. The x-limit was kept consistent at 100-10100 Hz, as we were not testing anything below 100Hz and this would just be extraneous noise from the environment, and we did not need to look at anything above 10000Hz. The y-limit changed depending on the maximum recorded during the first distance. As the first distance was closest, it stood to reason that this would be the “loudest” recording and thus the greatest y-value measured. This y-value was then kept consistent for the remainder of the recordings. To keep track of the plots created by Matlab, a title with the recording’s “code” (rX-dX-X) was placed on the plot as well as the creation of a filename with the same code.

APPENDIX C: CUBE SPEAKER RECORDER 2 PLOTS

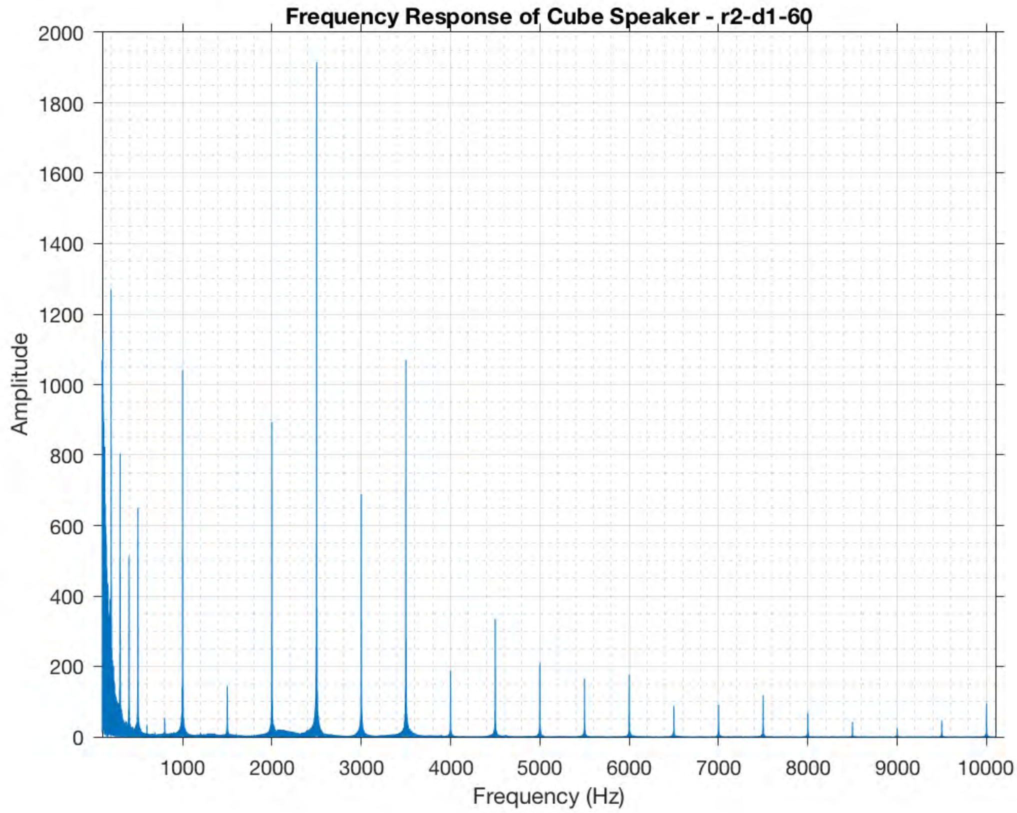
For the recorder along the central axis of the cube speaker, Recorder 2, a fast Fourier transform algorithm was applied to the data. This graphical representation is shown for each set of data, along each distance and spanning each angle, and shows a general trend between these factors. On the plot, there are a number of spikes; these correspond to the response of the speaker to each of the frequencies and give a means of comparing the speaker across all angles and distances. Large amounts of noise result more solid sections of blue, as seen in many of the trials at the lower frequencies. The data cursor function of Matlab was used to determine the peak amplitude of each spike, which was then used in combination with the other results to create the other graphs comparing the responses at various angles and distances.

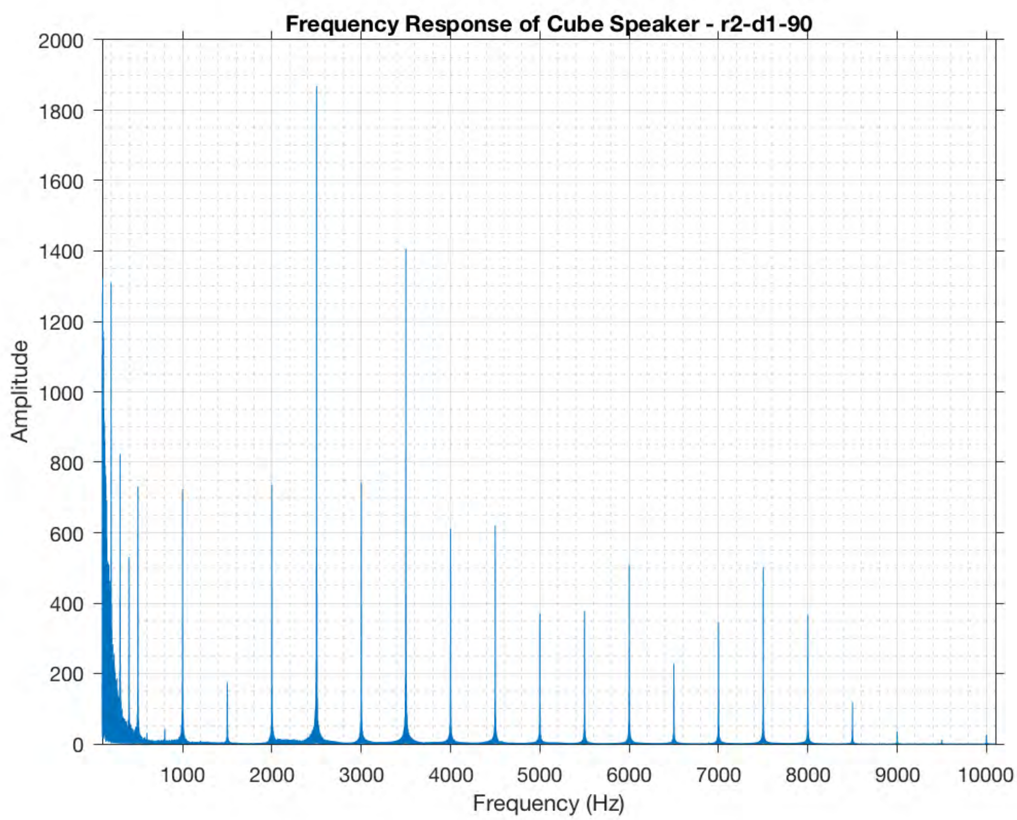


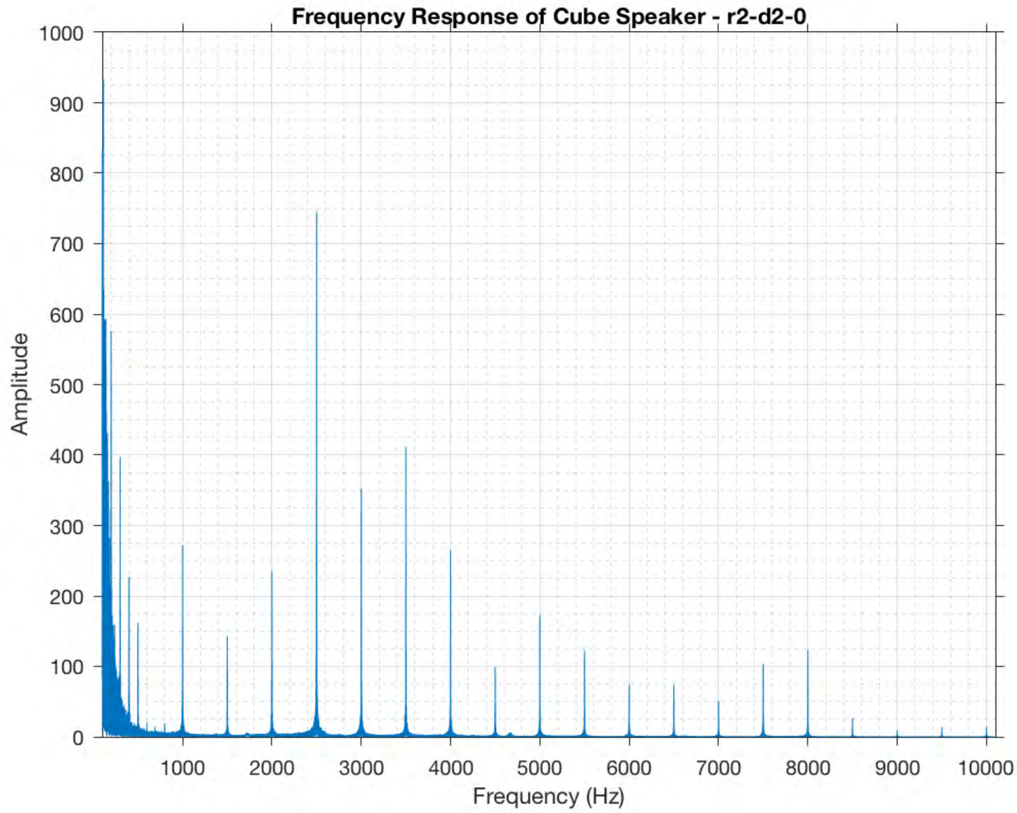
Frequency Response of Cube Speaker - r2-d1-0
Amplitude
0
200
400
600
800
1000
1200
1400
1600
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

