



## **Ahsanullah University of Science & Technology**

**Department of Electrical and Electronic Engineering**

### **Project Report**

Course No : EEE 3218  
Course Title : Digital Signal Processing Lab

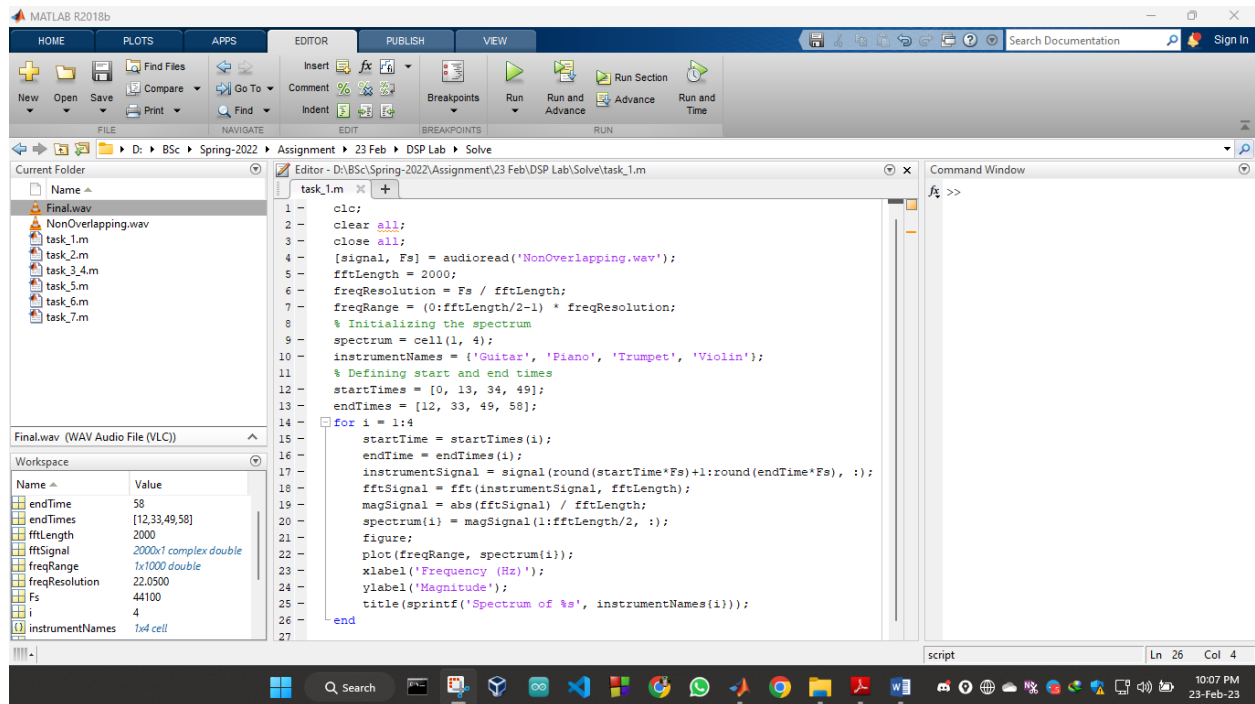
Submission Date : 23-02-2023

### **Submitted by**

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ID : 190205184  
Section : C2  
Year : 3<sup>rd</sup>  
Semester : 2<sup>nd</sup>

## Task 1:

Code:



The image shows the MATLAB R2018b environment. The main window displays a script named 'task\_1.m' with the following code:

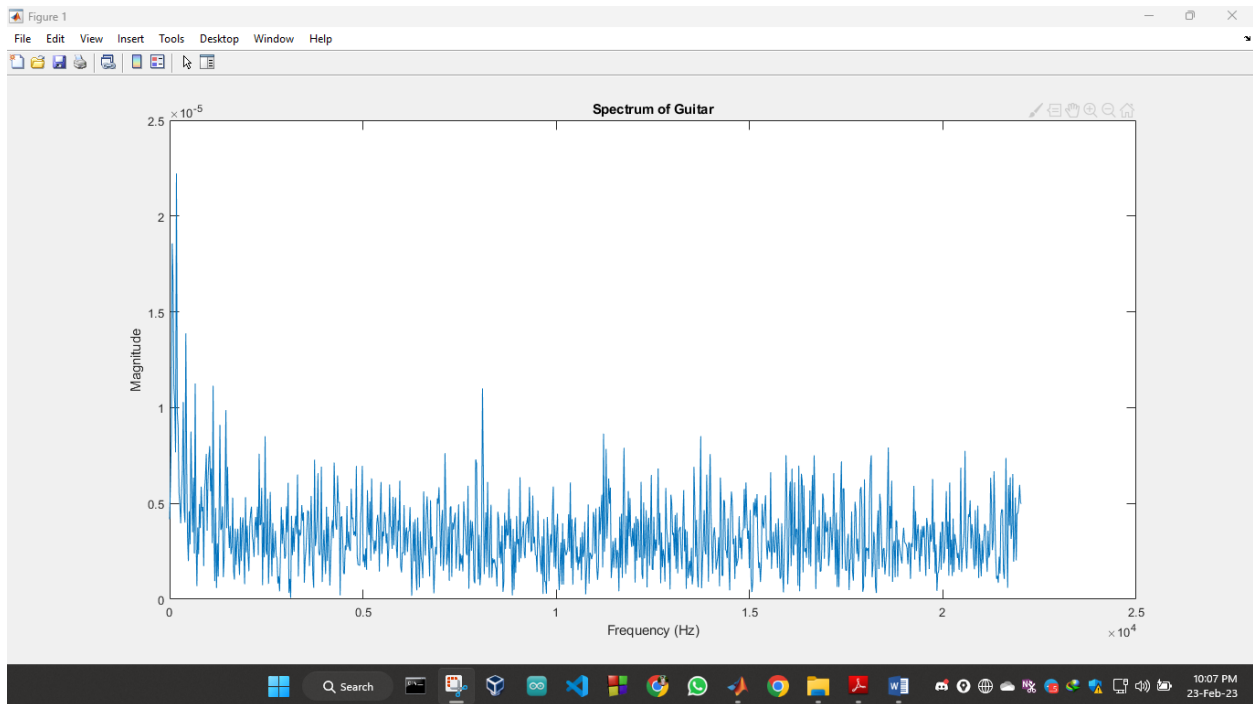
```
1 - clc;
2 - clear all;
3 - close all;
4 - [signal, Fs] = audioread('NonOverlapping.wav');
5 - fftLength = 2000;
6 - freqResolution = Fs / fftLength;
7 - freqRange = (0:fftLength/2-1) * freqResolution;
8 - % Initializing the spectrum
9 - spectrum = cell(1, 4);
10 - instrumentNames = {'Guitar', 'Piano', 'Trumpet', 'Violin'};
11 - % Defining start and end times
12 - startTimes = [0, 13, 34, 49];
13 - endTimes = [12, 33, 49, 58];
14 - for i = 1:4
15 -     startTime = startTimes(i);
16 -     endTime = endTimes(i);
17 -     instrumentSignal = signal(round(startTime*Fs)+1:round(endTime*Fs), :);
18 -     fftSignal = fft(instrumentSignal, fftLength);
19 -     magSignal = abs(fftSignal) / fftLength;
20 -     spectrum(i) = magSignal(1:fftLength/2, :);
21 -     figure;
22 -     plot(freqRange, spectrum(i));
23 -     xlabel('Frequency (Hz)');
24 -     ylabel('Magnitude');
25 -     title(sprintf('Spectrum of %s', instrumentNames(i)));
26 - end
27
```

The left sidebar shows the 'Current Folder' with files: 'Final.wav', 'NonOverlapping.wav', 'task\_1.m', 'task\_2.m', 'task\_3\_4.m', 'task\_5.m', 'task\_6.m', and 'task\_7.m'. Below it, the 'Workspace' table lists variables:

Name	Value
endTime	58
endTimes	[12, 33, 49, 58]
fftLength	2000
fftSignal	2000x1 complex double
freqRange	1x1000 double
freqResolution	22.0500
Fs	44100
i	4
instrumentNames	1x4 cell

