DSP Open Ended Lab-1

To convert analog (Voltage and current) signal into digital signal using ADC (Audio card). Display it on MATLAB/ Simulink environment

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Digital Signal Processing (EE-394)

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Introduction

We have always heard and studied being an electrical engineer that the AC mains coming to our houses is sinusoidal in nature. Though we can observe this on an oscilloscope but not everyone is at liberty to afford one.

So, through this lab we aim to provide a better and cost-effective circuitry to observe those sinusoids. The main idea behind this lab is to observe those sinusoids i.e., 220v 50 Hz signal; to observe AC mains input waveforms on a MATLAB/Simulink environment on a computer by analog to digital signal conversion.

Brief Summary of Lab's working:

Procedure is quite easy, simple and straightforward. Firstly, the 220V rms main AC voltage coming out of a power outlet is stepped down to 12V rms by a transformer. Next a Voltage Divider Circuit (VDR) divides/steps down the voltage further to just 1V. Next this 1V signal goes to an audio jack and through that into a sound card which acts as the Analog to Digital Converter (ADC). After that Digital Signal Processing comes into play. Our signal is transformed into a digital signal. That digital signal is plotted and observed on MATLAB or Simulink environment.

Nothing special needs to be done in order to observe waveforms of AC mains through this circuit. Just plug-in step-down transformer, put audio card into your computer and just run/simulate the code/Simulink simulation and there you have your 50hz input signal which you can observe.

This comprehensive report will walk through all of the technical aspects of this lab, the difficulties and challenges we faced during its accomplishing, all the technical reasons, why certain values are selected why not the other than that and how everything was determined in order to get to our ultimate result.

Components

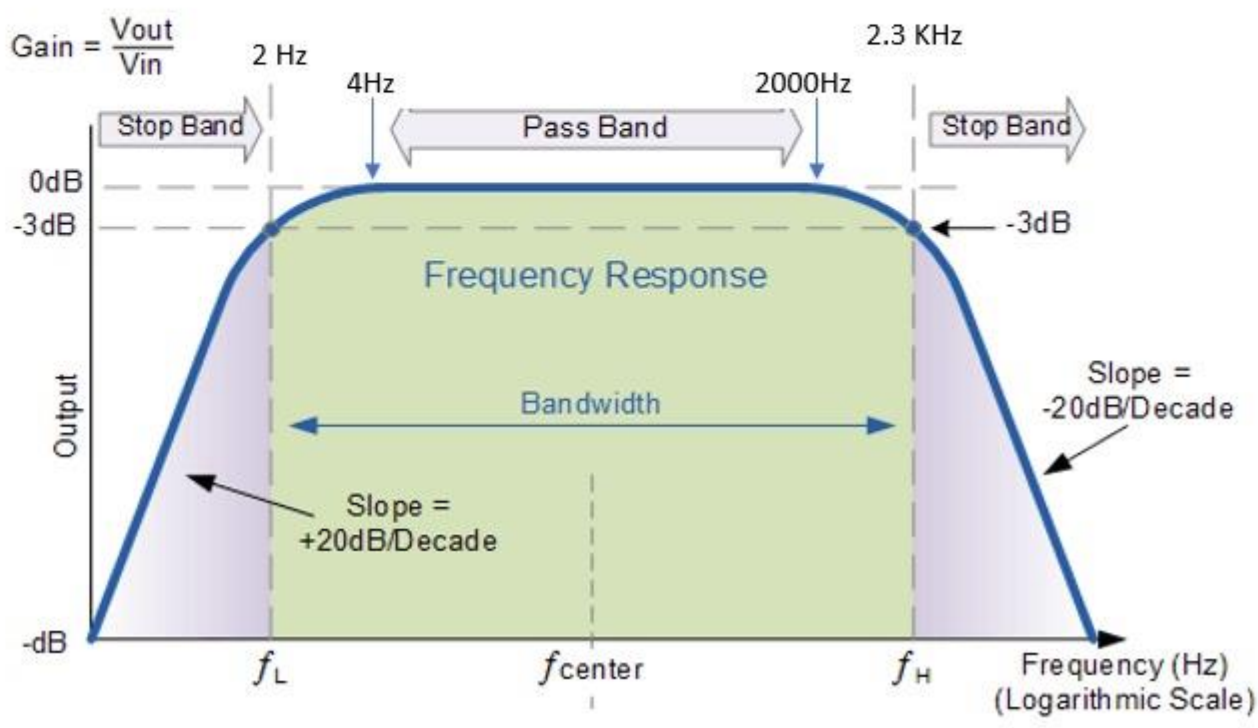
Following components were used:

1. Step-down transformer 240/13.7 V rms
2. 3D sound adapter (ADC)
3. Resistors
4. Audio Jack 3.5mm 3 pole
5. Split Core Current Transformer 10A/1V
6. Connectors
7. Veroboard
8. Connecting wires
9. PC with MATLAB environment

Frequency Response of ADC

Frequency Response of ADC:

3D channel sound adapter (Sound card) is basically a 16-bit sound card which can offer sampling rate of 48000 Hz. We know these 2 specifications by the help of windows by plugging in the sound card into our pc we are able to access this card and read its technical specifications. It has a range of $\pm$ 0.5V. It has a frequency response for listening side which varies from 4Hz to 2000hz which was calculated by hit and trials method. Frequency response was calculated by applying different frequencies via frequency generator and observing response or waveforms on oscilloscope and MATLAB. Firstly, the voltage out of function generator was out of range for our ADC so through our ADC we stepped it down within our acceptable range of ADC. Next, we connected our ADC to this VDR and started increasing frequencies. At 2Hz our gain was 0.707 which is cut off region. Then at 4 Hz the gain was 1 i.e., output was same as input. We kept on increasing the frequencies until 2000 Hz as gain was 1 means bode plot was following a straight line. Beyond this the gain started to decrease and at 2300Hz the gain became again 0.707 so that's another cut off point. The bandwidth is often defined by the frequency that is half-attenuated, or at the midpoint between the most output and no output. This is called the 3 dB bandwidth, also known as the cutoff frequency. Since we got two cut off frequencies so we can term our ADC as a "Bandpass Filter".

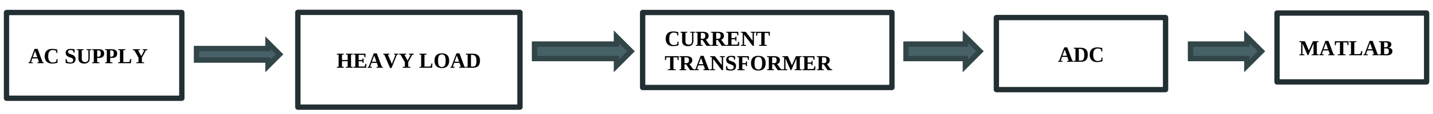


Technical Specifications of CT

Split Core Current Transformer which we used is of rating $10A/1V$. A current transformer is a measuring device which allows us to measure big scale currents on a normal multimeter. You see we can't measure huge currents of say just few Amperes with a normal meter. A current transformer (CT) is a type of transformer that **is used to reduce or multiply an Alternating Current (AC)**. It produces a current in its secondary which is proportional to the current in its primary. Here, our CT measures this current in terms of voltage means if it gets a signal of 10A current it produces a voltage of 1V only. Means it scales down by a factor of 10 which is read through the jack in terms of a voltage signal provided at its output which is inserted into sound card and accessed from PC.

Block Diagram for Voltage

Block Diagram for Current



Procedure

For Voltage Observation:

Here's how everything takes place:

1) First of all a step-down transformer steps down the main AC power coming from K.E which ranges from $220 - 240\ V\ rms$ to $13.4V\ rms$.

2) This $13.4\ V\ rms$ is further divided/stepped down to just $1V\ rms$ using VDR, voltage divider circuit which consists of 2 resistors in series. And as we know from our Basic Electrical Engineering knowledge that in a series circuit voltage divide but current remains the same.

3) In parallel to the 2nd resistor from VDR we connect our 3-pole audio jack.

4) From audio jack we give input to microphone input of sound card.

5) From sound card it goes into computer and onwards to MATLAB/Simulink where it is digitally plotted and the objective of this lab is achieved.

For Current Observation:

1) We connect a load of 100W light bulb. Now from simple $P = VI$ we are able to calculate current which will be flowing from this load $I = 100/240 = 0.41A$.

2) The CT has an audio jack as its output which is fed into microphone part of sound card and which is further plugged into our PC.

3) Our CT has a scaling down factor of 10 so it scales down this value and measures its corresponding voltage value of 0.41. So, in terms of voltage, we have a current of $0.041\ A$.

4) Based on these calculations we are able to see the digital plot of current on MATLAB/Simulink and the objective of this lab is achieved.