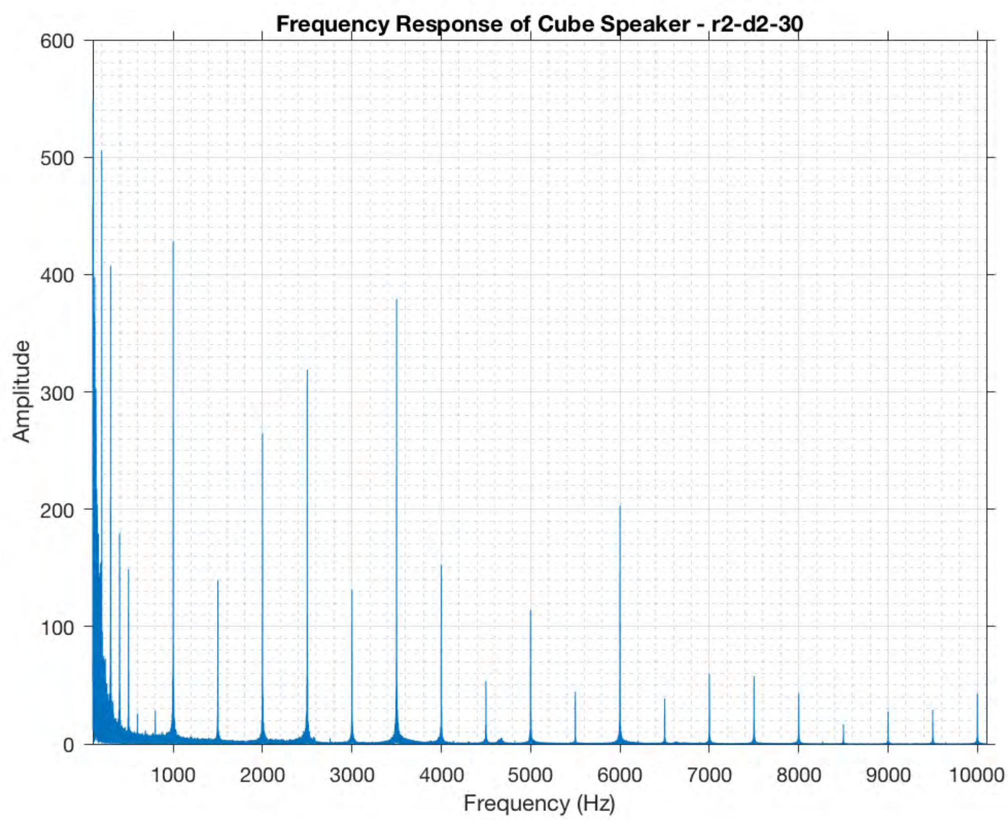


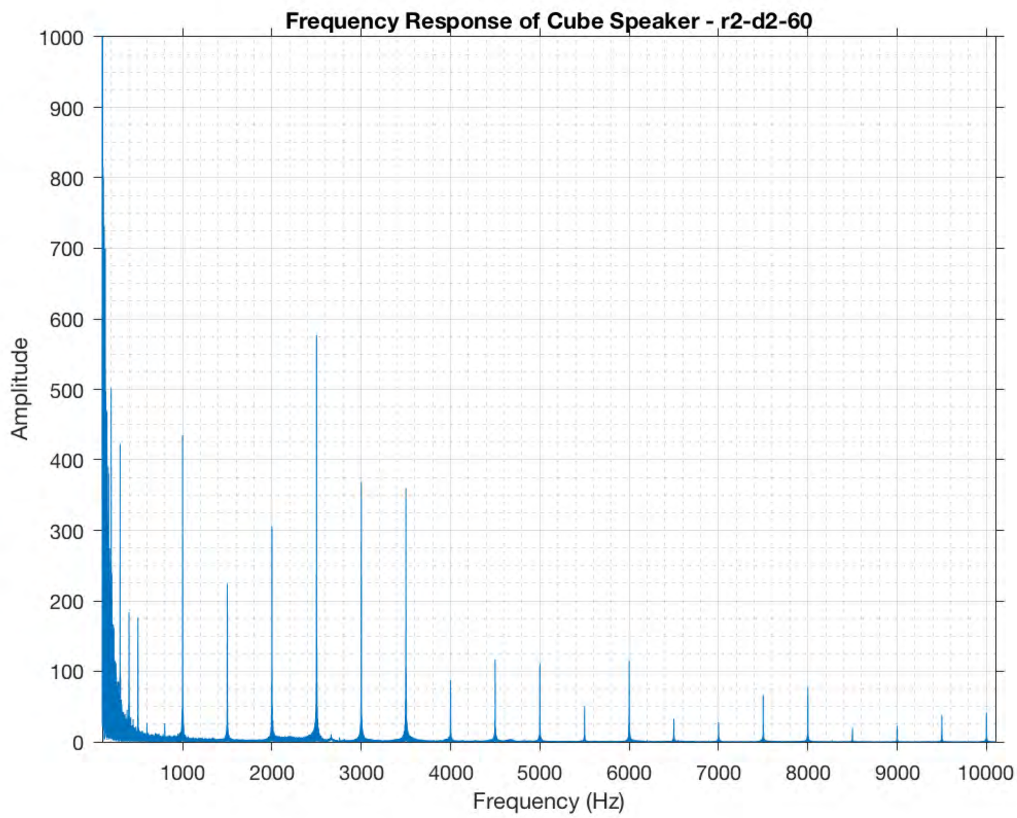
Frequency Response of Cube Speaker - r2-d1-30
Amplitude
0
200
400
600
800
1000
1200
1400
1600
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000
Frequency (Hz)

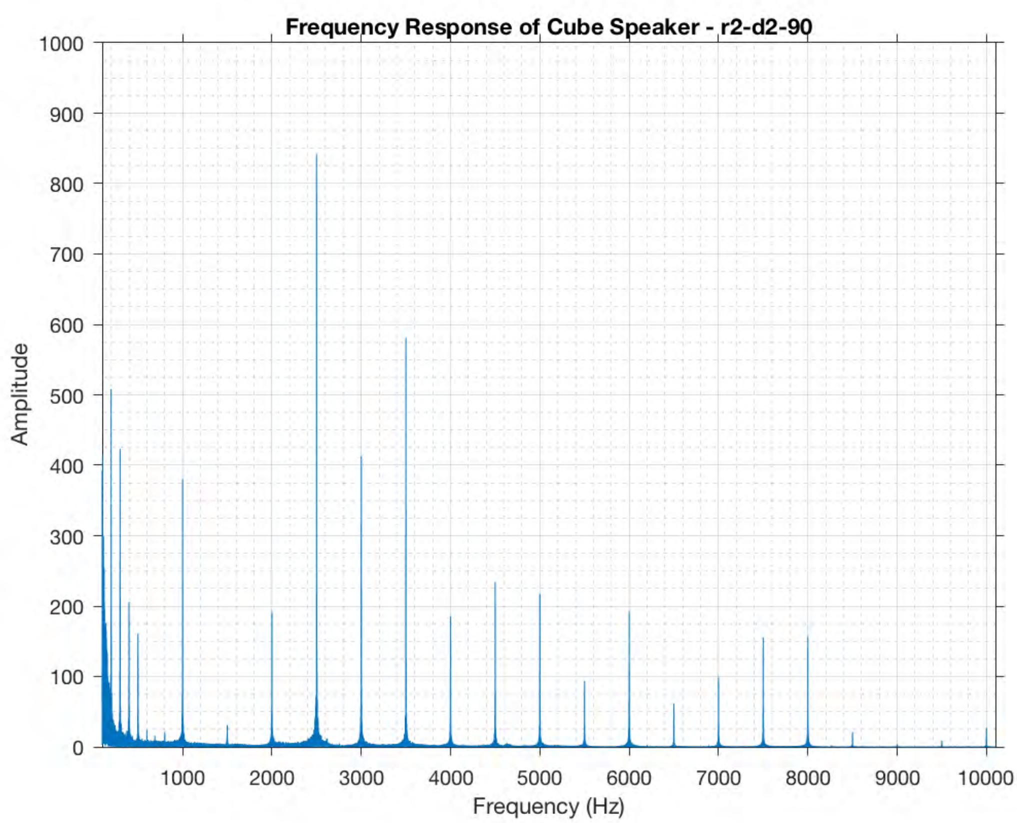

Frequency Response of Cube Speaker - r2-d1-60
Amplitude
Frequency (Hz)
0
200
400
600
800
1000
1200
1400
1600
1800
2000
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

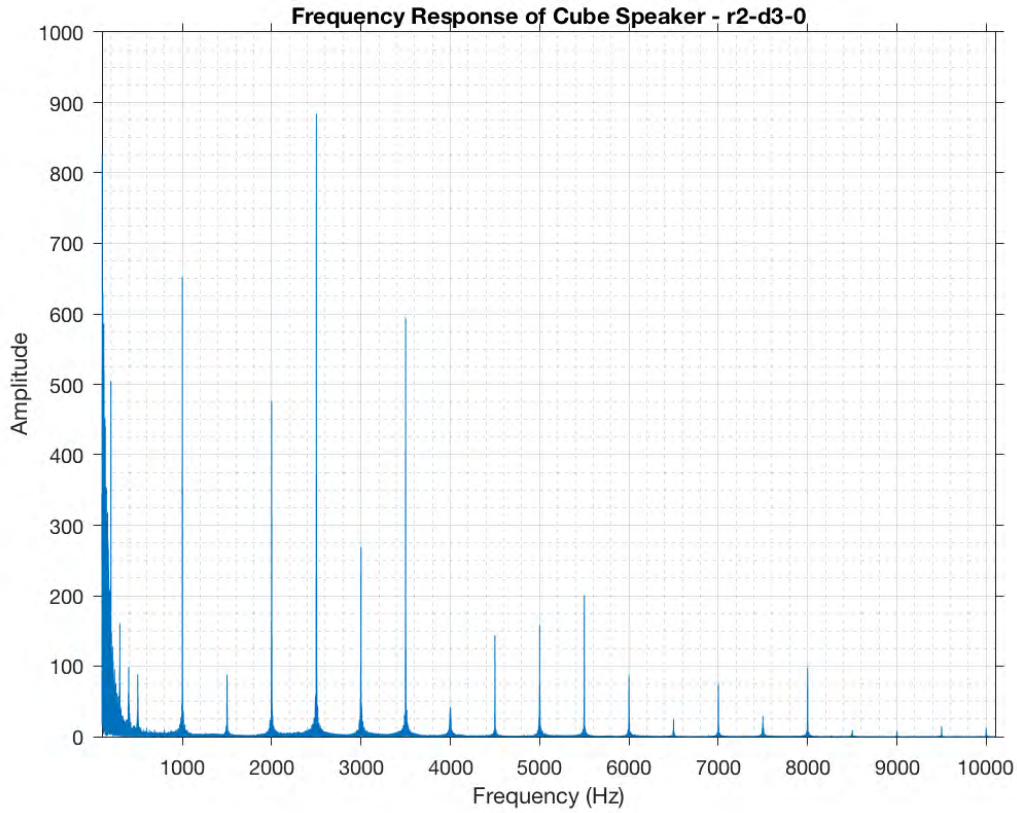

Frequency Response of Cube Speaker - r2-d1-90
Amplitude
Frequency (Hz)
0
200
400
600
800
1000
1200
1400
1600
1800
2000
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

