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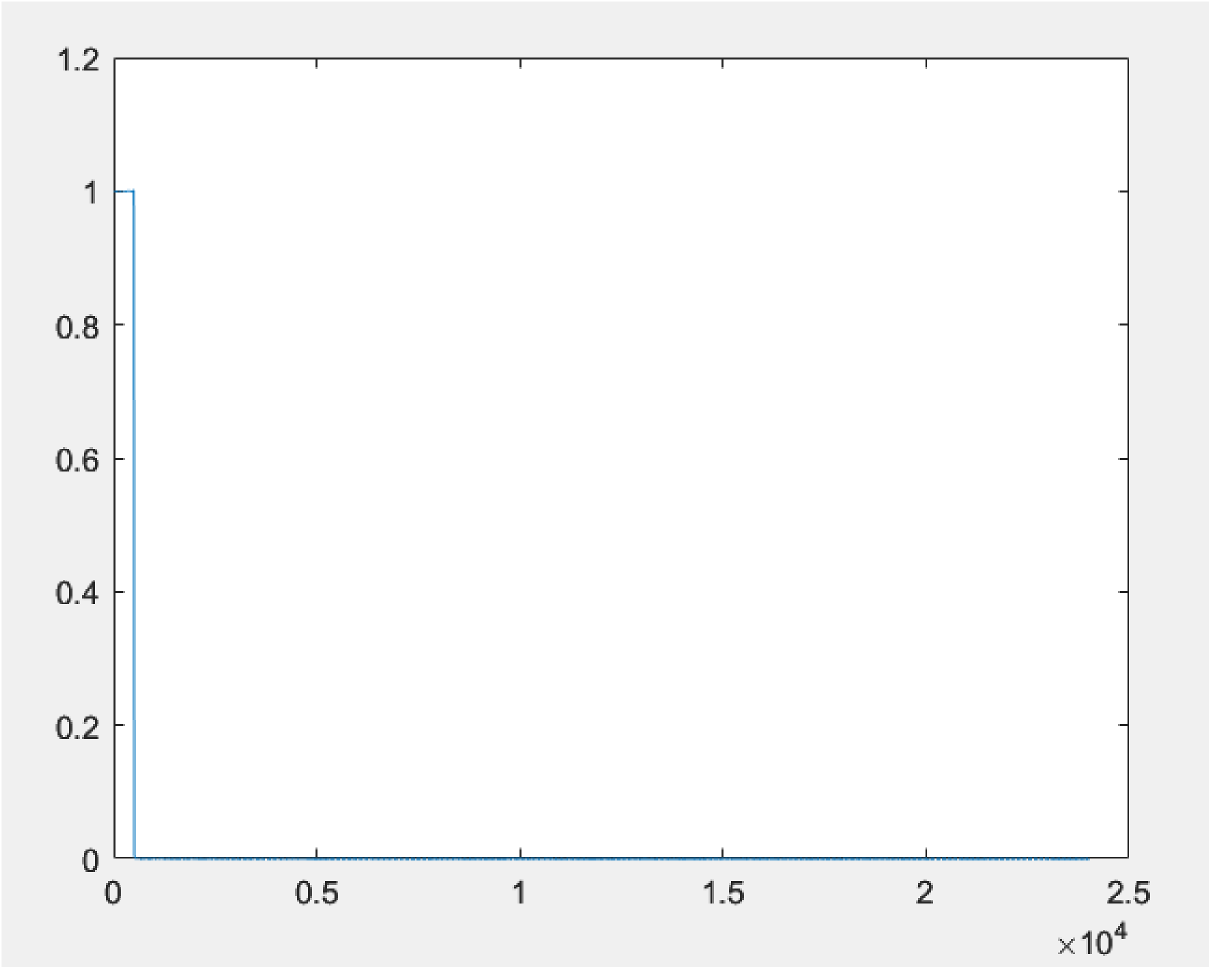
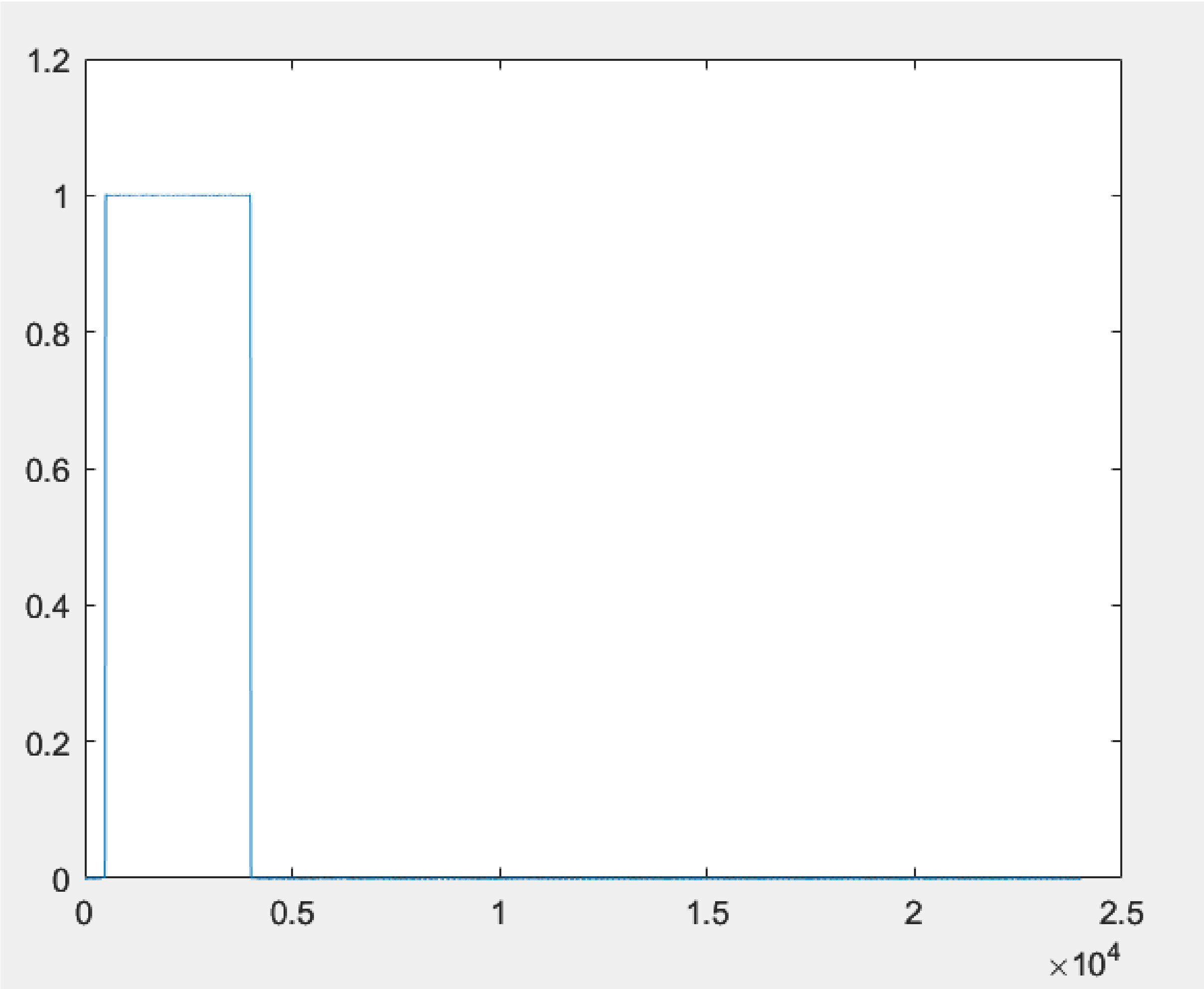
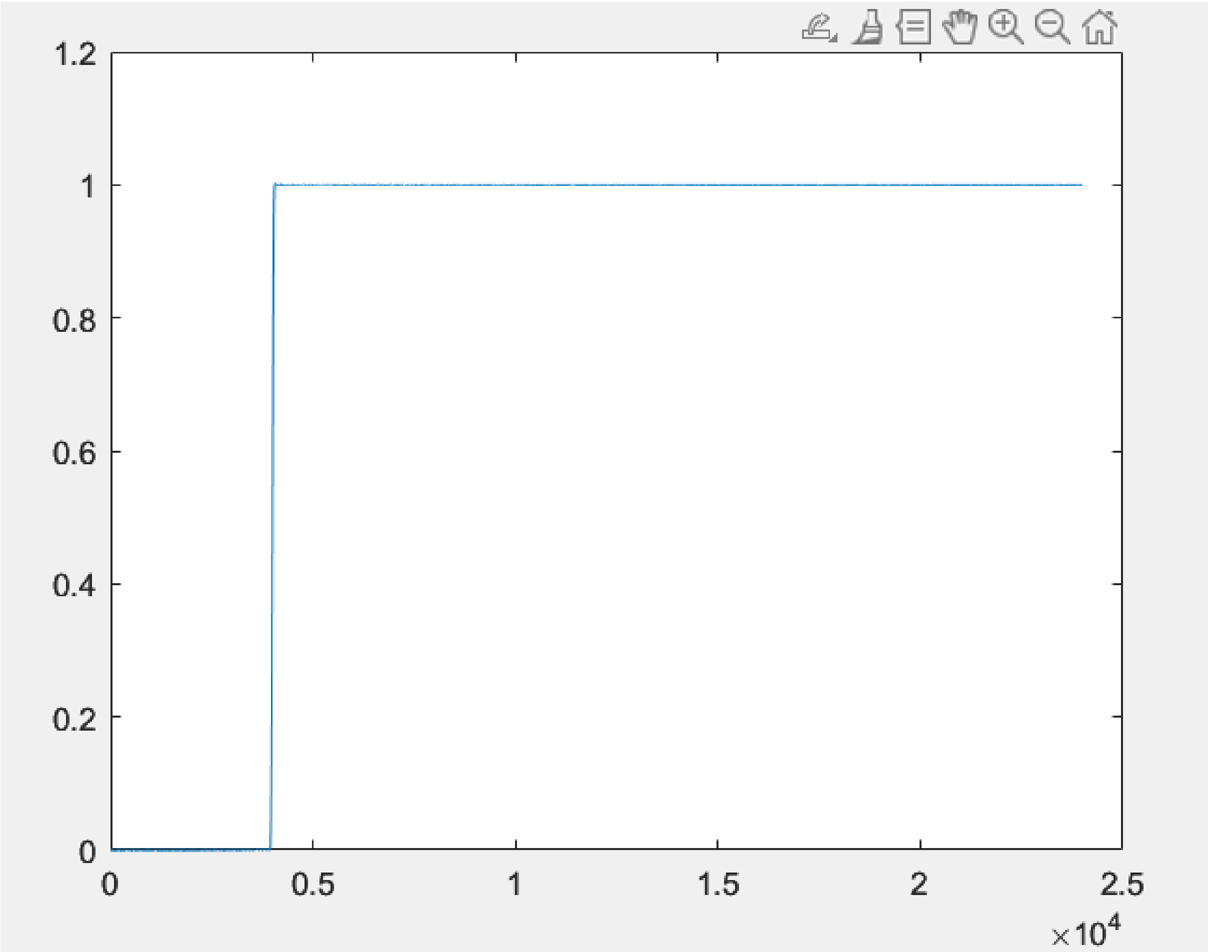