Calculations

Supply Voltage = $V_{rms} = 240\ V$
Input Voltage into ADC = 0.2 $V\ rms$
ADC drops = 0.02 $V\ rms$

1) VDR

$$R_1 = 18.7k\Omega$$
$$R_2 = 280\Omega\ (through\ potentiometer)$$
$$V = 13.4\ V$$

$$V_{R2} = V * \left(\frac{R_2}{R_1 + R_2}\right)$$

$$V_{R2} = 13.4 * \left(\frac{280}{18.7k + 280}\right)$$

$$\boxed{\boldsymbol{V_{R2} = 0.2\ V\ rms}}$$

<u>For Current of CT</u>: Bulb of 100 watts was used

$$P = I\ V$$
$$100 = I * (240)$$
$$\boxed{\boldsymbol{I = 0.4166\ A}}$$

2) Resolution

$$\boxed{\boldsymbol{Resolution = 16\ bits}}$$

3) Sampling Frequency

$$\boxed{\boldsymbol{Sampling\ Frequency = 48000\ Hz}}$$

4) Gain Factor

<u>For Voltage:</u>

As the ADC drops 0.02 V, so the input voltage in rms appeared in MATLAB was 0.18 V

$$Gain\ factor\ =\ V_{out}\ /\ V_{in}$$

$$Gain\ Factor\ =\ 240/\ 0.18$$

$$\boxed{\boldsymbol{Gain\ Factor\ =\ 1333.333}}$$

<u>For Current:</u>

The current drawn by our load as seen above was 0.4 A. Since CT transformer read the corresponding current value 10 times scaled down and in terms of voltage. So, the input to our ADC is as appeared on MATLAB was:

$$\boldsymbol{I\ (in\ rms) = 0.0416\ A}$$

As our ADC drops some voltage so the current appeared on MATLAB 0.04 A in terms of voltage as 0.04 V

Hence Gain Factor for current will be:

$$\boldsymbol{Gain\ Factor\ =\ 10} \qquad \text{(i.e., CT reads 10A/1V)}$$

Waveforms

Voltage:

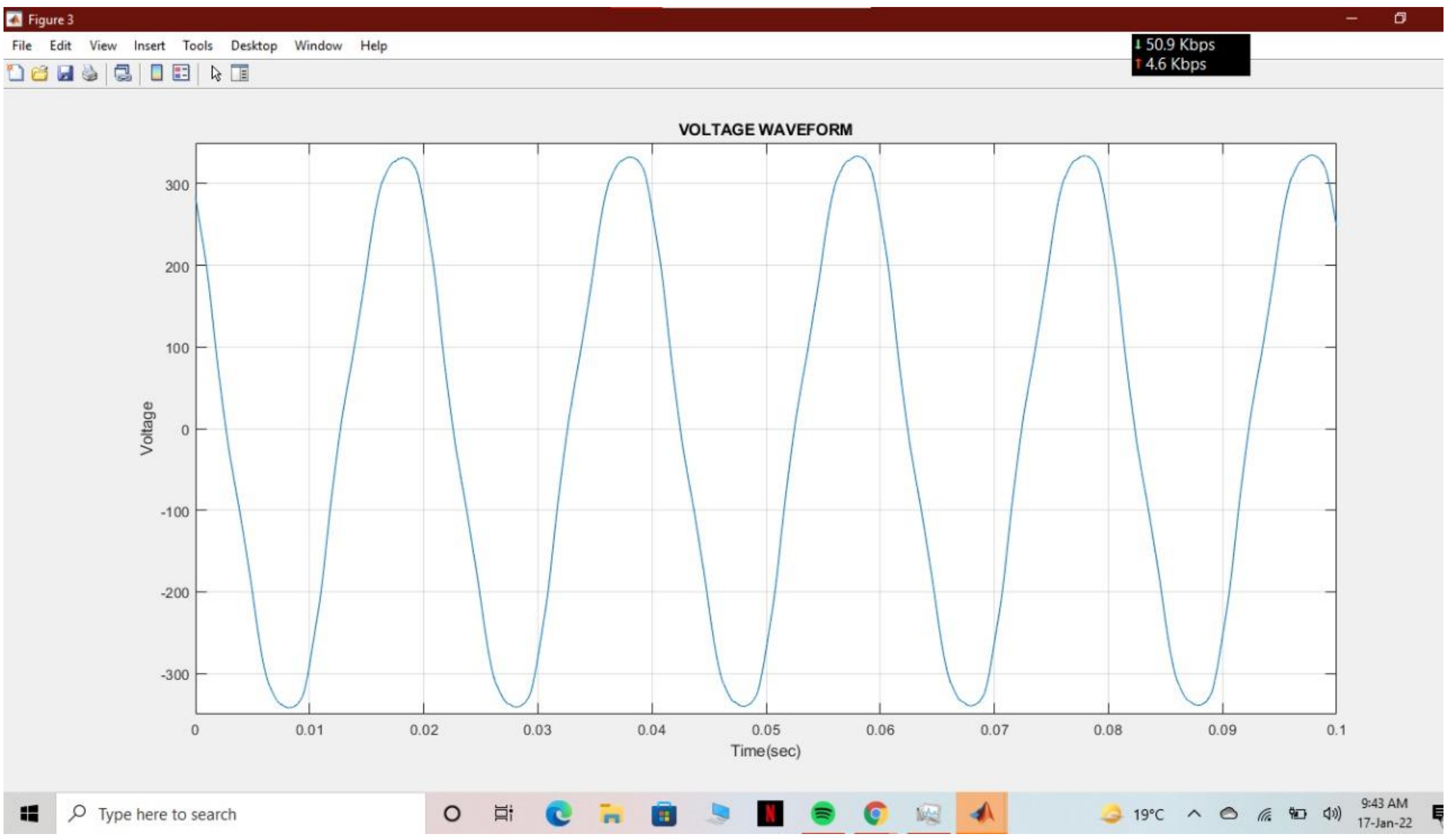


Fig. 1. Voltage waveform observed

Waveforms

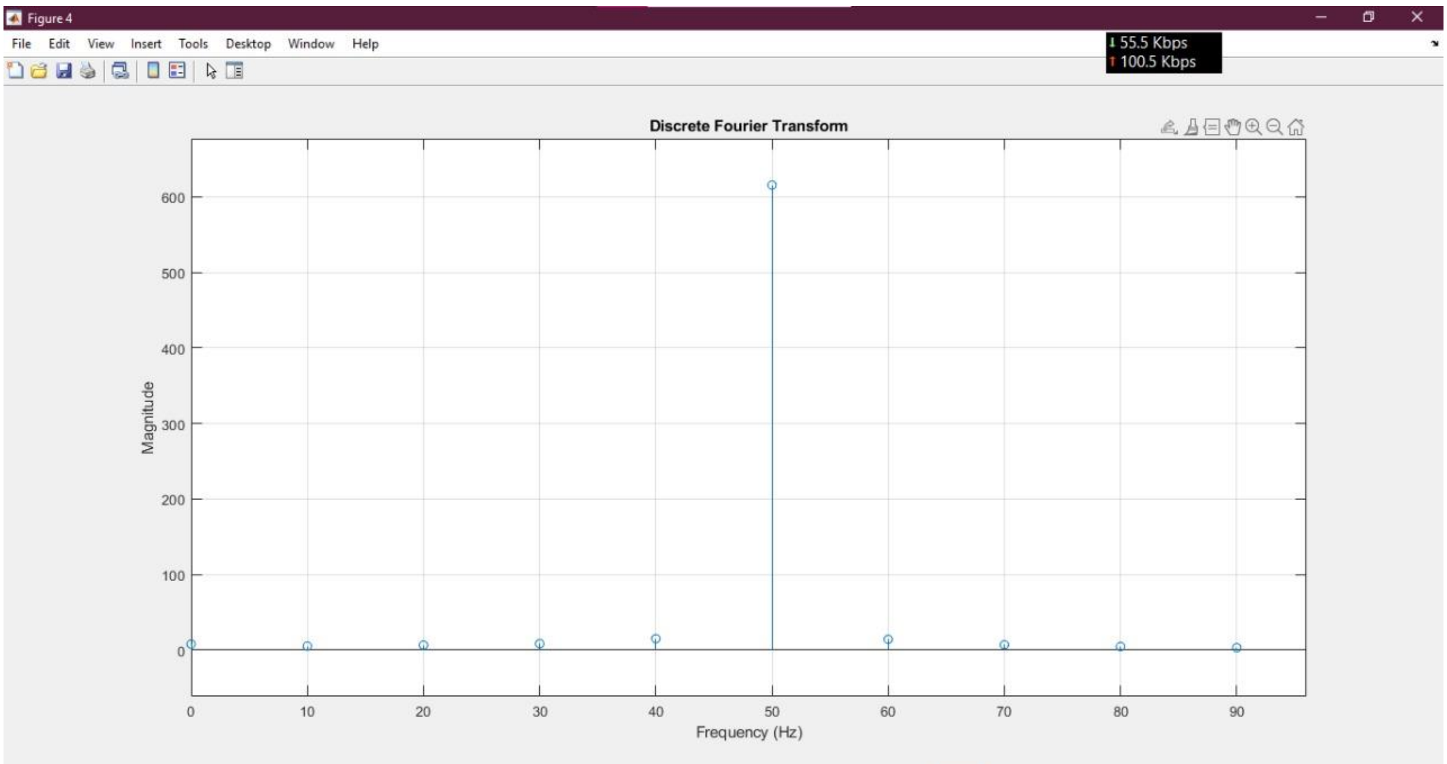


Fig. 2. Frequency Components present in voltage waveform as seen in its DFT

Magnitude here of DFT is very large due to the reason that we haven’t normalized it that’s why. Same goes for in the current waveform too.

Waveforms

Current:

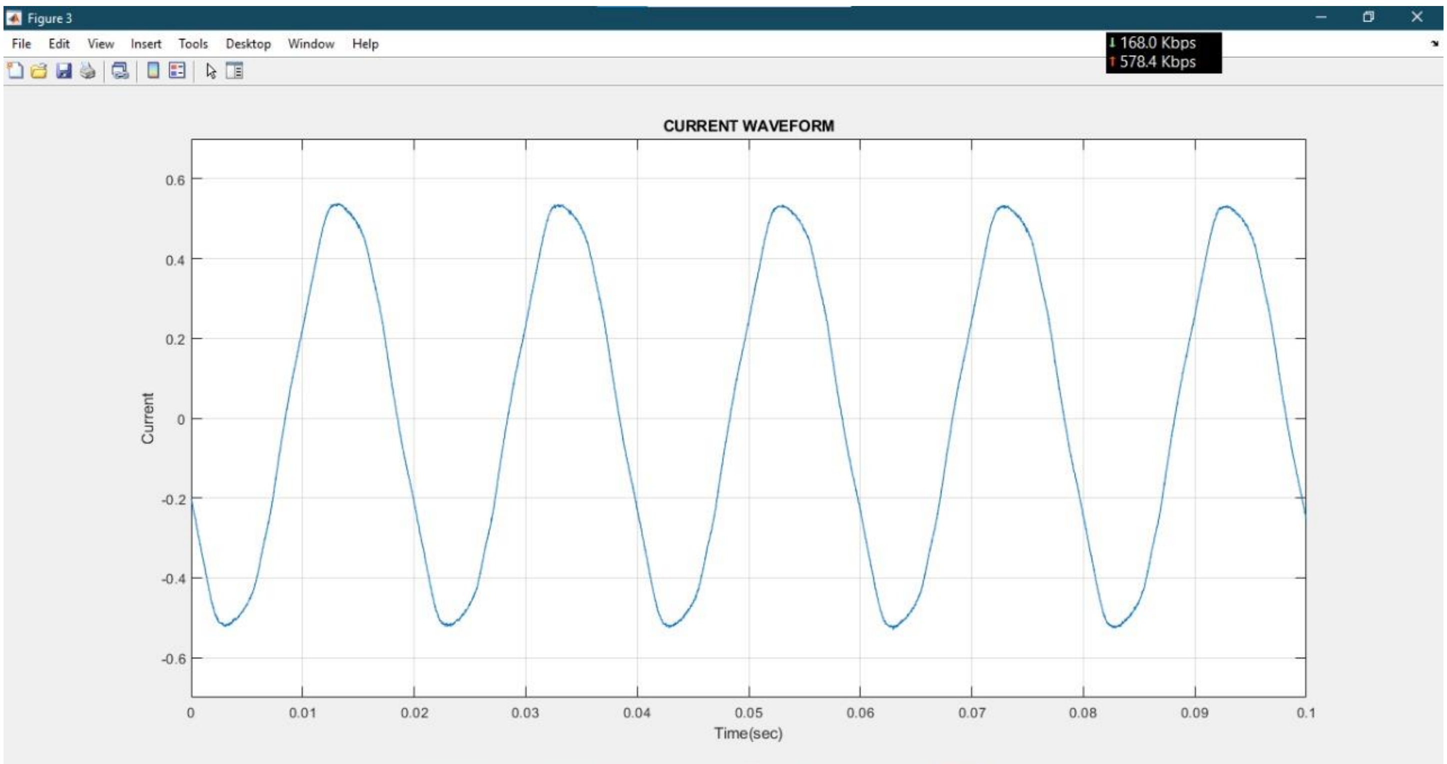


Fig. 3. Current Waveform Observed

Waveforms

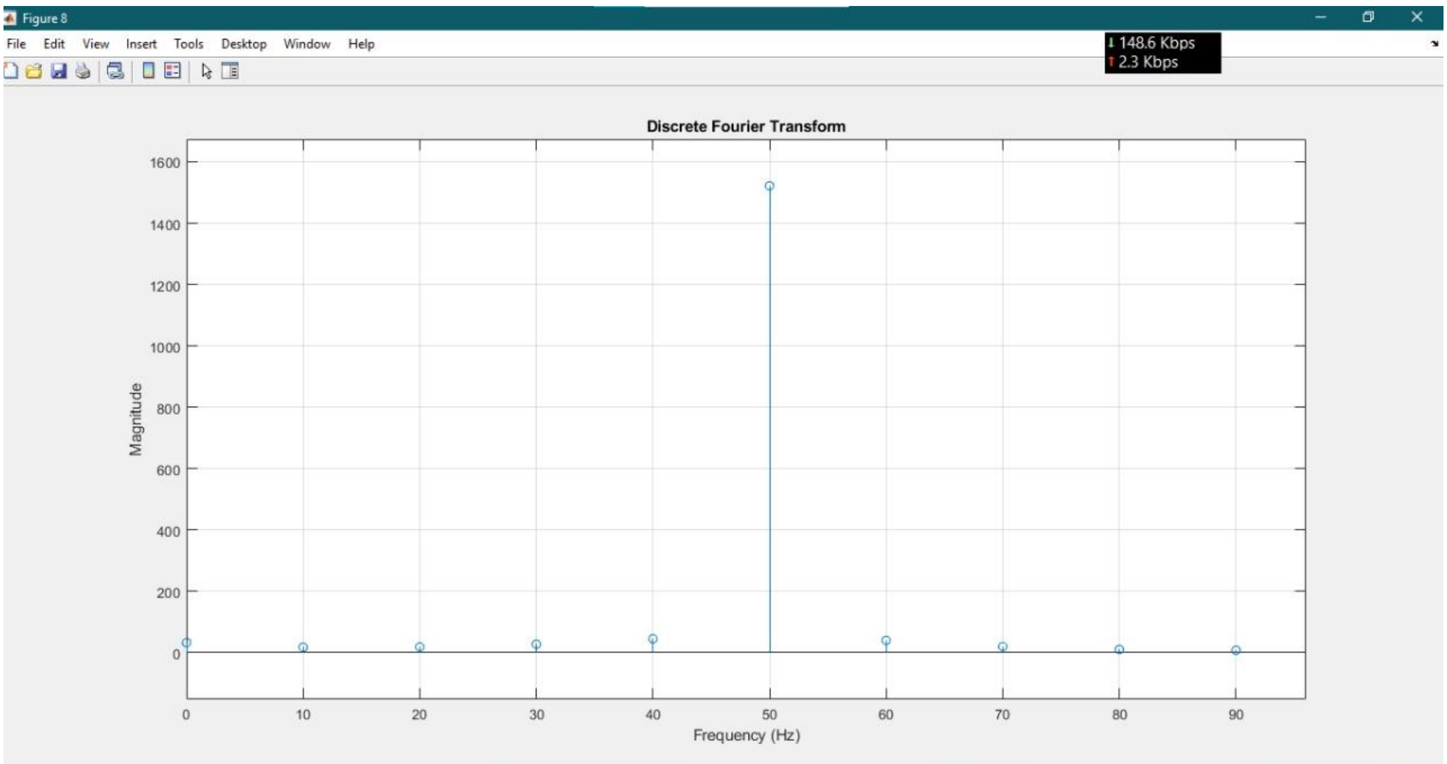
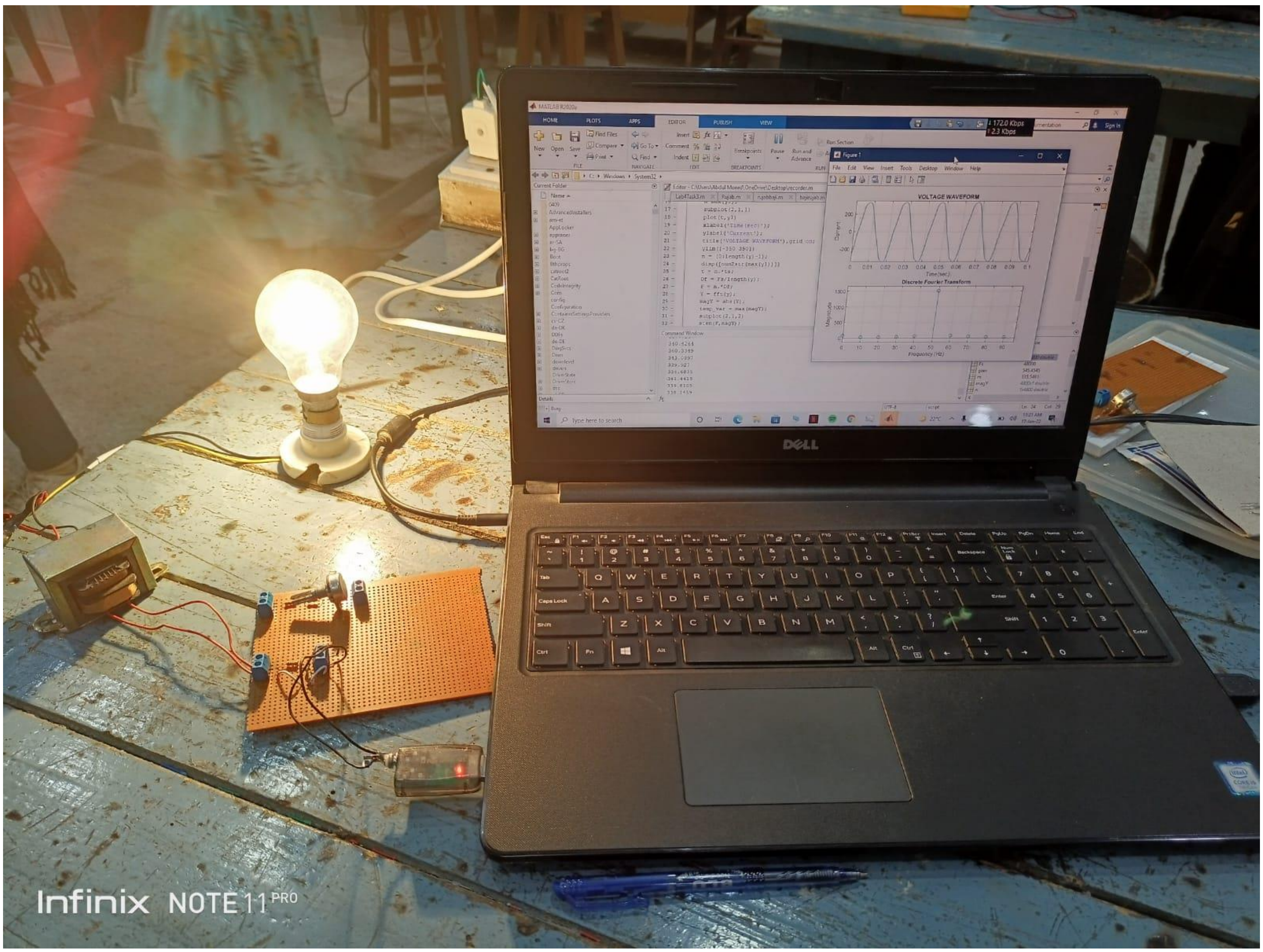


Fig. 4. Current's frequency components present in its waveform as seen in its DFT

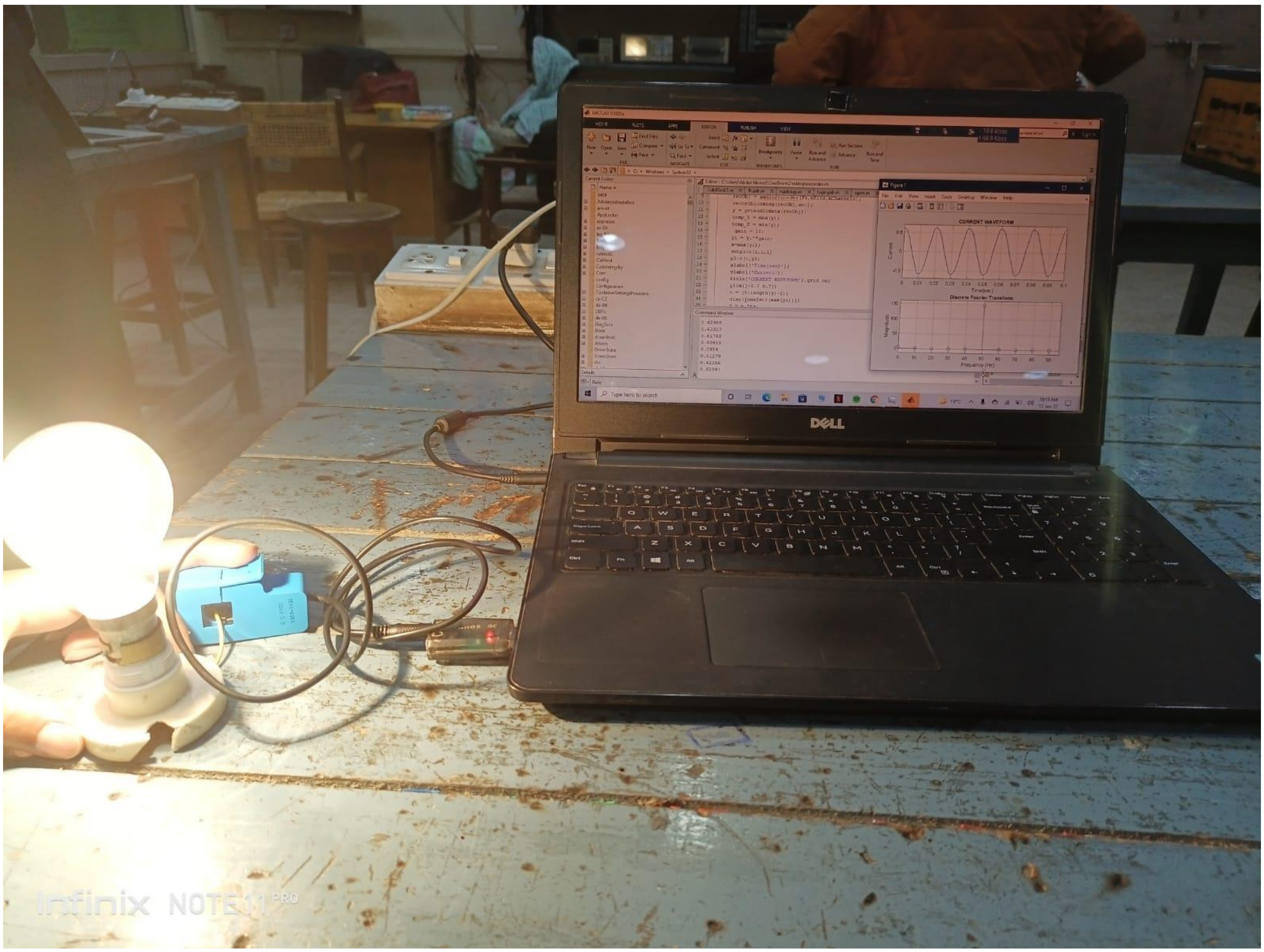
Experimental Setup

For Voltage:



Experimental Setup

For Current:



MATLAB Code for Voltage

```
%For Voltage
clc,clear,close all
Fs = 48000 ;
nBits = 16 ;
nChannels = 1 ;
ts = 1/Fs;
sec = 0.1;
t = [ts:ts:sec];
while 1
    recObj = audiorecorder(Fs,nBits,nChannels);
    recordblocking(recObj,sec);
    y = getaudiodata(recObj);
    temp_1 = max(y);
    temp_2 = min(y);
     gain = (240/0.18);
    y1 = y.'*gain;
    m=max(y1);
    subplot(2,1,1)
    plot(t,y1)
    xlabel('Time(sec)');
    ylabel('Voltage');
    title('VOLTAGE WAVEFORM'),grid on;
    ylim([-350 350])
    n = (0:length(y)-1);
    disp([num2str(max(y1))])
    t = n.*ts;
    Df = Fs/length(y);
    F = n.*Df;
    Y = fft(y);
    magY = abs(Y);
    temp_var = max(magY);
    subplot(2,1,2)
    stem(F,magY);
    xlabel('Frequency (Hz)');
    ylabel('Magnitude');
    ylim([((-10/100)*temp_var) (temp_var+(temp_var*(10/100)))]),
    xlim([0 Fs/500]),grid on;
    title('Discrete Fourier Transform');

end
```

MATLAB Code for Current

```
%For Current
clc,clear,close all
Fs = 48000 ;
nBits = 16 ;
nChannels = 1 ;
ts = 1/Fs;
sec = 0.1;
t = [ts:ts:sec];
while 1
    recObj = audiorecorder(Fs,nBits,nChannels);
    recordblocking(recObj,sec);
    y = getaudiodata(recObj);
    temp_1 = max(y);
    temp_2 = min(y);
     gain = 10;
    y1 = y.'*gain;
    m=max(y1);
    subplot(2,1,1)
    plot(t,y1)
    xlabel('Time(sec)');
    ylabel('Current');
    title('CURRENT WAVEFORM'),grid on;
    ylim([-0.7 0.7])
    n = (0:length(y)-1);
    disp([num2str(max(y1))])
    t = n.*ts;
    Df = Fs/length(y);
    F = n.*Df;
    Y = fft(y);
    magY = abs(Y);
    temp_var = max(magY);
    subplot(2,1,2)
    stem(F,magY);
    xlabel('Frequency (Hz)');
    ylabel('Magnitude');
    ylim([((-10/100)*temp_var) (temp_var+(temp_var*(10/100)))]),
    xlim([0 Fs/500]),grid on;
    title('Discrete Fourier Transform');

end
```