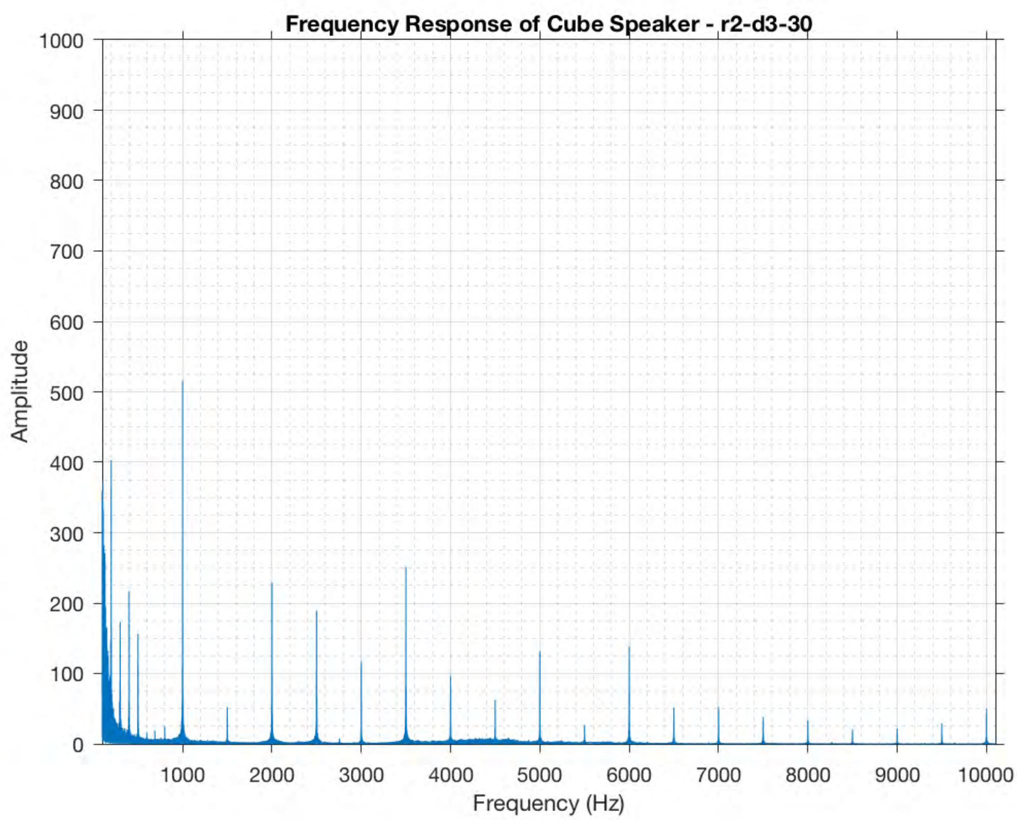

Frequency Response of Cube Speaker - r2-d2-0
Amplitude
Frequency (Hz)
1000
900
800
700
600
500
400
300
200
100
0
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

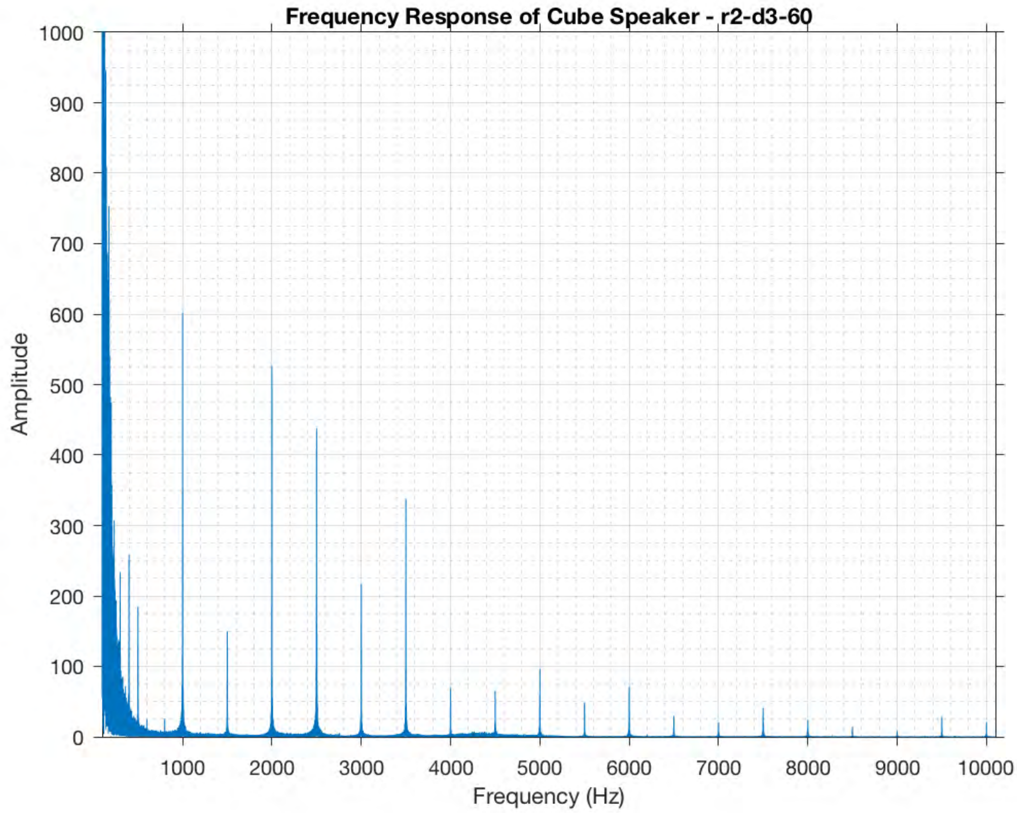

Frequency Response of Cube Speaker - r2-d2-30
Amplitude
Frequency (Hz)
600
500
400
300
200
100
0
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

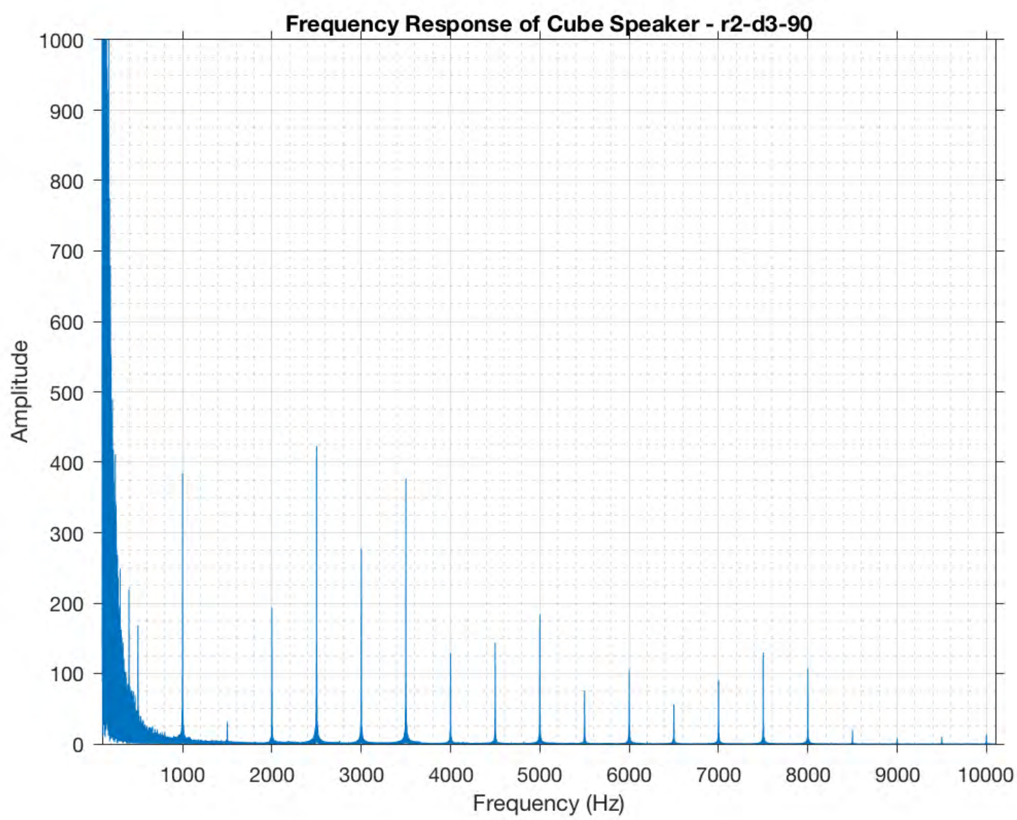

Frequency Response of Cube Speaker - r2-d2-60
Amplitude
Frequency (Hz)
0
100
200
300
400
500
600
700
800
900
1000
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

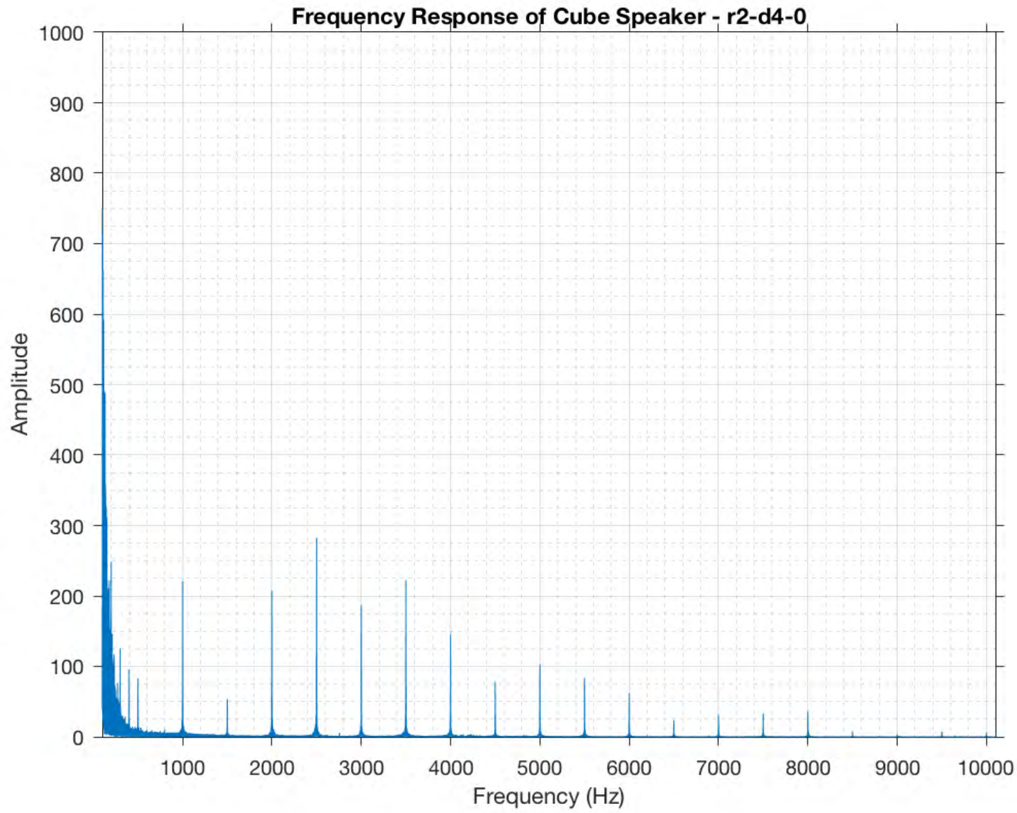

Frequency Response of Cube Speaker - r2-d2-90
Amplitude
Frequency (Hz)
0
100
200
300
400
500
600
700
800
900
1000
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