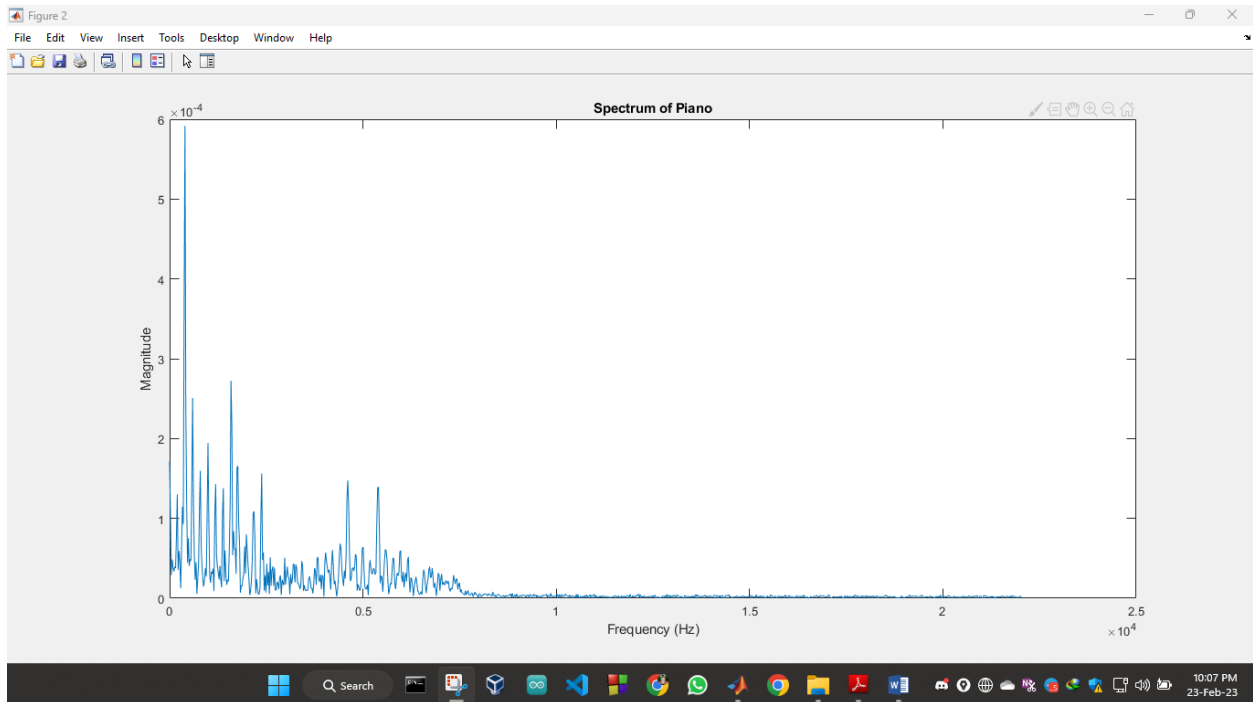
The bottom status bar shows 'script' and 'Ln 26 Col 4'. The system clock in the bottom right corner indicates '10:07 PM 23-Feb-23'.

## Guitar:



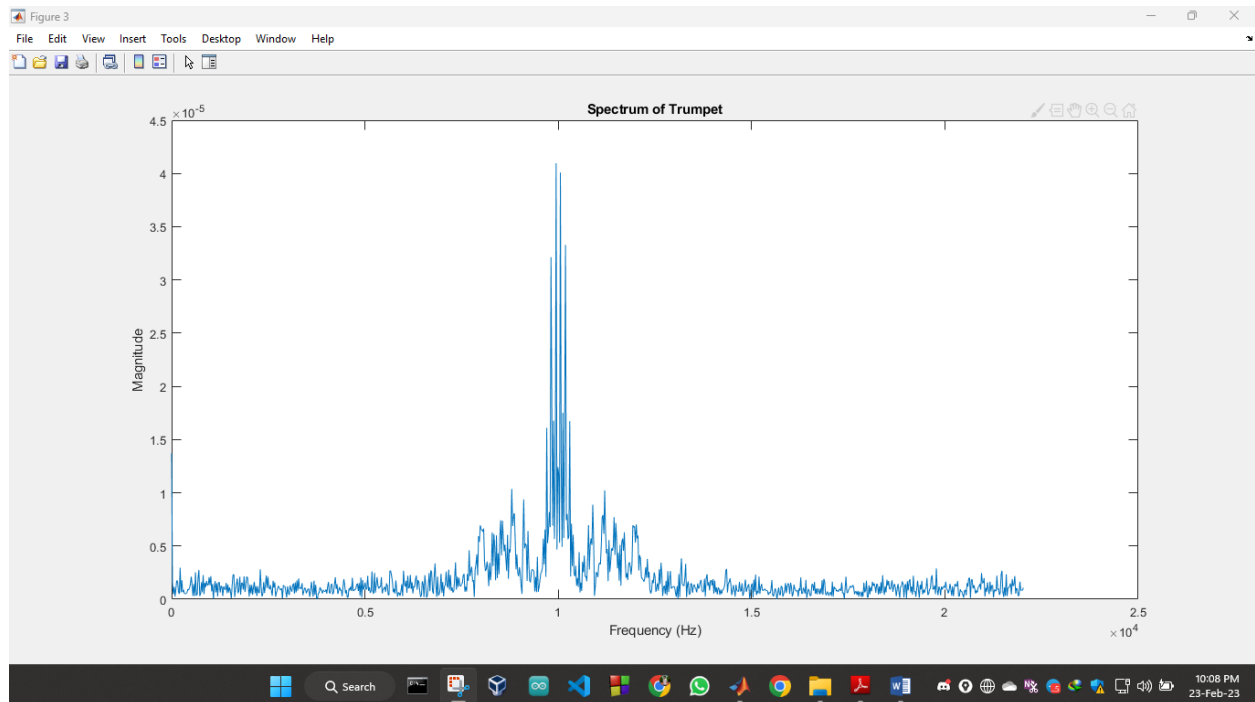
It has a frequency range between 0 Hz to 22010 Hz.