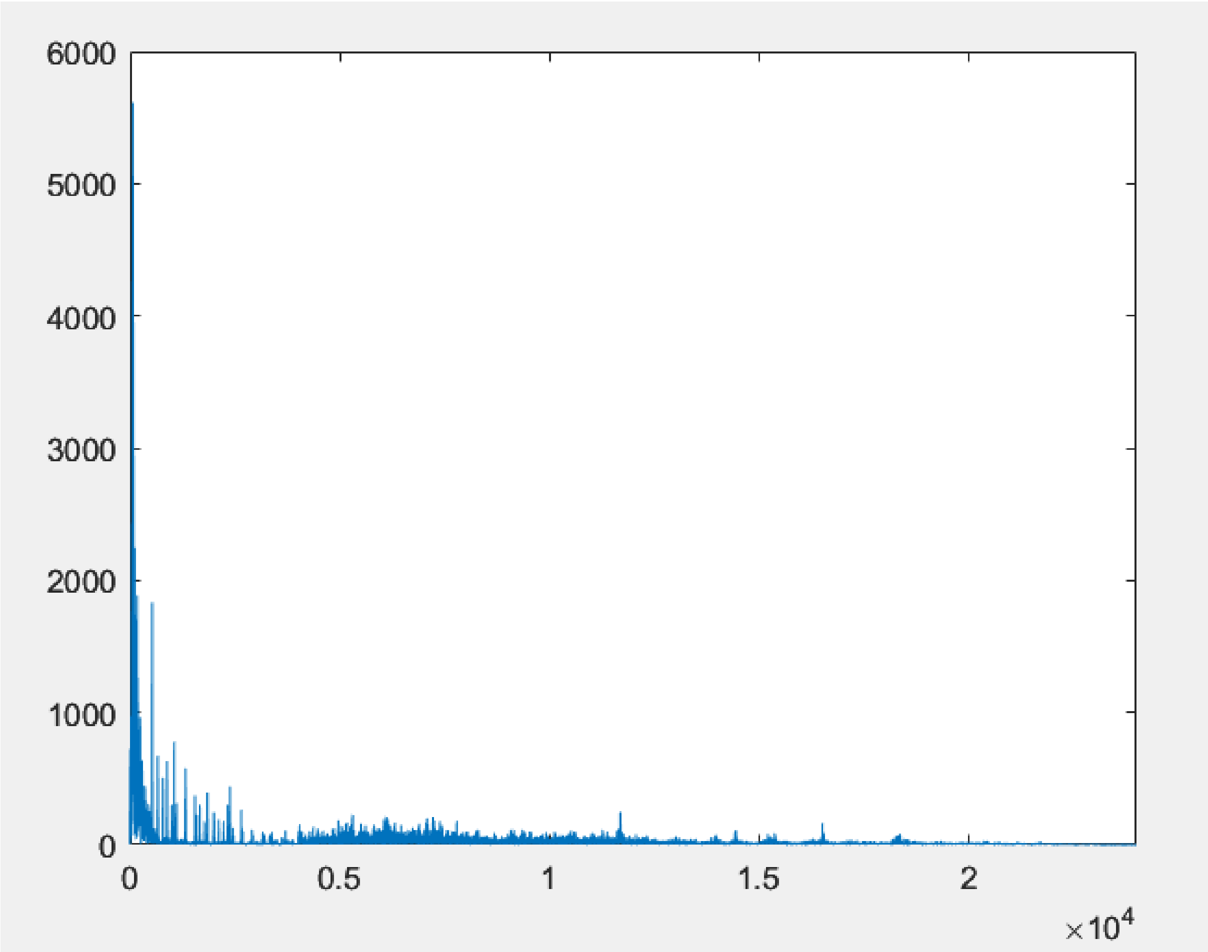
1. I used window-based fir filtering method fir1. It takes 3 parameters in the code which are filter order, normalized cutoff, and filter type. As in described in description, filter types are lowpass for drum, bandpass for piano and highpass for cymbal. Normalized cutoff depends on sample frequency and cutoff frequency. Filter order is a value that I increased until I get sufficient audio. It should not be increased much more for computer health. https://www.mathworks.com/help/signal/ref/fir1.html is useful to learn and use fir1.

2. 