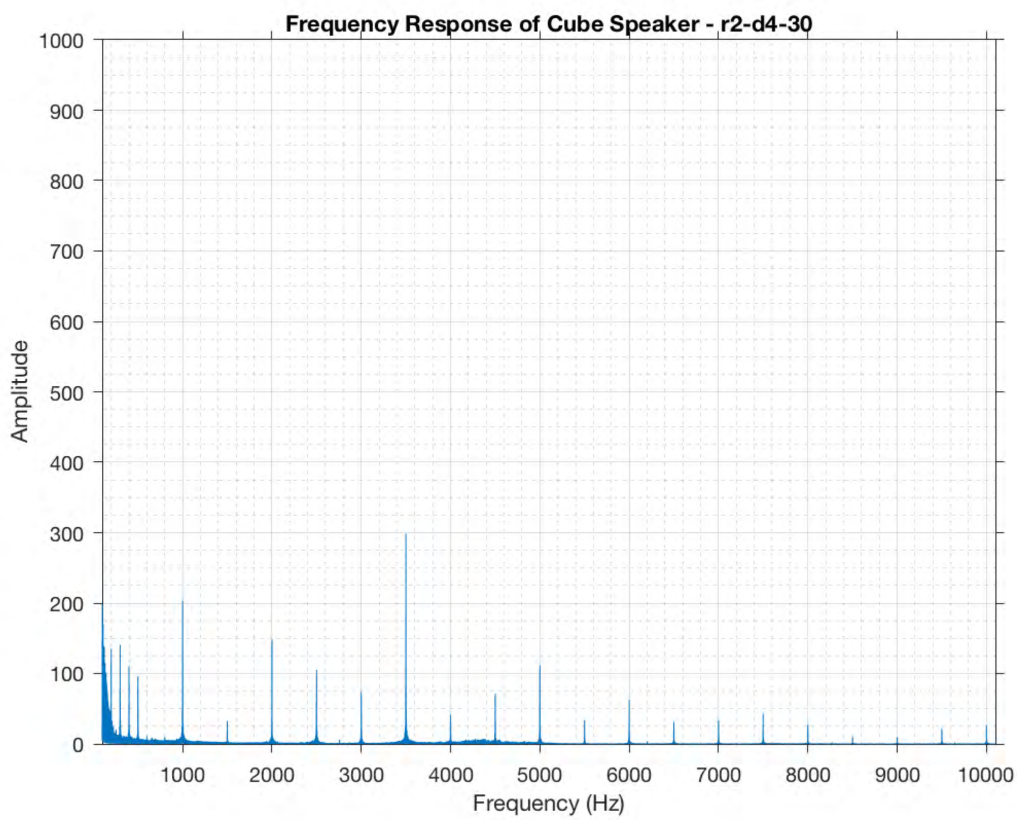

Frequency Response of Cube Speaker - r2-d3-0
Amplitude
Frequency (Hz)
0
100
200
300
400
500
600
700
800
900
1000
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

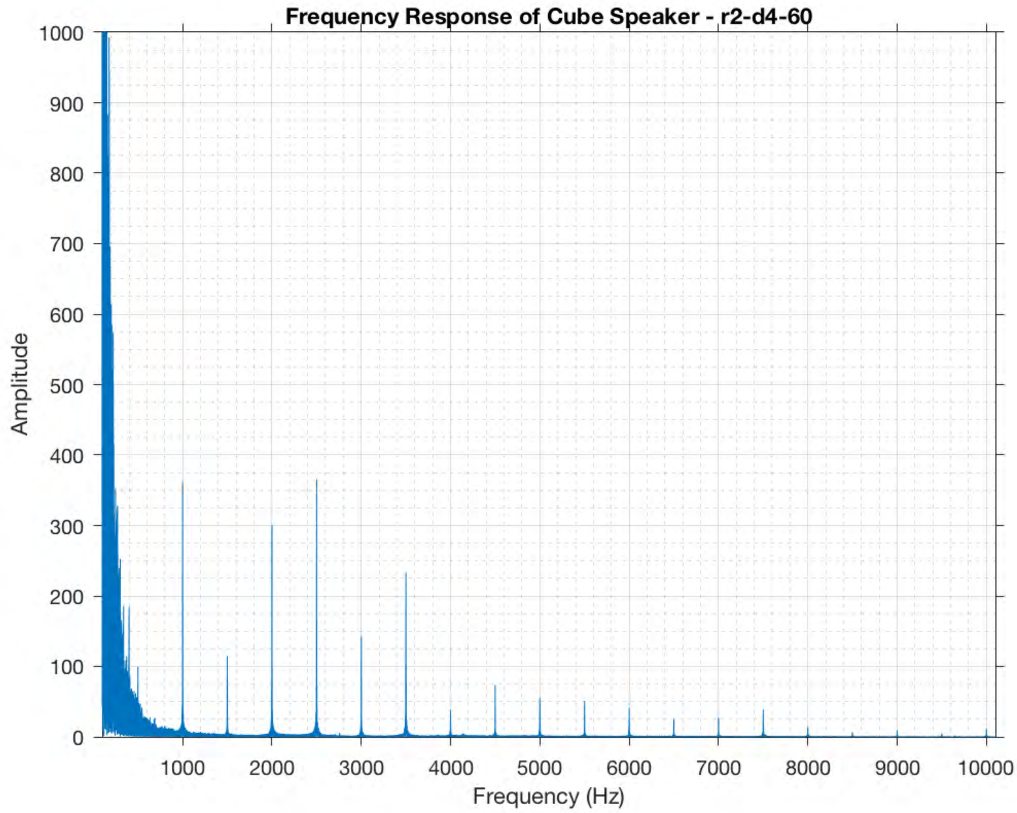

Frequency Response of Cube Speaker - r2-d3-30
Amplitude
Frequency (Hz)
0
100
200
300
400
500
600
700
800
900
1000
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

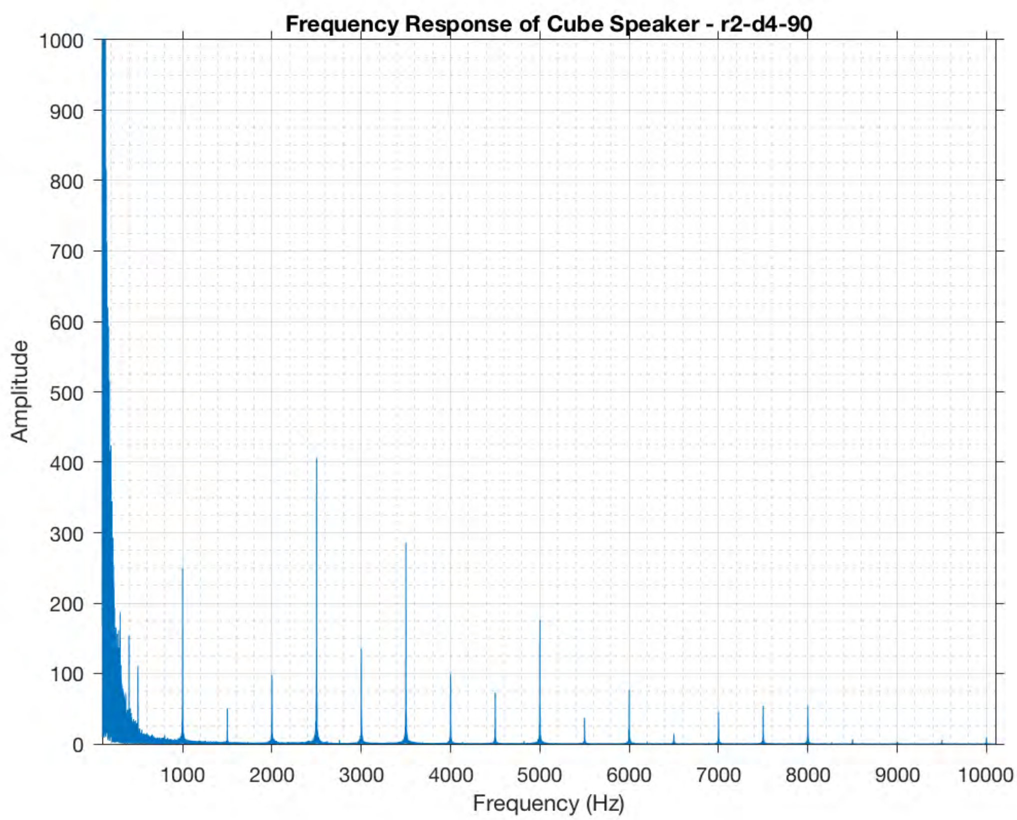

Frequency Response of Cube Speaker - r2-d3-60
Amplitude
Frequency (Hz)
1000
900
800
700
600
500
400
300
200
100
0
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

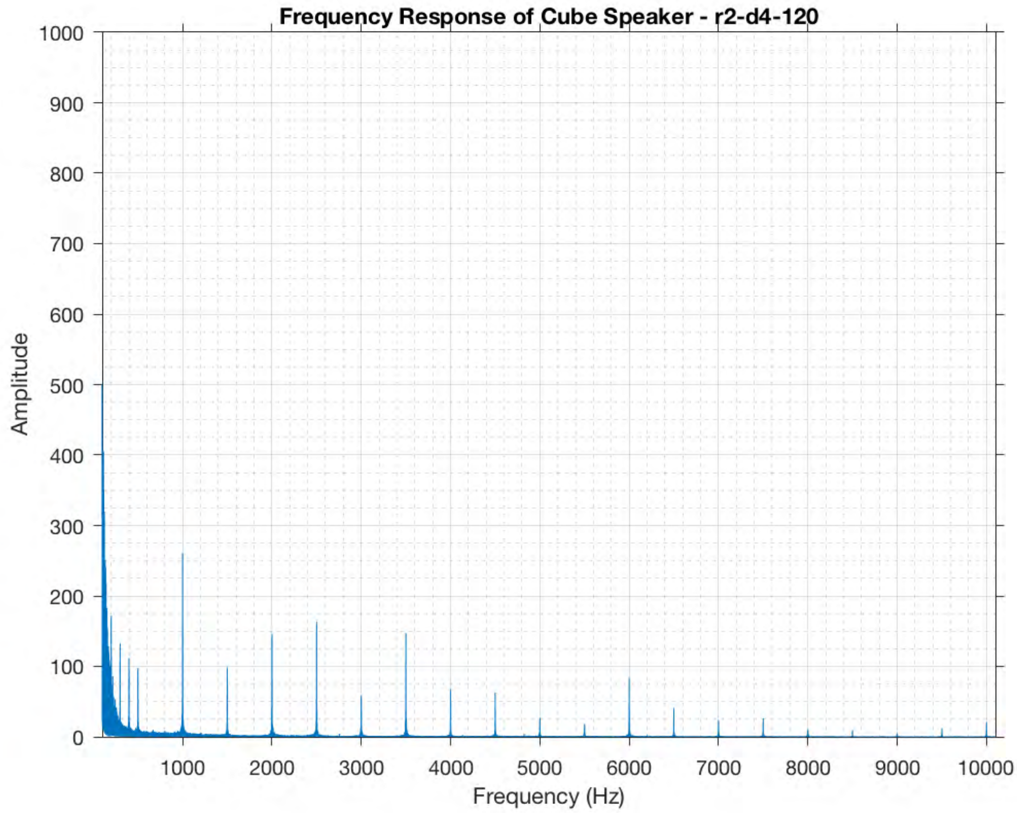

Frequency Response of Cube Speaker - r2-d3-90
Amplitude
Frequency (Hz)
1000
900
800
700
600
500
400
300
200
100
0
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