## Piano:



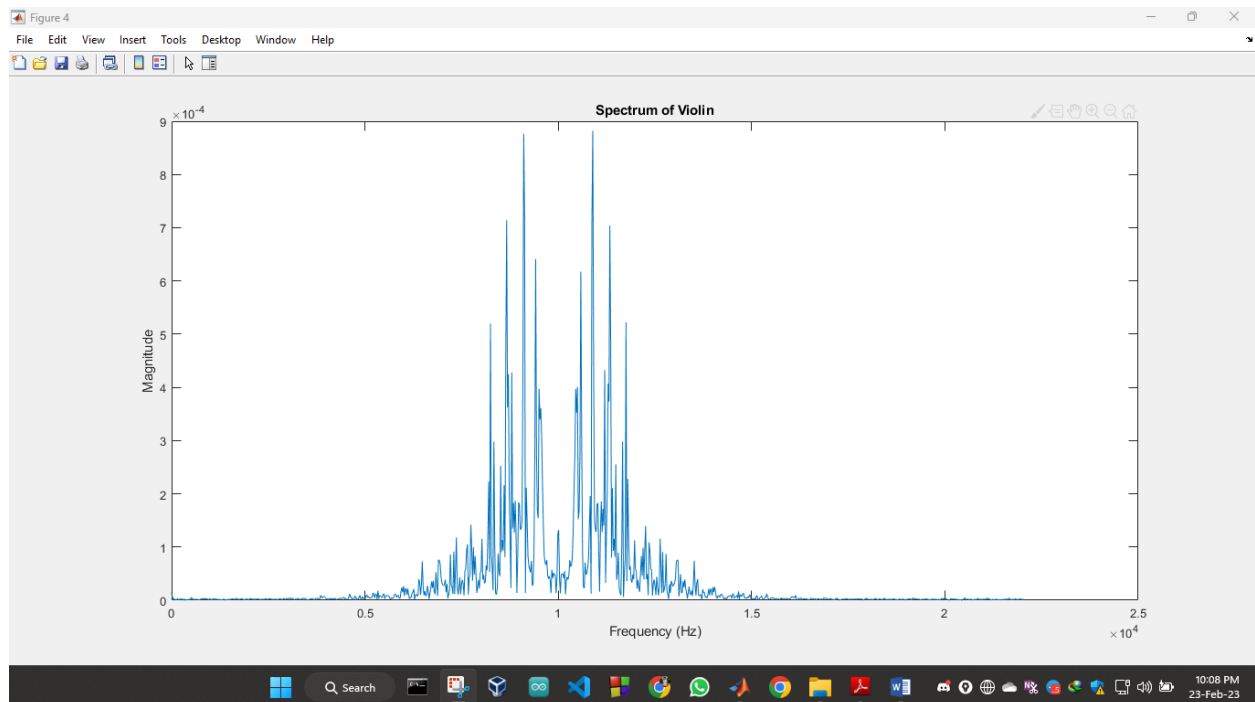
It has a frequency range between 0 Hz to 7629 Hz.

## Trumpet:



It has a frequency range between 0 Hz to 22030 Hz.