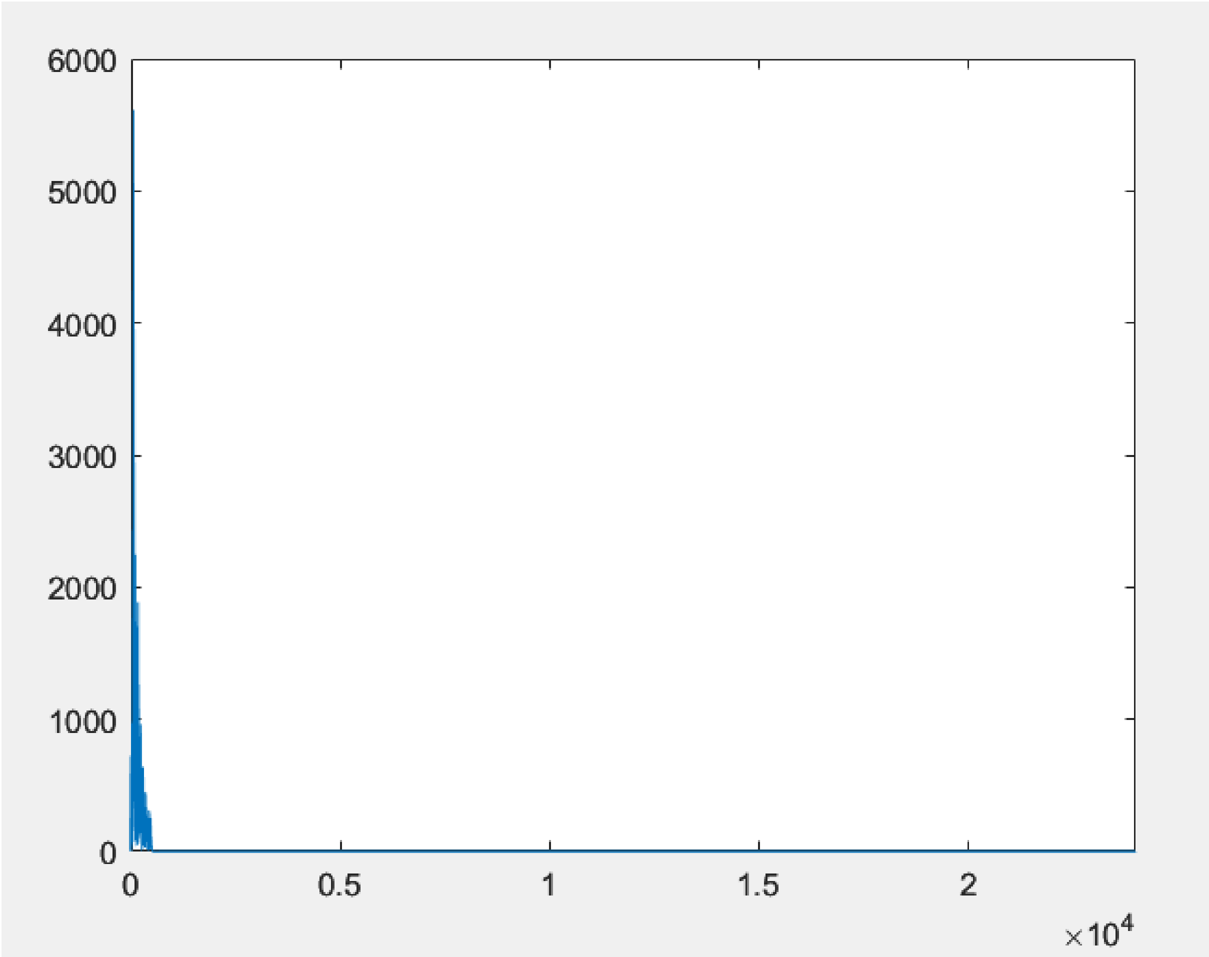

1.2
1
0.8
0.6
0.4
0.2
0
0
0.5
1
1.5
2
2.5
×10^4

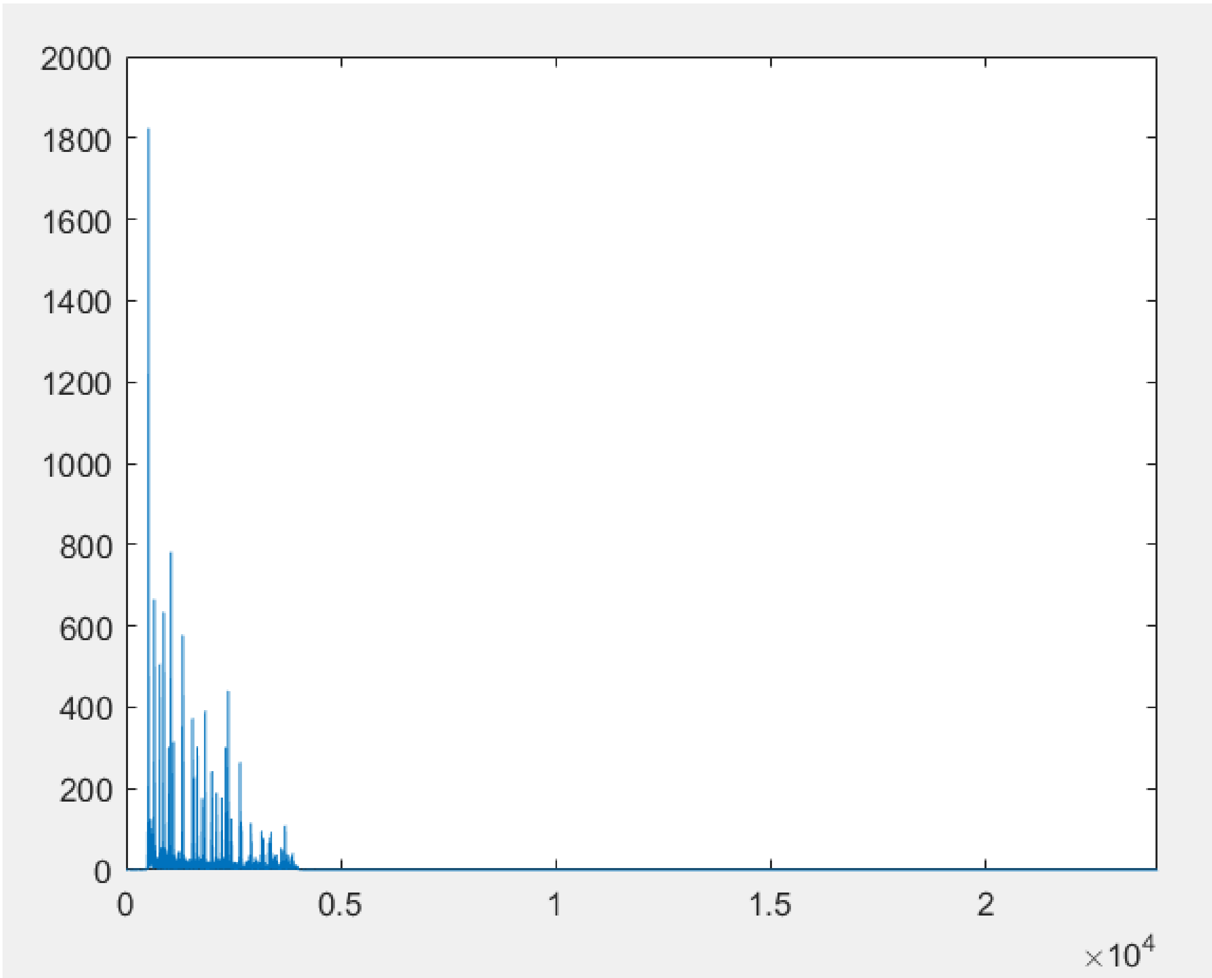


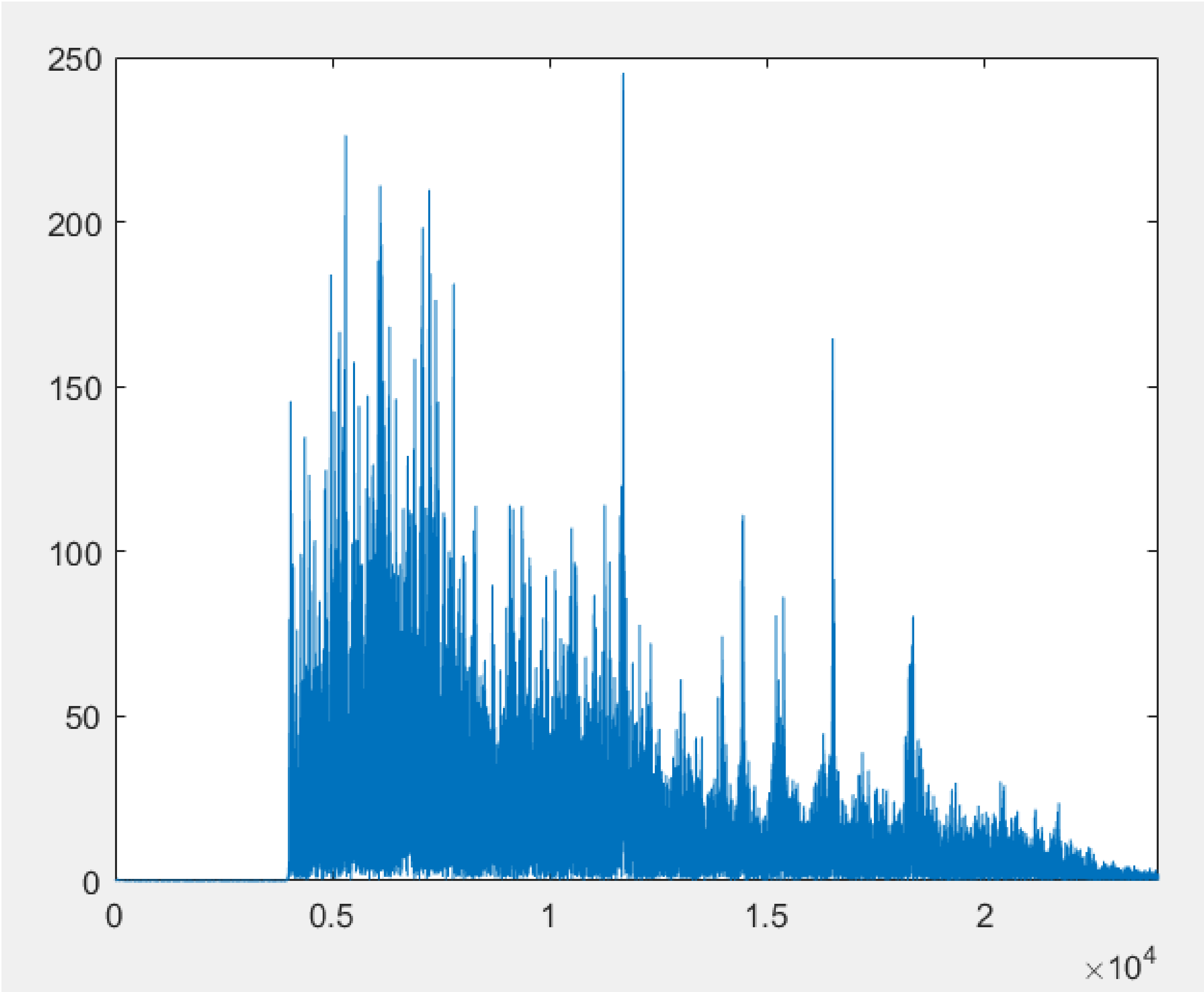
3. First is drum's frequency response of magnitude, second is piano's frequency response and third is symbal's frequency response. First plot shows 1 until 500 Hz which can be thought as it accepts until 500 Hz. Second plot shows 1 between 500 Hz and 4000 Hz. Third plot shows 1 after 4000 Hz. Filters cut frequency values in audio file. I think drum sound has more frequency than 500 in some of the audio, so bandpass filter does not completely work. However, symbal voice is completely different frequencies from the others, so it can be filtered better.
4.



6000
5000
4000
3000
2000
1000
0
0
0.5
1
1.5
2
$\times 10^4$


6000
5000
4000
3000
2000
1000
0
0
0.5
1
1.5
2
$\times 10^4$


2000
1800
1600
1400
1200
1000
800
600
400
200
0
0
0.5
1
1.5
2
$\times 10^4$


250
200
150
100
50
0
0
0.5
1
1.5
2
×10^4