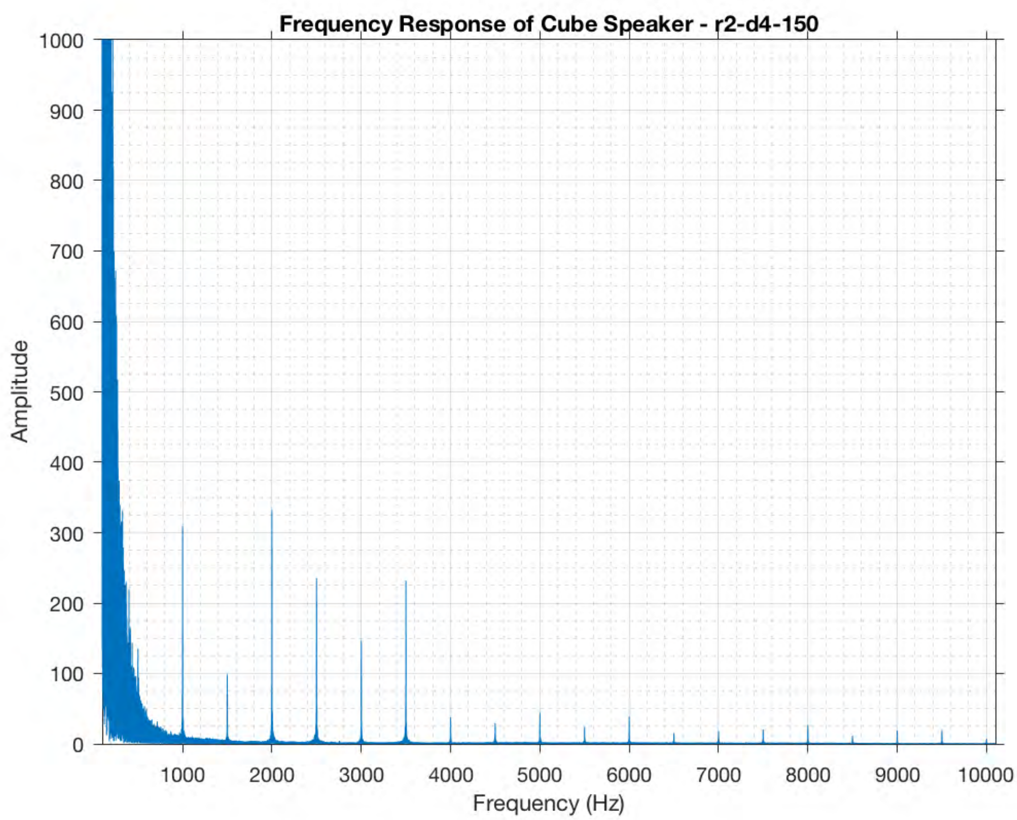

Frequency Response of Cube Speaker - r2-d4-0
Amplitude
1000
900
800
700
600
500
400
300
200
100
0
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000
Frequency (Hz)

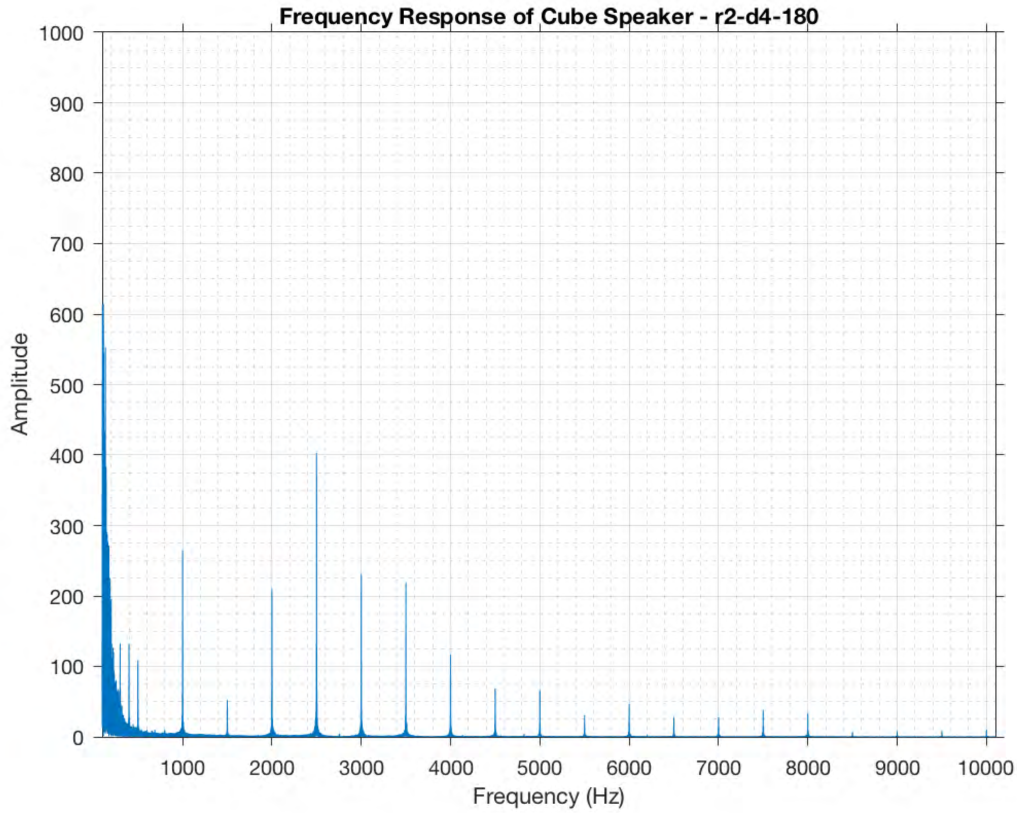

Frequency Response of Cube Speaker - r2-d4-30
Amplitude
1000
900
800
700
600
500
400
300
200
100
0
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000
Frequency (Hz)

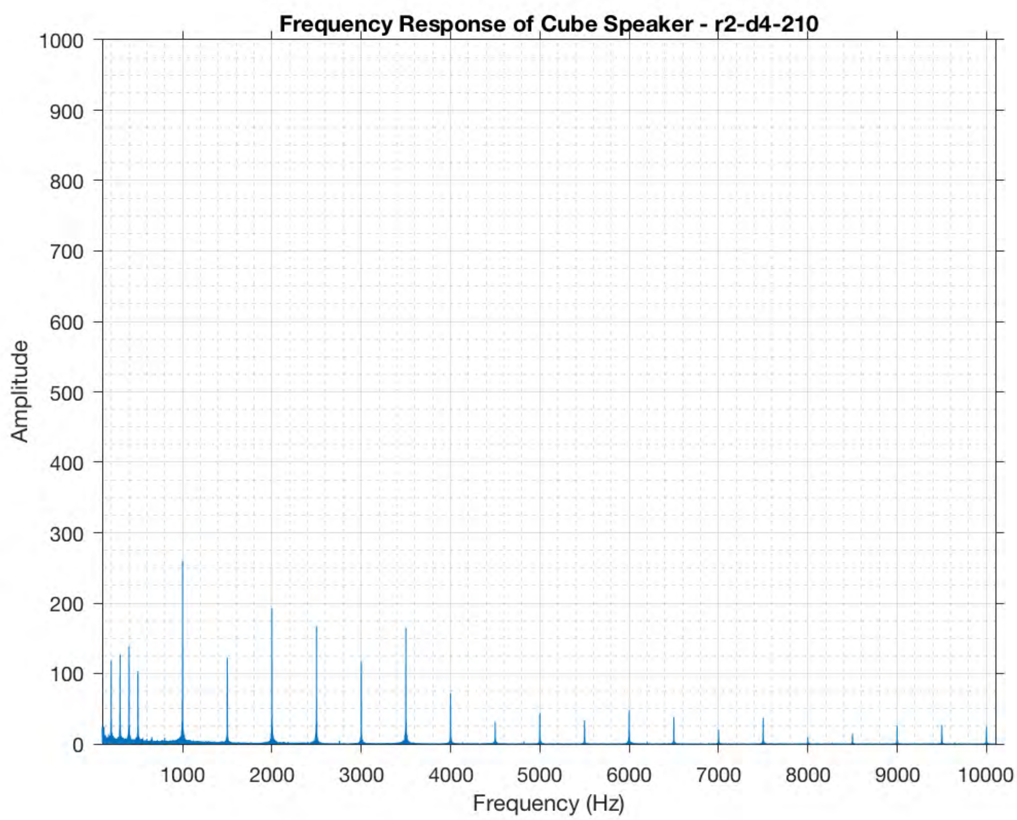

Frequency Response of Cube Speaker - r2-d4-60
Amplitude
Frequency (Hz)
0
100
200
300
400
500
600
700
800
900
1000
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

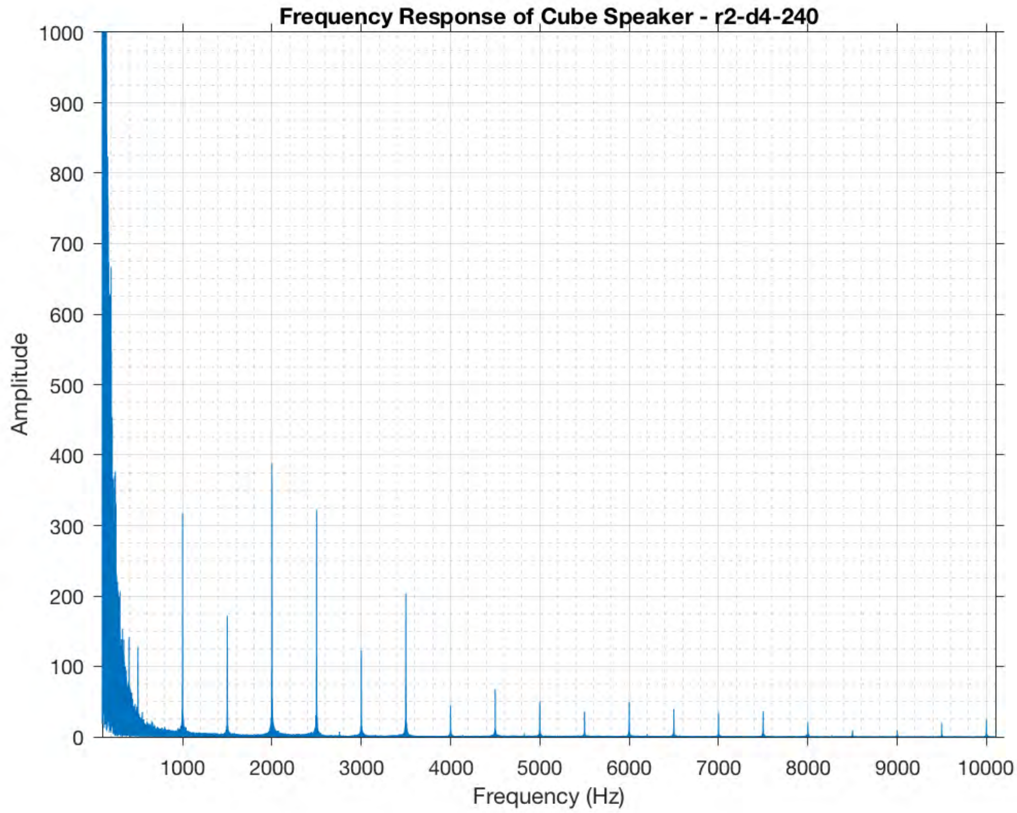

Frequency Response of Cube Speaker - r2-d4-90
Amplitude
Frequency (Hz)
0
100
200
300
400
500
600
700
800
900
1000
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