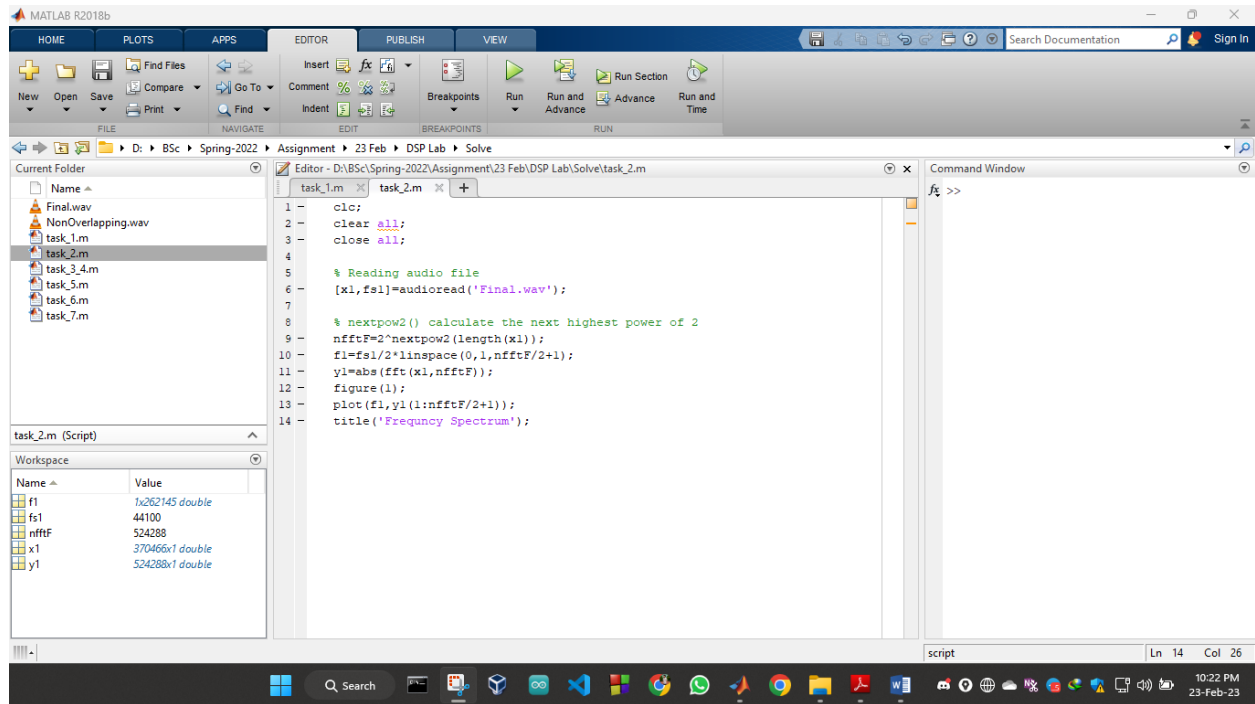
## Violin:



It has a frequency range between 5005 Hz to 16140 Hz.

## Task 2:

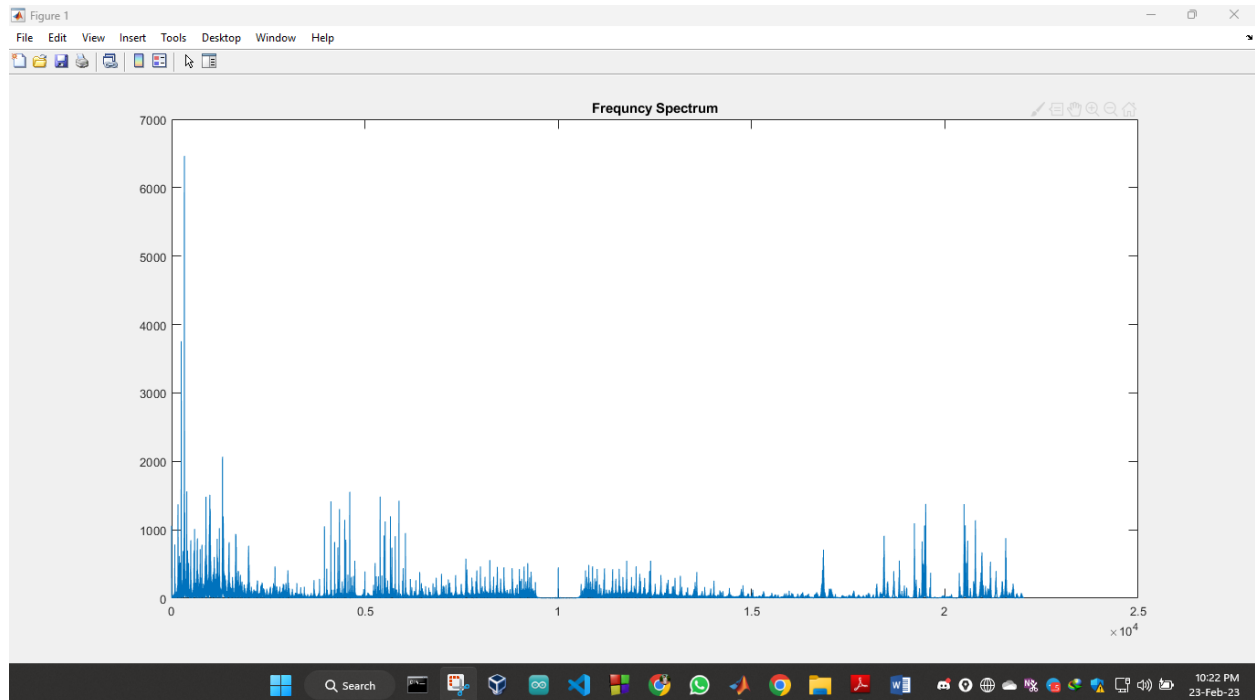
Code:



```
1 - clc;
2 - clear all;
3 - close all;
4
5 % Reading audio file
6 [x1,fs1]=audioread('Final.wav');
7
8 % nextpow2() calculate the next highest power of 2
9 nfftF=2^nextpow2(length(x1));
10 f1=fs1/2* linspace(0,1,nfftF/2+1);
11 y1=abs(fft(x1,nfftF));
12 figure(1);
13 plot(f1,y1(1:nfftF/2+1));
14 title('Frequency Spectrum');
```

