

Math 240 (E1): Scientific Computation

Final Solution Code with Steps Implemented

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Steps Taken to Filter Signal:

1. Load the data by `audioread()`.
2. Assign variable `S` to the sound file.
3. Look at pressure vs. time and notice the interference in amplitude when unwanted tones are played.
4. Perform Fast Fourier Transform to move to frequency domain.
5. Plot power vs. frequency and notice the large spikes around certain frequencies. These are the tones which need to be removed.
6. The filtering process iterates through and removes any of those large spikes in frequency. I arrived at 2500 iterations from some experimentation, and 2500 seemed to be the optimal choice for minimizing computation time while maximizing tone removal.
7. Plot the new power vs. frequency of the filtered data in the frequency domain.
8. Perform inverse Fourier Transform to arrive back in time domain.
9. Plot the new pressure vs time signal with tones removed.
10. Write the filtered sound wave by using `audiowrite()`.

```

1  audioinfo('prob.wav')
2
3  S=audioread('prob.wav');
4
5  figure(1)
6  plot(S)
7  title('Unfiltered Wave')
8  xlabel('Sample Number');
9  ylabel('Amplitude');
10
11  FSF=fft(S);
12
13  N=length(FSF);
14  power = abs(FSF(1:N)).^2;
15  freq = (1:N)/N;
16
17  figure(2)
18  plot(FSF,'ro')
19  title('Fourier Coefficients in the Complex Plane');
20  xlabel('Real Axis');
21  ylabel('Imaginary Axis');
22
23  figure(3)
24  plot(freq,power)
25  xlabel('Cycles/Sample Interval')
26  ylabel('Power');
27  title('Periodogram')
28
29
30  index=find(power==max(power));
31  period=1./freq;
32  %Herz=44100/period(index)
33  mainPeriodStr=num2str(period(index));
34  figure(4)
35  plot(period(index),power(index),'r.', 'MarkerSize',25);
36  text(period(index)+0.01*period(index),power(index),['Period = ...',mainPeriodStr]);
37
38  tic
39  for i = 1:2500
40      power = abs(FSF(1:N)).^2;
41      index=find(power==max(power));
42      FSF(index(1)) = 0;
43      FSF(index(2))=0;
44  end
45  toc
46
47  power = abs(FSF(1:N)).^2;
48  plot(freq,power)
49
50  INFT = ifft(FSF);
51  figure(5)
52  plot(INFT)
53  title('Filtered Wave')
54  xlabel('Sample Number');
55  ylabel('Amplitude');
56

```

```
57 audiowrite('FilteredSound.wav', INFT, 44100)
```

Note: The time to filter the data on the last trail run of the M-File was
61.095220 seconds