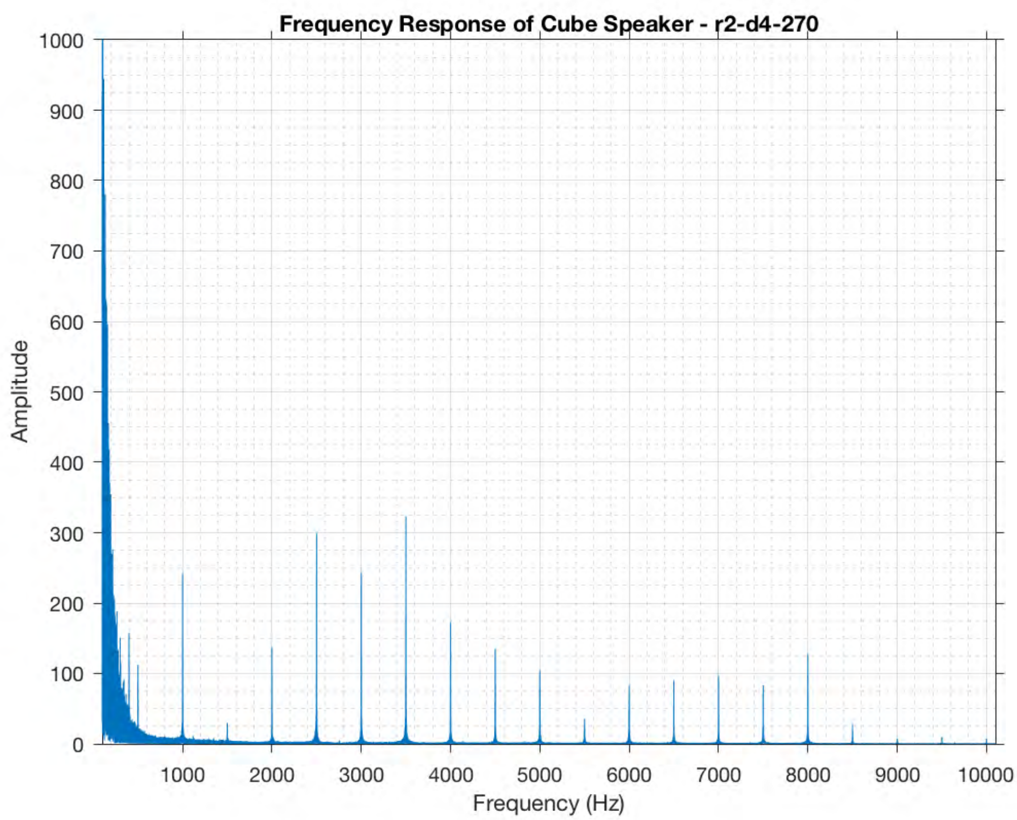

Frequency Response of Cube Speaker - r2-d4-120
Amplitude
Frequency (Hz)
0
100
200
300
400
500
600
700
800
900
1000
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

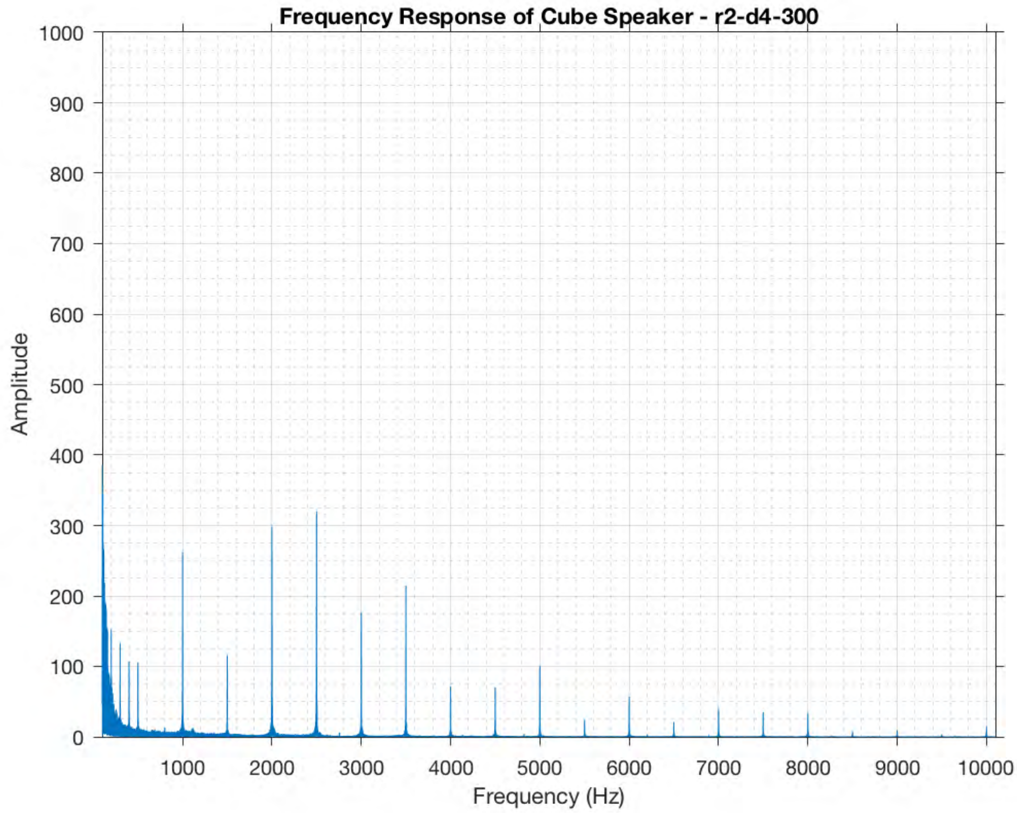

Frequency Response of Cube Speaker - r2-d4-150
Amplitude
Frequency (Hz)
0
100
200
300
400
500
600
700
800
900
1000
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000


Frequency Response of Cube Speaker - r2-d4-180
Amplitude
Frequency (Hz)
1000
900
800
700
600
500
400
300
200
100
0
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

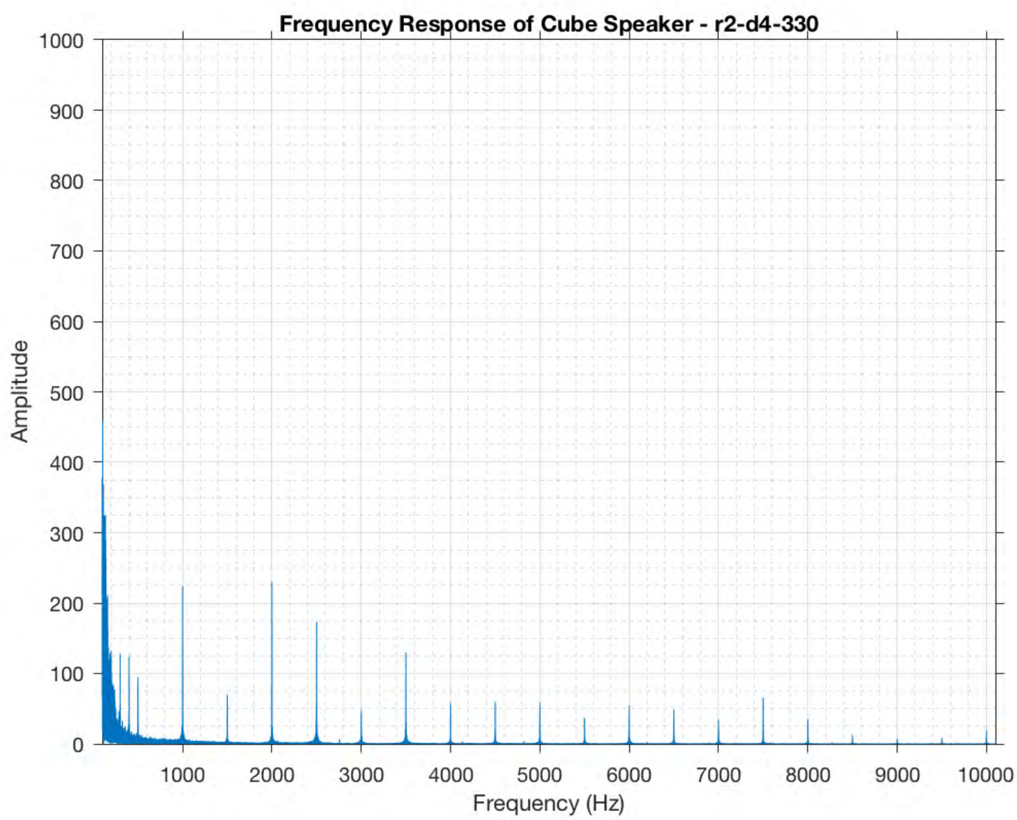

Frequency Response of Cube Speaker - r2-d4-210
Amplitude
Frequency (Hz)
1000
900
800
700
600
500
400
300
200
100
0
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000


Frequency Response of Cube Speaker - r2-d4-240
Amplitude
Frequency (Hz)
0
100
200
300
400
500
600
700
800
900
1000
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