Name	Value
f1	1x262145 double
fs1	44100
nfftF	524288
x1	370466x1 double
y1	524288x1 double

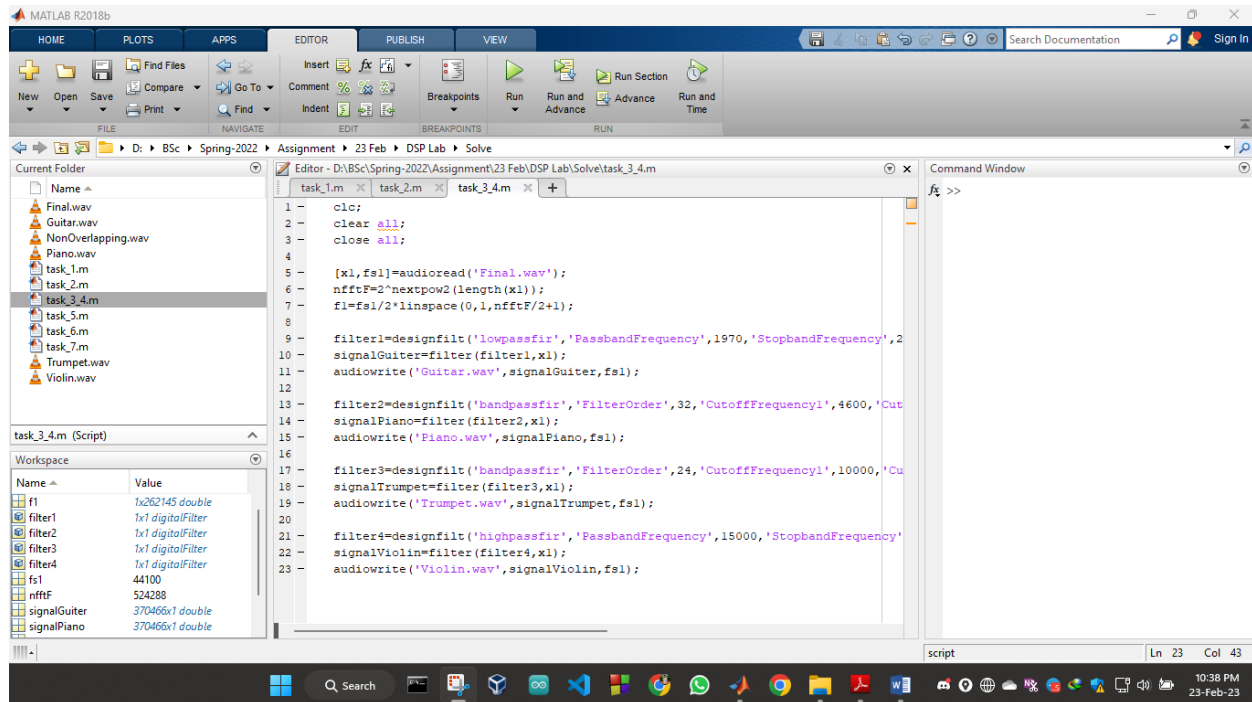
Spectrum:



The frequency range is between 0 Hz to 22050 Hz.