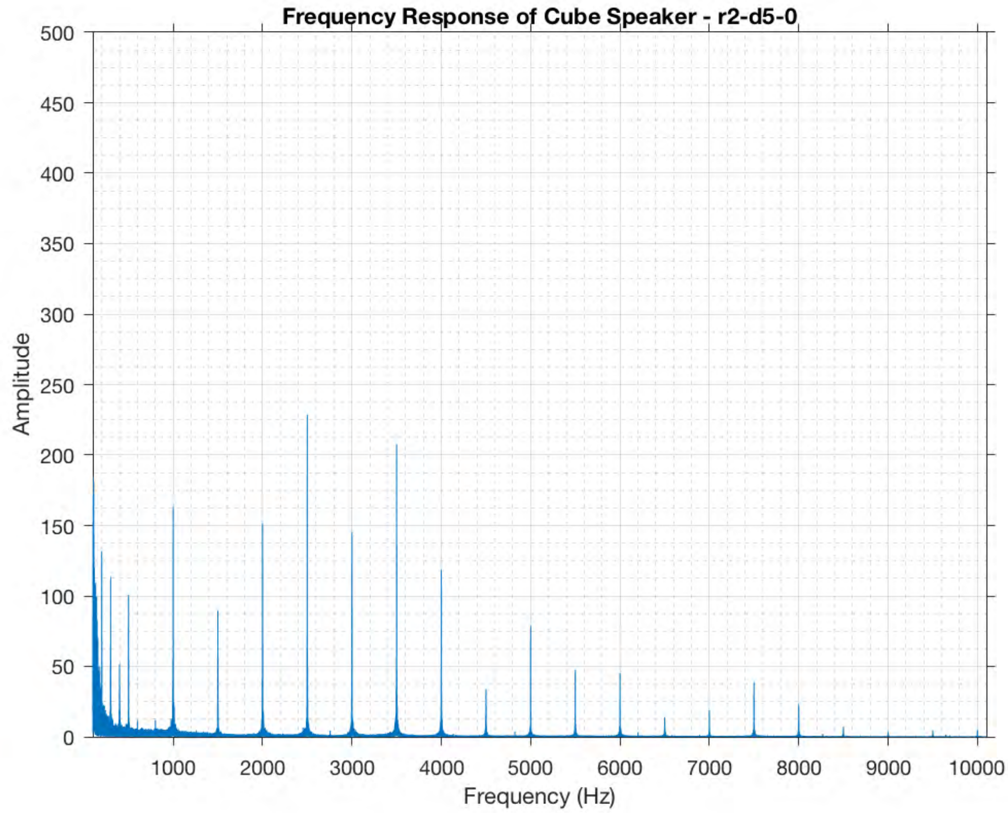

Frequency Response of Cube Speaker - r2-d4-270
Amplitude
Frequency (Hz)
0
100
200
300
400
500
600
700
800
900
1000
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

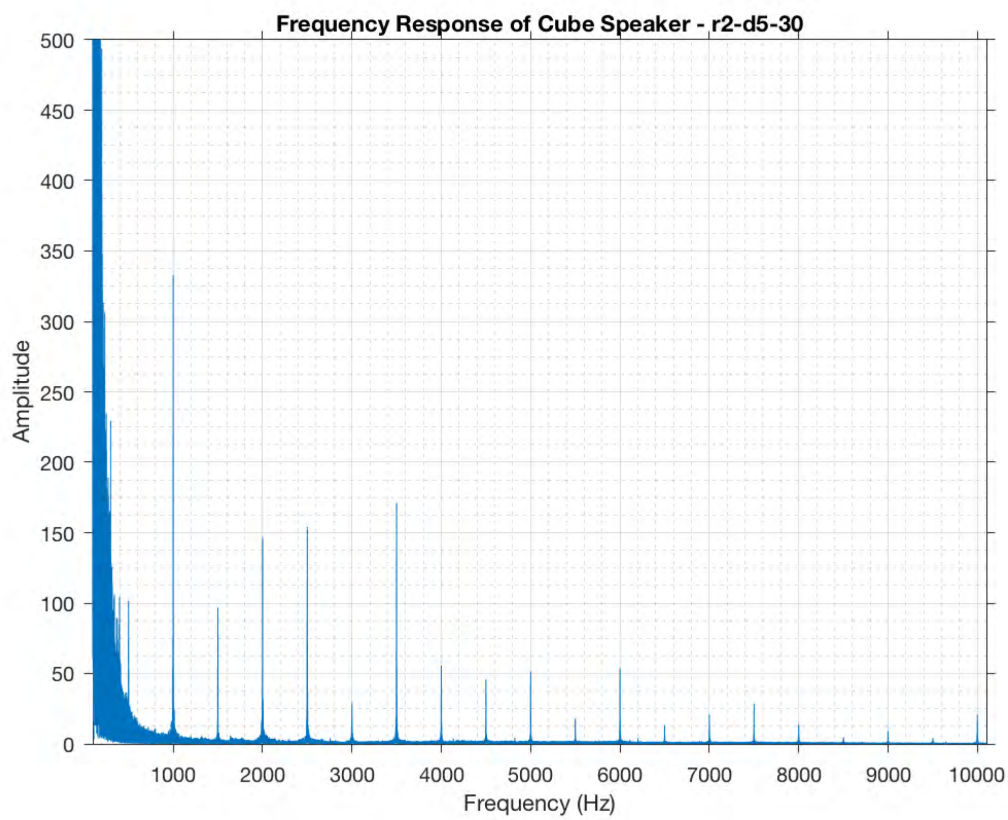


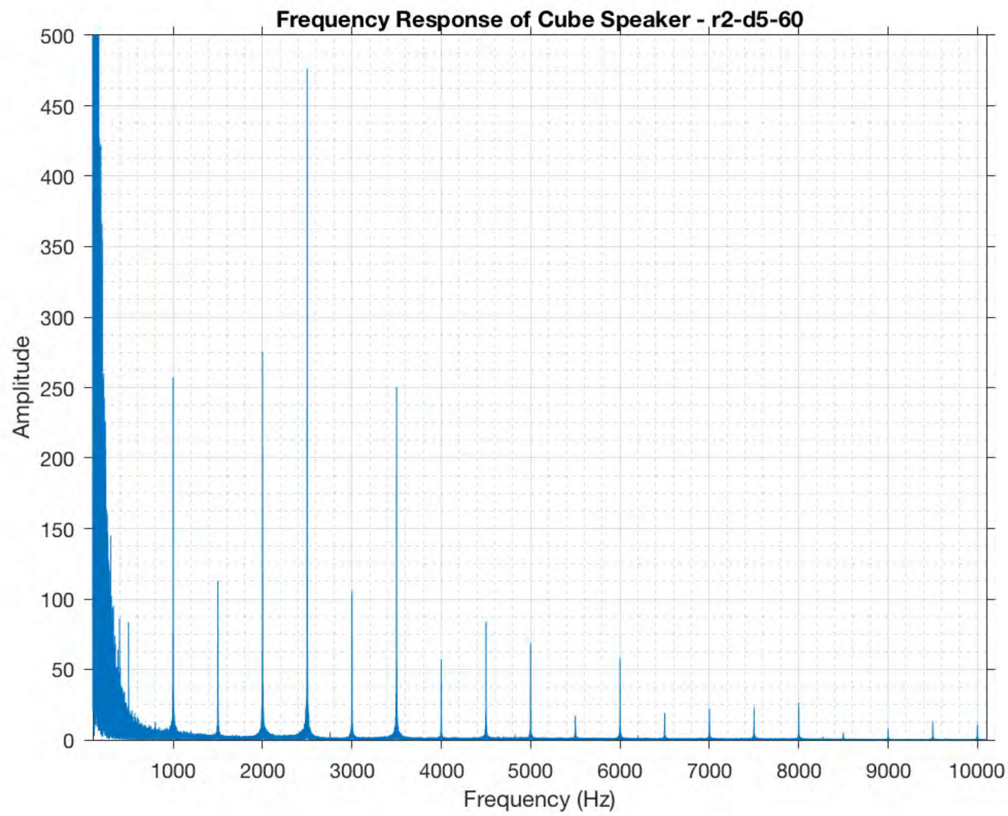
Frequency Response of Cube Speaker - r2-d4-300
Amplitude
Frequency (Hz)
1000
900
800
700
600
500
400
300
200
100
0
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

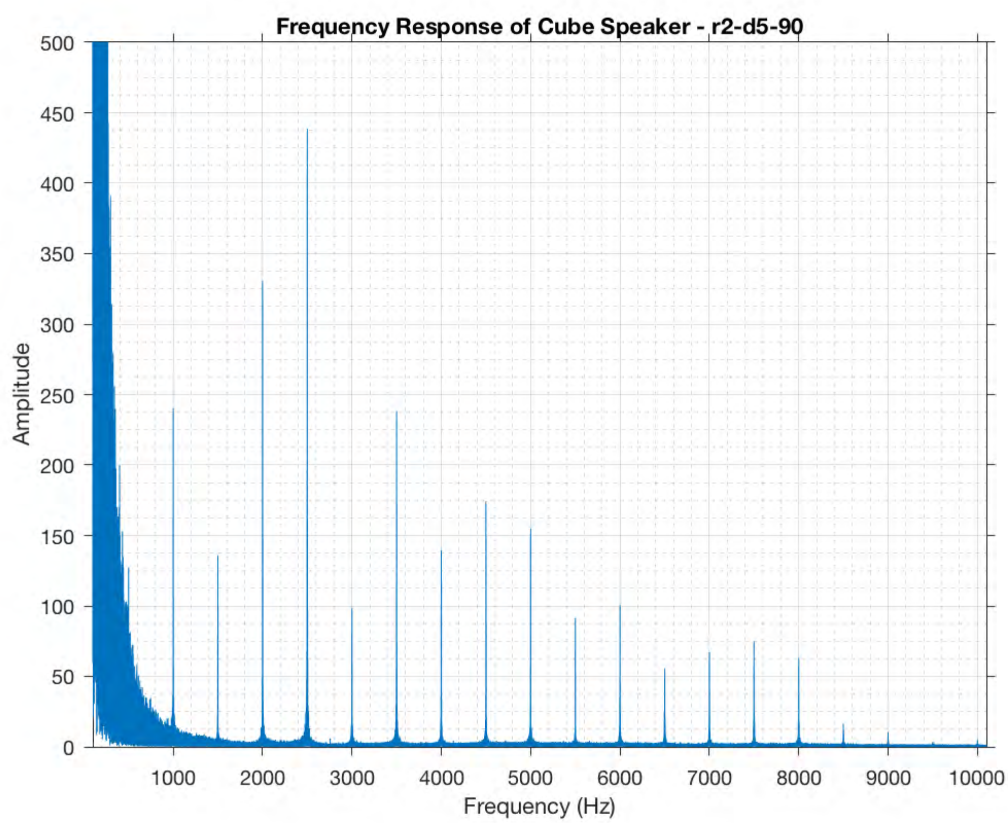


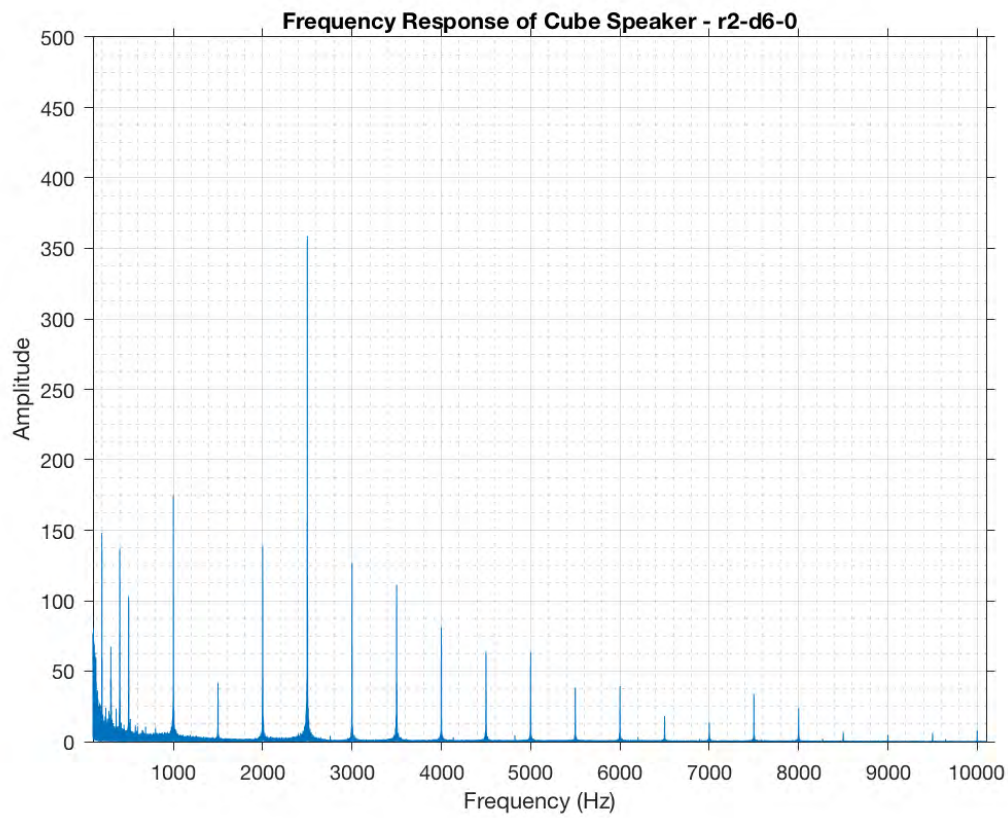
Frequency Response of Cube Speaker - r2-d4-330
Amplitude
Frequency (Hz)
1000
900
800
700
600
500
400
300
200
100
0
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

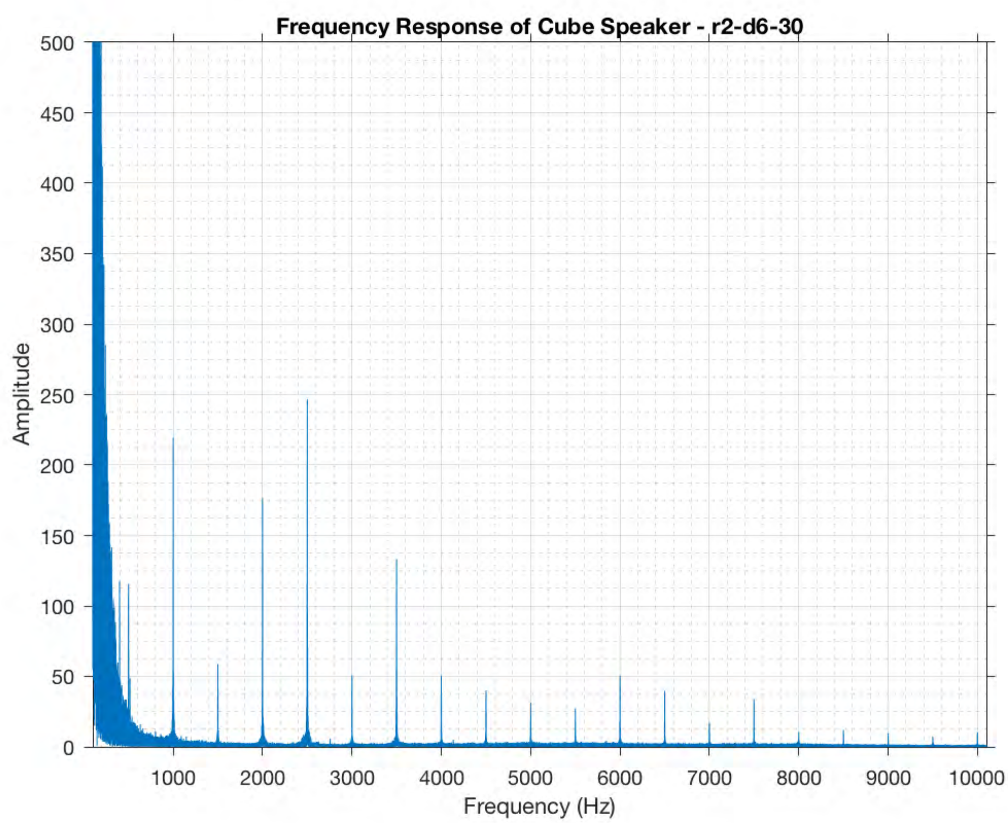

Frequency Response of Cube Speaker - r2-d5-0
Amplitude
Frequency (Hz)
500
450
400
350
300
250
200
150
100
50
0
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

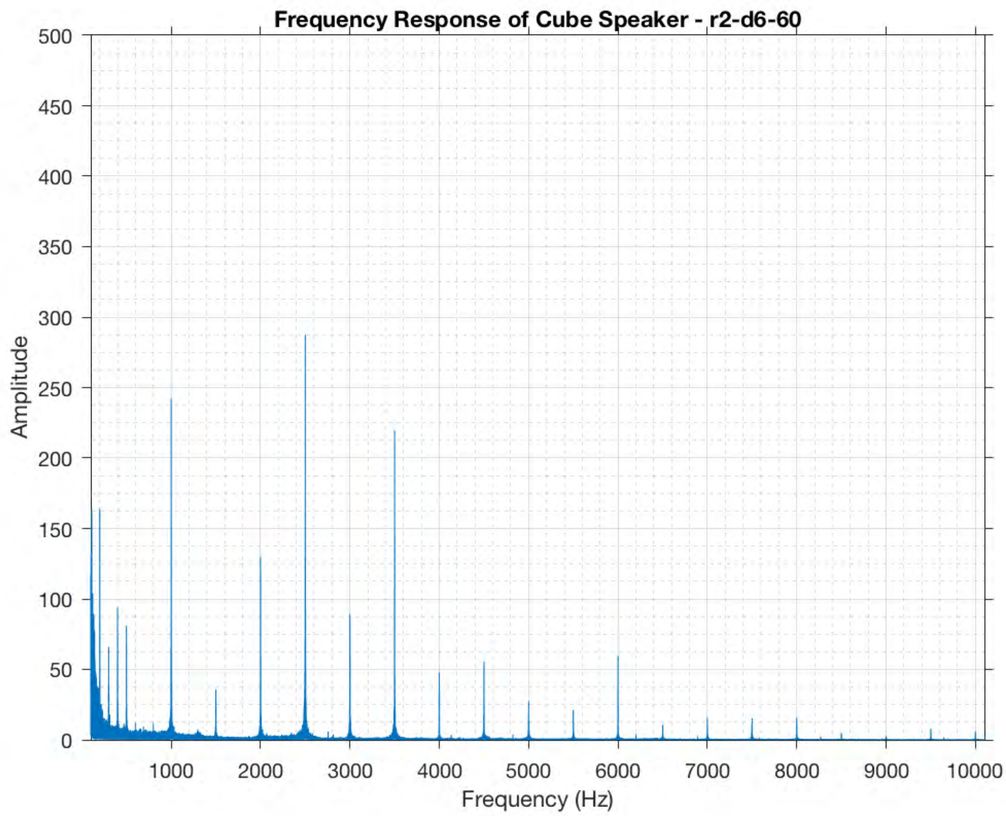

Frequency Response of Cube Speaker - r2-d5-30
Amplitude
Frequency (Hz)
500
450
400
350
300
250
200
150
100
50
0
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000

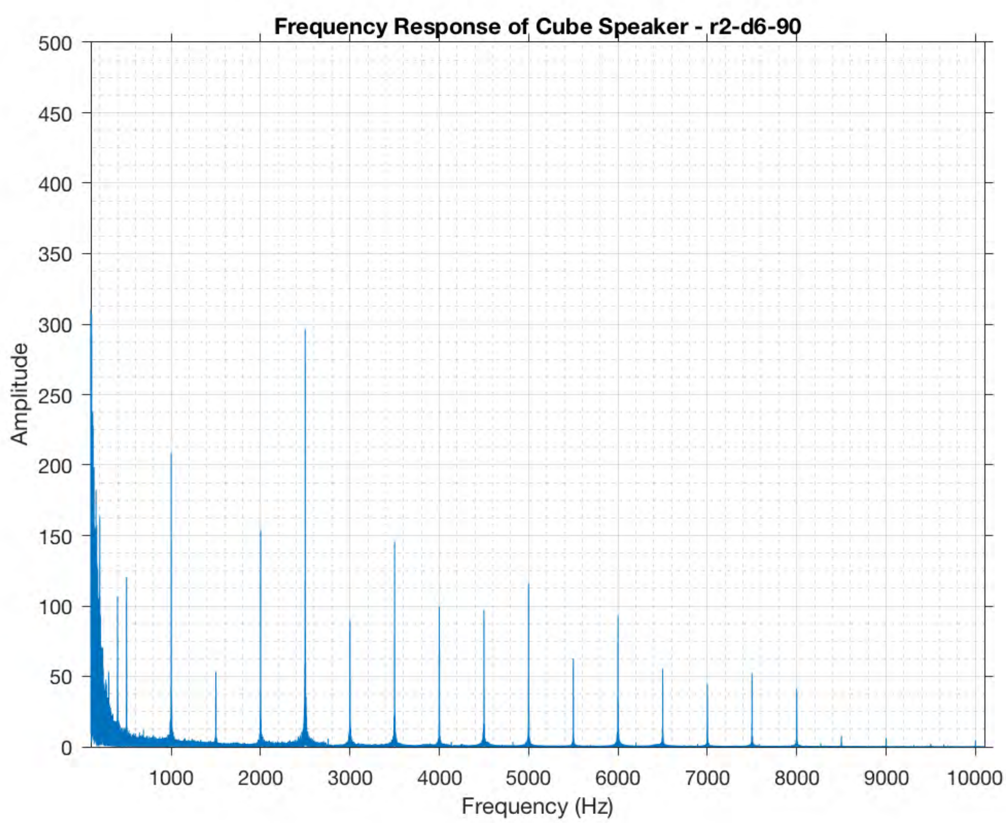

Frequency Response of Cube Speaker - r2-d5-60
Amplitude
Frequency (Hz)
0
50
100
150
200
250
300
350
400
450
500
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000


Frequency Response of Cube Speaker - r2-d5-90
Amplitude
Frequency (Hz)
0
50
100
150
200
250
300
350
400
450
500
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000


Frequency Response of Cube Speaker - r2-d6-0
Amplitude
Frequency (Hz)
500
450
400
350
300
250
200
150
100
50
0
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000


Frequency Response of Cube Speaker - r2-d6-30
Amplitude
Frequency (Hz)
500
450
400
350
300
250
200
150
100
50
0
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000


Frequency Response of Cube Speaker - r2-d6-60
Amplitude
Frequency (Hz)
500
450
400
350
300
250
200
150
100
50
0
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000


Frequency Response of Cube Speaker - r2-d6-90
Amplitude
Frequency (Hz)
500
450
400
350
300
250
200
150
100
50
0
1000
2000
3000
4000
5000
6000
7000
8000
9000
10000