## Task 3 & 4:

Code:



```
1 - clc;
2 - clear all;
3 - close all;
4
5 - [x1,fs1]=audioread('Final.wav');
6 - nfftF=2^nextpow2(length(x1));
7 - f1=fs1/2* linspace(0,1,nfftF/2+1);
8
9 - filter1=designfilt('lowpassfir','PassbandFrequency',1970,'StopbandFrequency',2290,'PassbandRipple',1,'StopbandAttenuation',94);
10 - signalGuitar=filter(filter1,x1);
11 - audiowrite('Guitar.wav',signalGuitar,fs1);
12
13 - filter2=designfilt('bandpassfir','FilterOrder',32,'CutoffFrequency1',4600,'CutoffFrequency2',6100,'PassbandRipple',1,'StopbandAttenuation',77);
14 - signalPiano=filter(filter2,x1);
15 - audiowrite('Piano.wav',signalPiano,fs1);
16
17 - filter3=designfilt('bandpassfir','FilterOrder',24,'CutoffFrequency1',10000,'CutoffFrequency2',11000,'PassbandRipple',1,'StopbandAttenuation',77);
18 - signalTrumpet=filter(filter3,x1);
19 - audiowrite('Trumpet.wav',signalTrumpet,fs1);
20
21 - filter4=designfilt('highpassfir','PassbandFrequency',15000,'StopbandFrequency',14900,'PassbandRipple',1,'StopbandAttenuation',77);
22 - signalViolin=filter(filter4,x1);
23 - audiowrite('Violin.wav',signalViolin,fs1);
```

Name	Value
f1	1x262145 double
filter1	1x1 digitalFilter
filter2	1x1 digitalFilter
filter3	1x1 digitalFilter
filter4	1x1 digitalFilter
fs1	44100
nfftF	524288
signalGuitar	370466x1 double
signalPiano	370466x1 double

**Guitar:** Guitar uses low pass filter with pass band frequency of 1970 Hz, stopband frequency of 2290 Hz, pass band ripple 1, stopband attenuation of 94.

**Piano:** Piano uses band pass filter with order 32, cutoff frequency 4600Hz and 6100 Hz.

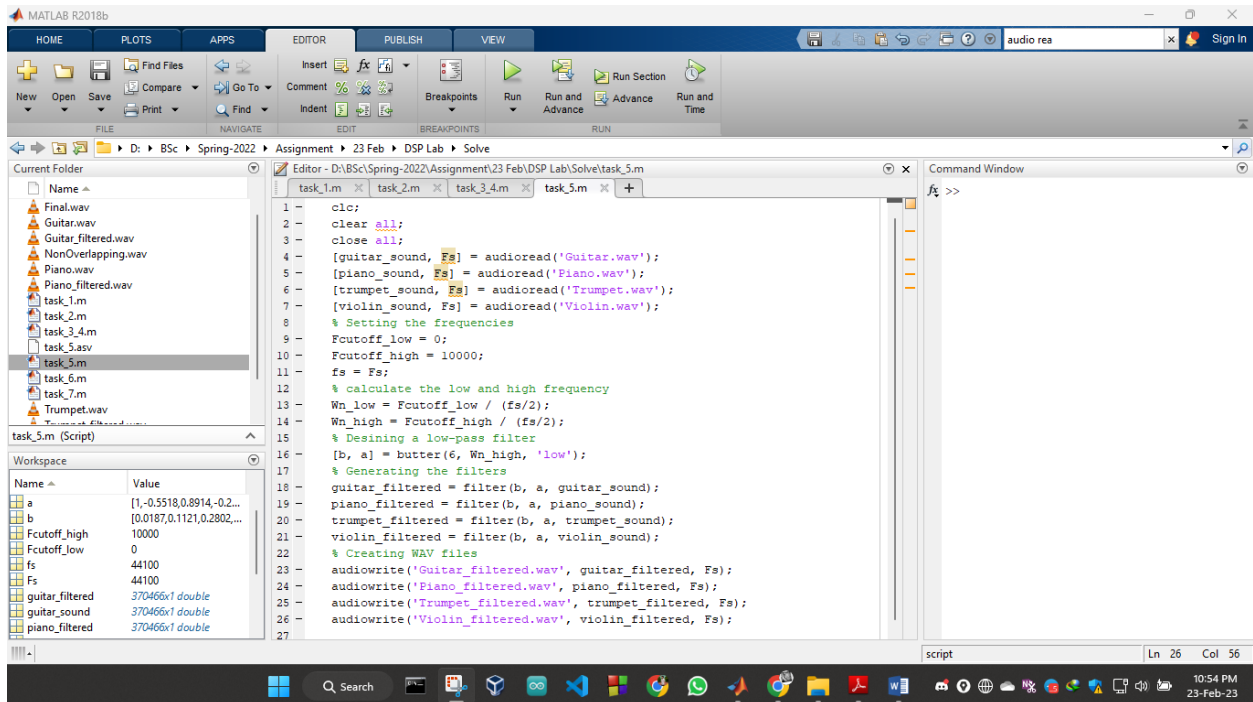
**Trumpet:** Trumpet uses band pass filter with filter order 24, cutoff frequency 10000Hz and 11000 Hz.

**Violin:** Violin uses high pass filter with pass band frequency of 15000 Hz, stopband frequency of 14900 Hz, pass band ripple 1, stopband attenuation of 77.

Extracted files are Guitar.wav, Piano.wav, Trumpet.wav and Violin.wav.

## Task 5:

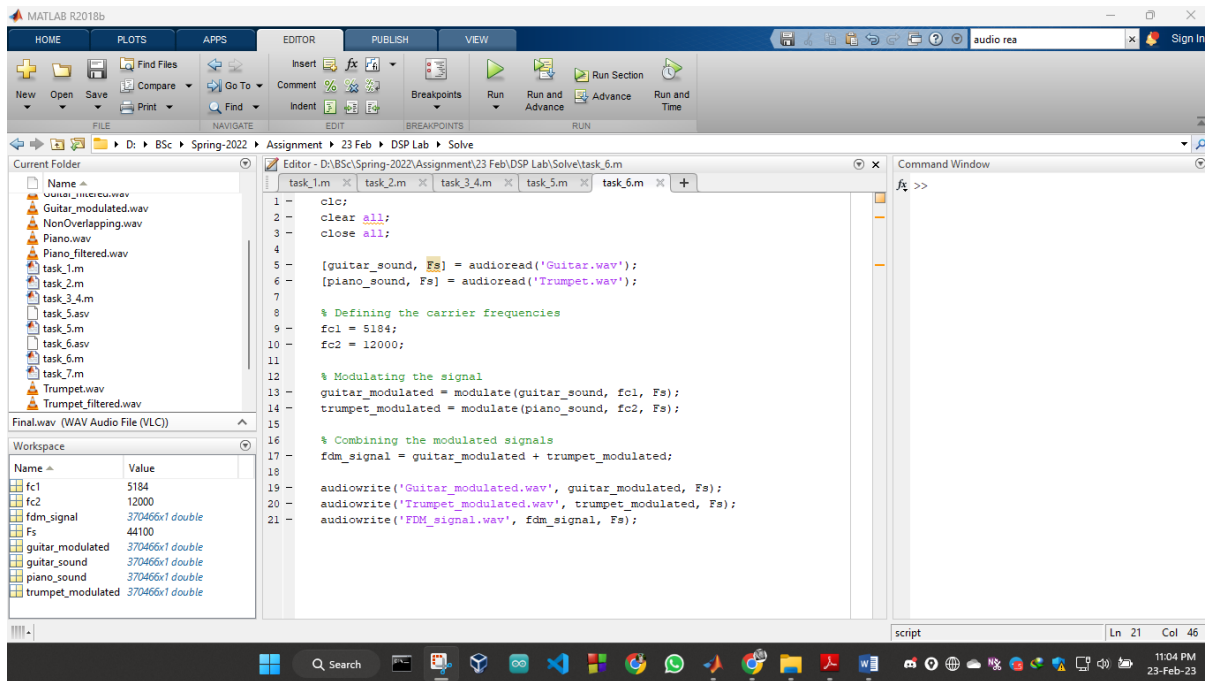
I can pass the individual wav files separately through a channel of bandwidth 0 to 10 kHz by setting the lower frequency to 0 Hz and higher frequency to 10000 Hz and using the low pass filter with butter function with filter order 6.



### Task 6:

Sending guitar and trumpet signals through a 2-channel FDM link using carrier frequency of 5184 Hz and 12000 Hz.

Code:



## Task 7:

Code:

