DSP Open Ended Lab-2

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

Prepared by:

Ali Aman EE-19136
Hammad Ali Khan EE-19139
Moeed Alam EE-19170
Syed Muhammad bin Tahir EE-16309

Date:

01-02-2022

Course title:

Digital Signal Processing (EE-394)

Teacher's name:
Muhammad Omar

Table of Contents

Contents

Introduction

The very basic law in electrical engineering is Ohm's law that simply states that voltage and current follow linear relationship for linear loads. By linear loads we mean that their voltage and current characteristics are linear; a change in voltage is linear to their change in current. Also, the voltage and current waveform of such loads are sinusoidal in nature as was observed in open ended lab 1.

Now there's something called "Non-linear loads". By nonlinear loads we mean such loads don't follow ohm's law i.e., their current doesn't linearly vary with the voltage. Now such loads draw a current which are not necessarily sinusoidal hence causing Current Harmonics. Their pattern can be quite complex depending upon the load itself and its interaction with other components of the system to which its connected.

Non-linear load currents create distortion in the pure sinusoidal voltage/current waveform supplied by the utility, and this may result in resonance. The even harmonics do not normally exist, and this can be proved using Fourier series, in power system due to symmetry between the positive- and negative- halves of an AC cycle. Further, if the waveforms of the three phases are symmetrical, the harmonic multiples of three are suppressed by delta (Δ) connection of transformers or motors.

Now Current Harmonics. Harmonics in electrical power system are in simple words frequencies. Harmonics of a voltage or current is a waveform whose frequency are an integral multiple of fundamental frequency. By integral multiple we mean for example third harmonic is the third multiple of fundamental frequency. It's well established at this point that this fundamental frequency in our case is 50Hz, so third harmonic of 50Hz will obviously be 150 Hz. As discussed above these waveforms aren't necessarily sinusoids and can be way more complex, that depends. Regardless of how complex the current waveform becomes, the Fourier series transform makes it possible to deconstruct the complex waveform into a series of simple sinusoids, which start at the power system fundamental frequency and occur at integer multiples of the fundamental frequency.

Another term which goes side by side along with DFT is Leakage. If the input signal contains any frequency which is ***NOT*** integer multiple of analysis frequencies (which is the most probably happening case), it will show up in all DFT output bins or *leaks out to all DFT output bins*. This effect is simply known as **DFT Leakage/ Spectral Leakage**. We typically say that input signal energy shows up in all of the DFT's output bins. Think of bins as buckets to which you pour energy from infinite amount of possible frequency components. If the signal's frequency is not matched with exact frequency bin, then it smears leaks over rest of the bins.

Introduction

Another way to think about DFT leakage is, when the input signal has frequencies which do not complete integer number of cycles in the sample interval ***N* (Length of DFT)**, the input to DFT seems like **abruptly starting** and **abruptly ending** giving rise to side lobes in the DFT output. A selection of finite input samples means that we have multiplied the input sequence by a rectangular window. The DFT output will follow the Sinc shape which is the Fourier Transform of the rectangular function.

For real life signals, DFT leakage can never be entirely eliminated because no matter what point DFT you take and no matter what your sampling rate is, there is a very high probability that the input would contain a frequency which is not an integer multiple of analysis frequencies and DFT leakage *would* happen. We can minimize DFT leakage by multiplying the input with a smooth window which will make the input go from zero to maximum and back to zero in a very smooth and slow fashion. This reduces the abruptness of the input which minimizes the higher sides lobes in the DFT output.

So, through this lab we aim to see such phenomenon that is observing frequency components at integral multiples of base frequency and applying Discrete Fourier Transform (DFT) on distorted current waveform drawn by our harmonic load and observe the phenomenon of leakage and if leakage happens then minimizing it with the help of windowing tool.

The arrangement and procedure are approximately same as were in open ended lab 1. The only change here is connected load which will be a Compact Fluorescent Lamp (CFL) or LED bulb which is in fact a harmonic load and will produce a distorted current waveform. Our objective is to observe it; the current waveform and perform spectral analysis to it which involves applying Fourier transform to the digital signal, observing the frequency components present in it, see the phenomenon of leakage if happening and if it is then applying windowing to at least minimize it as far as we can.

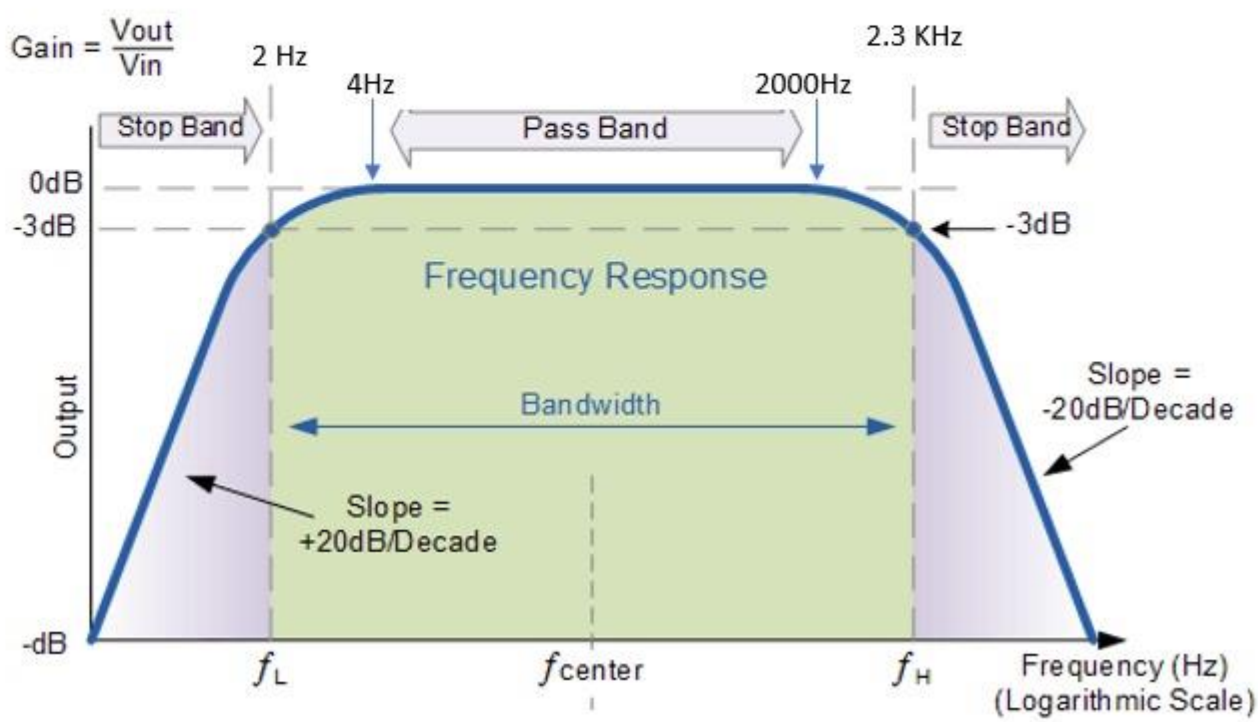
Components

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
10. Harmonics producing load (CFL/LED bulb)

Frequency Response of ADC

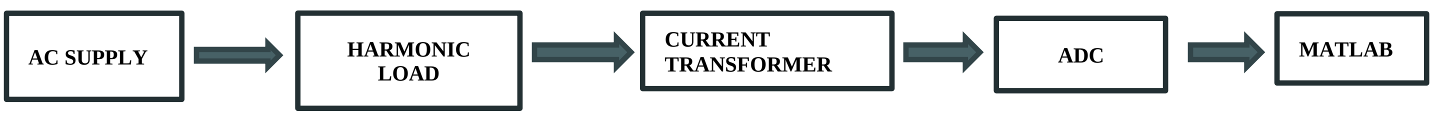
3D channel sound adapter (Sound card) is basically a 16-bit sound card which can offer sampling rate of 48000 Hz. We know these 2 specifications by the help of windows by plugging in the sound card into our pc we are able to access this card and read its technical specifications. It has a range of $\pm$ 0.5. It has a frequency response for listening side which varies from 4 Hz to 2000 Hz 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 2 Hz 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 2300 Hz 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 Specification 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 above 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 soundcard and accessed from PC.

Block Diagram for Current



Block Diagram for Voltage

Procedure

For Voltage Observation:

Here's how everything takes place:

1) Harmonic load was connected in parallel to the K.E supply mains. In parallel to this a step-down transformer was connected to step down the voltage.

2) Voltage is then 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 2^{nd} 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 $25W$ CFL bulb which is a harmonic load. Now from simple $P = VI$ we are able to calculate current which will be flowing from this load is $I = 25/240 = 0.104\ A$.

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 $0.0104A$. So, in terms of voltage, we have a current of $0.0104\ A$. since it's a harmonic load, it induces harmonics so its magnitude will not be this value instead it will be of greater magnitude which will be evident later on in the observations.

4) Based on these calculations we are able to see the digital plot of current wave which is indeed distorted due harmonic present in it on MATLAB/Simulink and the objective of this lab is achieved.

Calculations

Supply Voltage = $V_{rms} = 240\ V$
Transformer steps down to $V_{rms} = 13.4\ V$

1) VDR

$$R_1 = 820\Omega$$
$$R_2 = 22000\Omega$$
$$V = 13.4\ V$$

$$V_{R1} = V * \left(\frac{R_2}{R_1 + R_2}\right)$$

$$V_{R1} = 13.4 * \left(\frac{820}{820 + 22000}\right)$$

$$\boxed{\boldsymbol{V_{R1} = 0.48\ V\ rms}}$$

<u>For Current of CT</u>: Bulb of 25 watts was used

$$P = I\,V$$
$$25 = I * (240)$$
$$\boxed{\boldsymbol{I = 0.104\ A}}$$

2) Resolution

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

3) Sampling Frequency

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

4) Gain Factor

For Voltage:

As the ADC drops 0.02, so the input voltage in rms appeared in MATLAB was 0.46 $V\ rms$

Input Voltage into ADC = 0.48 $V\ rms$

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

$$Gain\ Factor\ =\ 240/\ 0.46$$

$$\boxed{\boldsymbol{Gain\ Factor\ =\ 521.739}}$$

For Current:

The current drawn by our load as seen above was 0.104 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:

$$\boxed{\boldsymbol{I\ (in\ rms) = 0.0104\ A}}$$

Since this is harmonic load so the actual current appeared on MATLAB is in form of voltage is $\boldsymbol{I\ (in\ rms) = 0.021\ A}$**.**

Hence Gain Factor for current will be:

$$\boldsymbol{Gain\ Factor\ =\ 10}$$

So actual current value will be $\boldsymbol{0.21\ A}$

Waveforms

Voltage:

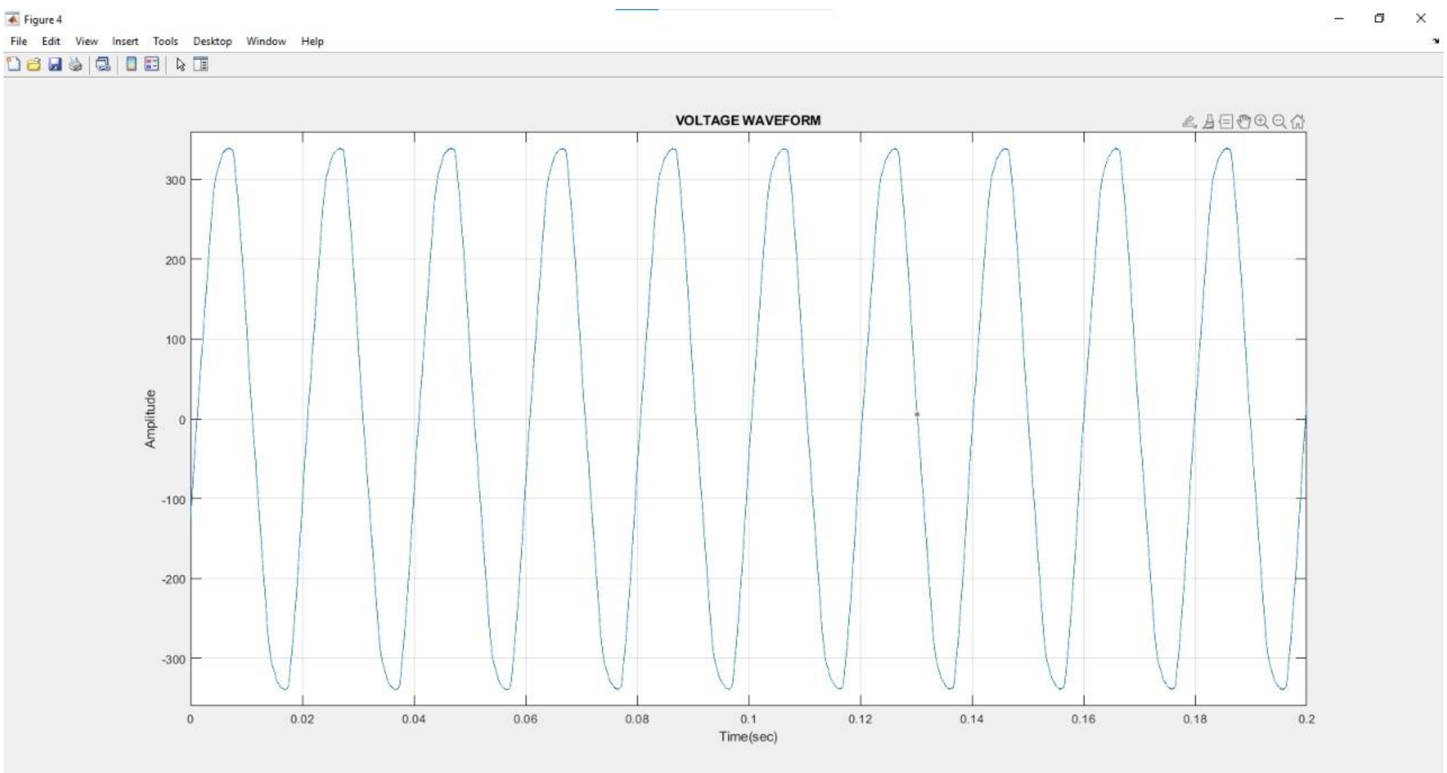


Fig. 1. Observed Voltage Waveform

Waveforms

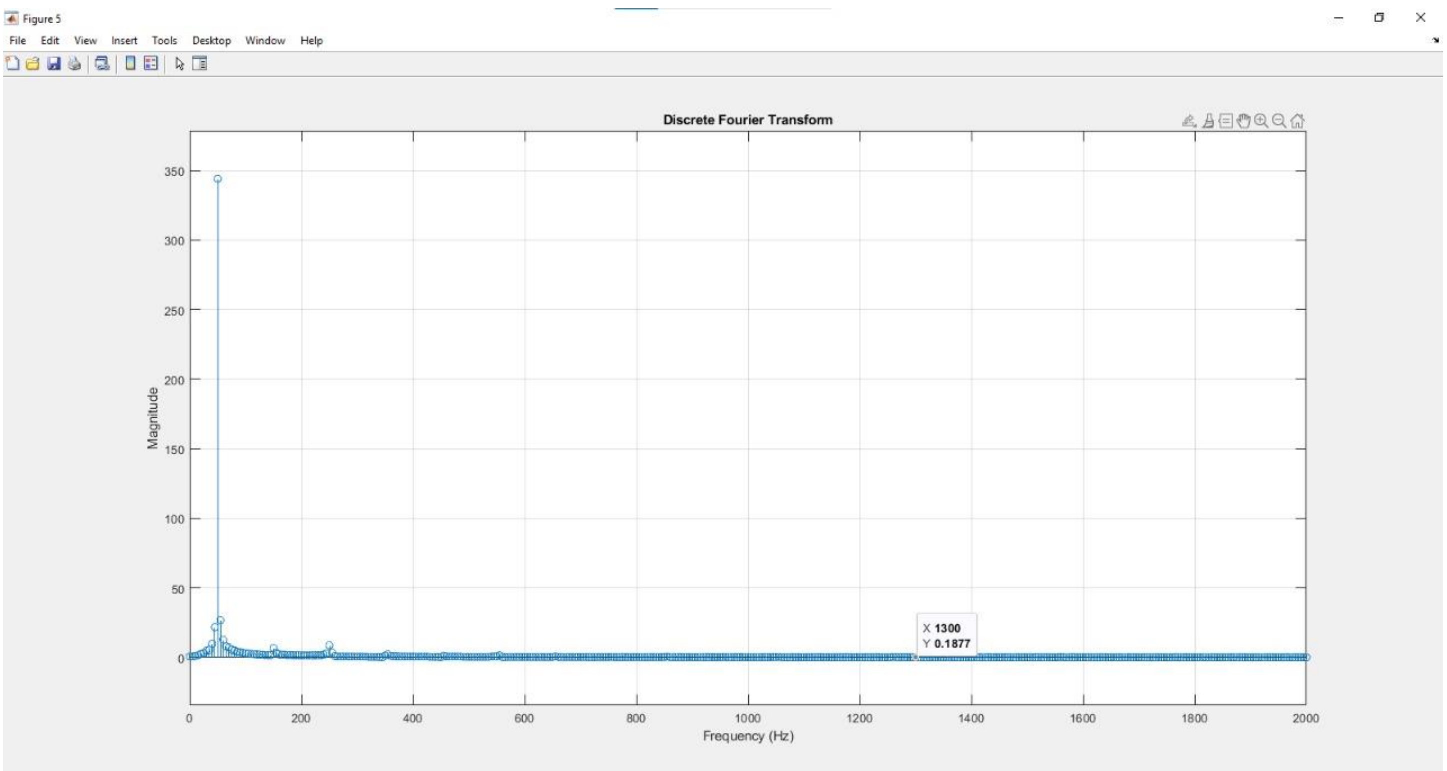


Fig. 2. DFT of observed Voltage Waveform

It’s weird seeing leakage in voltage waveform’s DFT. We as a group are unable to answer this. One theory which we have discussed and mutually agreed is that since the observed leakage are in close vicinity of 50 Hz so we can say that they are sort of insignificant being close to 50Hz. However, if these were observed around 40Hz 60Hz etc. we would then have considered it as a major fault or blunder in our observations.

Waveforms

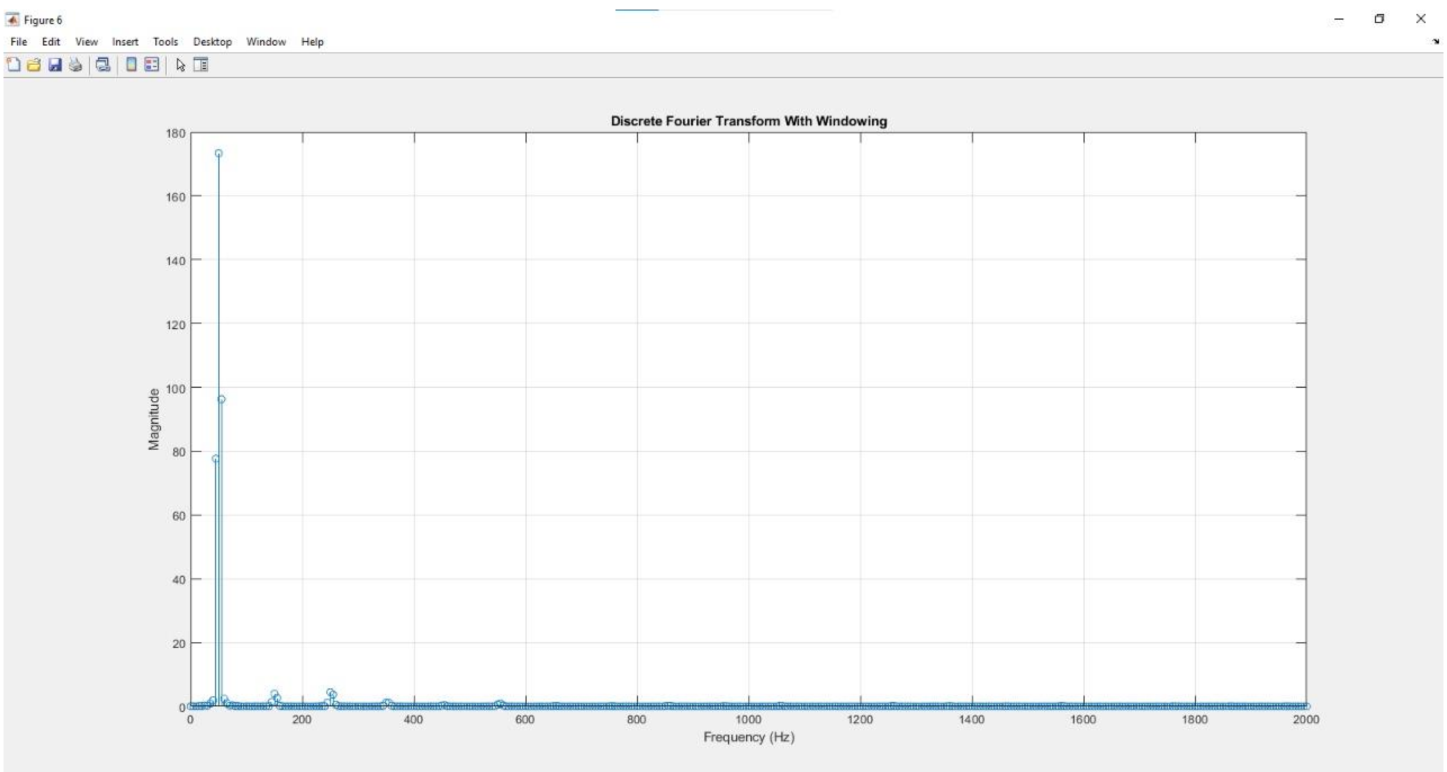


Fig. 3. DFT of observed Voltage Waveform

Notice there is leakage present still in the DFT of observed voltage waveform after window application but it's minimized to some extent.

Waveforms

Current:

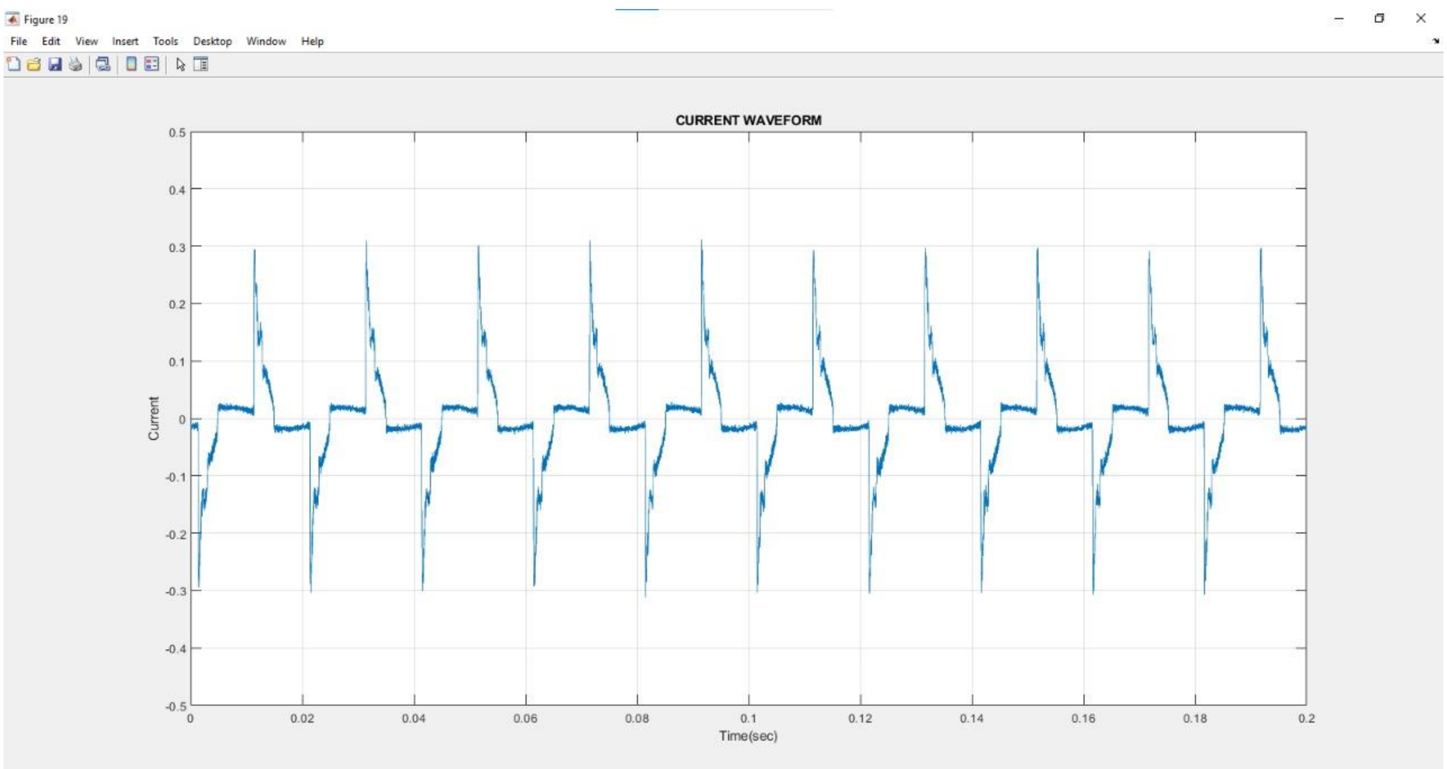


Fig. 4. Observed Distorted Current Waveform

Waveforms

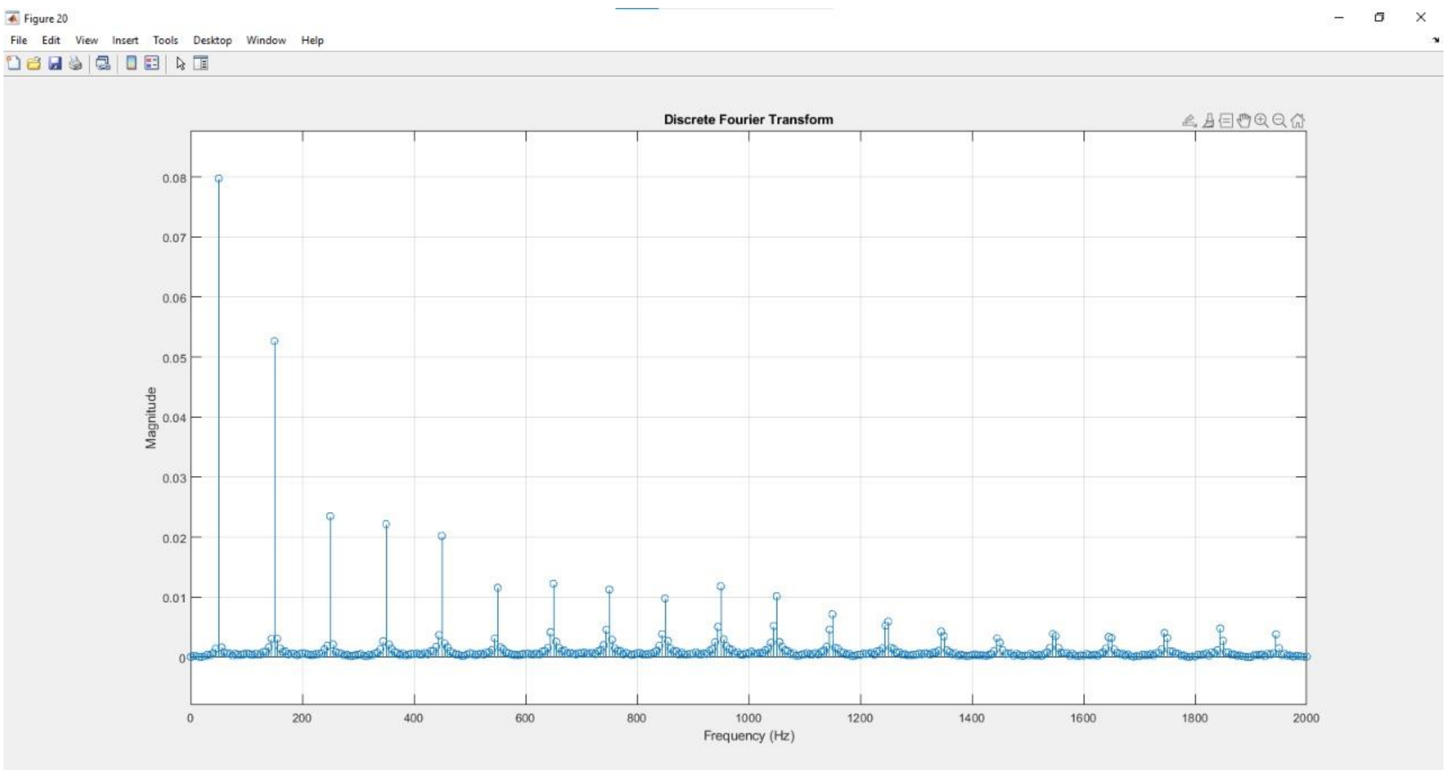


Fig. 5. DFT of observed current waveform

Observe that at odd harmonics are present and that main contributor is the fundamental frequency i.e., 50 Hz and after that every harmonic is small in magnitude than fundamental frequency but their frequency is $n\ times$ the fundamental.

Waveforms

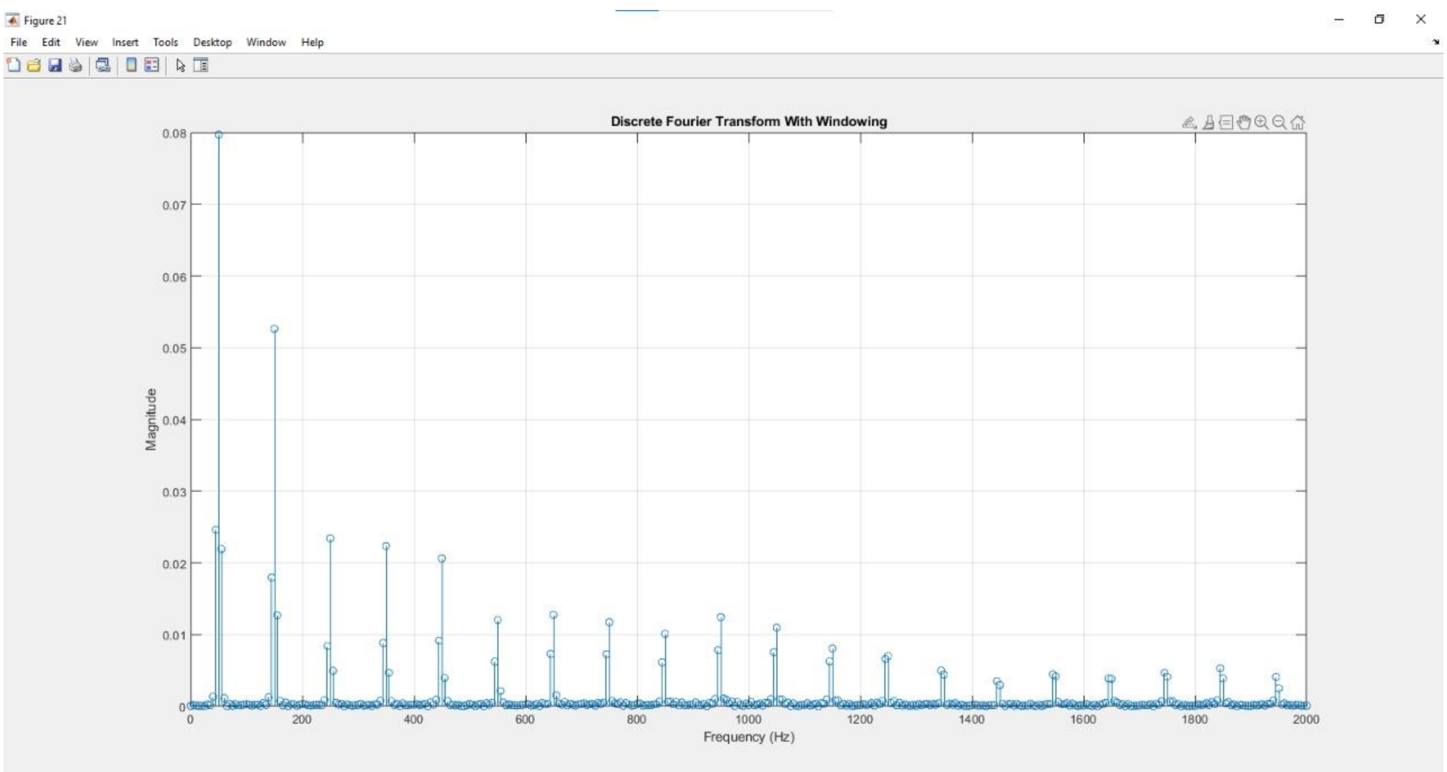
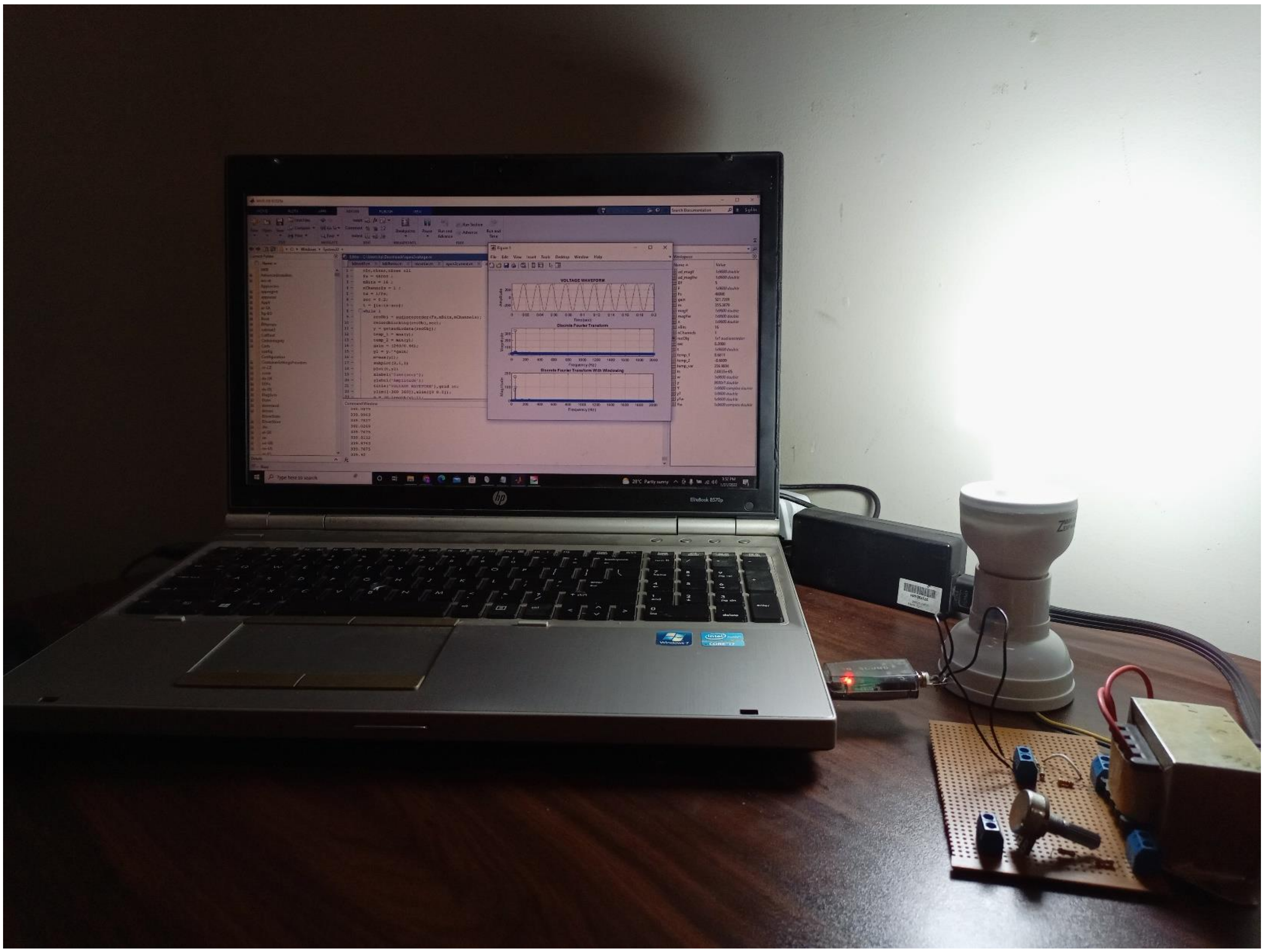


Fig. 6. DFT of current waveform after window application

Observe that after window application leakage is minimized to a great extent only that remains are of odd harmonics which is understandable.

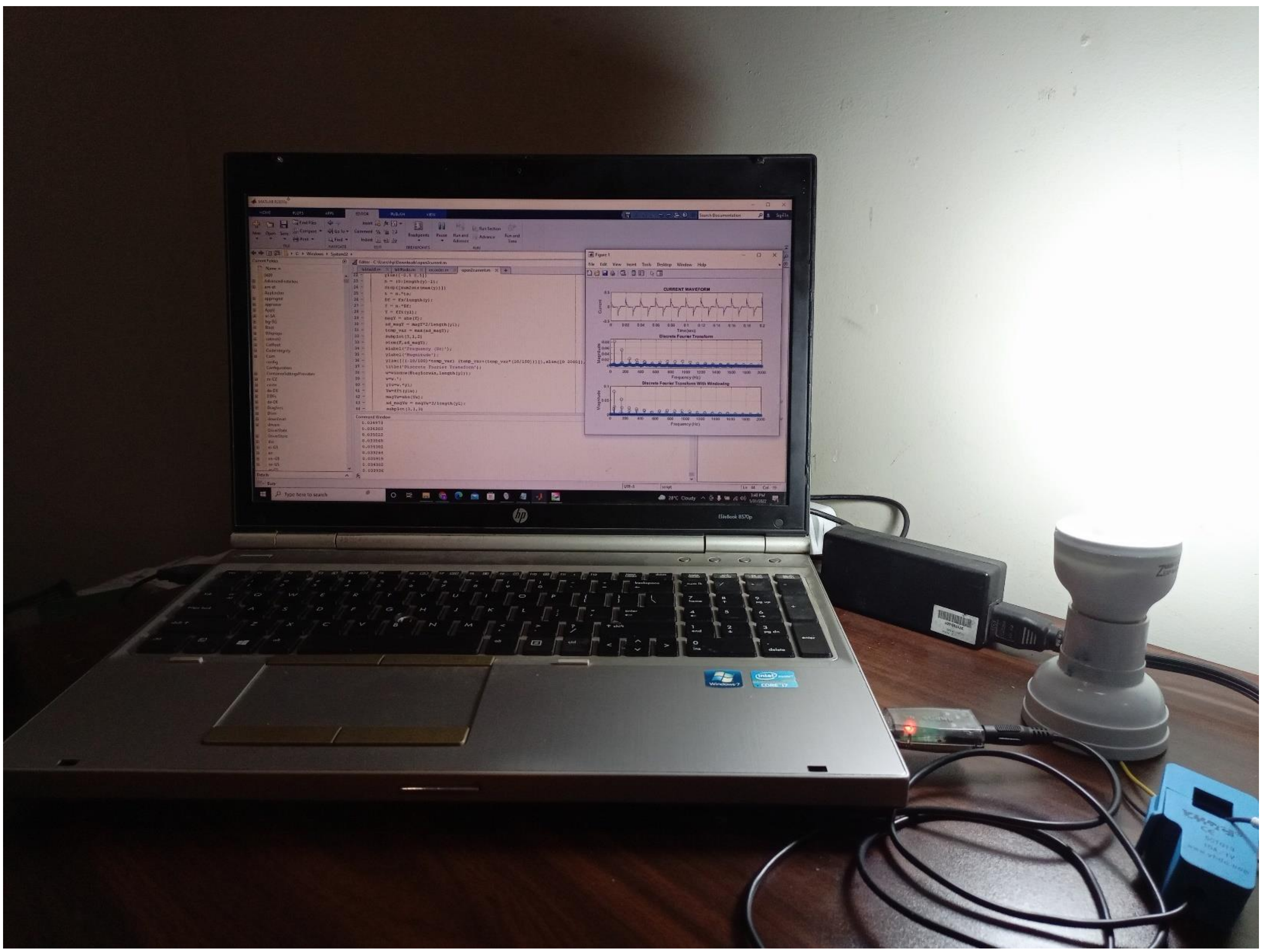
Experimental Setup

Voltage:



Experimental Setup

Current:



MATLAB Code for Voltage

```
%For Voltage

clc,clear,close all
Fs = 48000 ;
nBits = 16 ;
nChannels = 1 ;
ts = 1/Fs;
sec = 0.2;
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 = (240/0.46);
    y1 = y.'*gain;
    m=max(y1);

    subplot(3,1,1)
    plot(t,y1)
    xlabel('Time(sec)');
    ylabel('Amplitude');
    title('VOLTAGE WAVEFORM'),grid on;
    ylim([-360 360]),xlim([0 0.2]);

    n = (0:length(y)-1);
    disp([num2str(m)])
    t = n.*ts;
    Df = Fs/length(y);
    F = n.*Df;
    Y = fft(y1);
    magY = abs(Y);
    ad_magY = magY*2/length(y1);
    temp_var = max(ad_magY);

    subplot(3,1,2)
    stem(F,ad_magY);
    xlabel('Frequency (Hz)');
    ylabel('Magnitude');
    ylim([((-10/100)*temp_var) (temp_var+(temp_var*(10/100)))]),xlim([0 2000]),grid
on;
    title('Discrete Fourier Transform');
    w=window(@hann,length(y1));
```

MATLAB Code for Voltage

```
    w=w.';
     y1w=w.*y1;
     Yw=fft(y1w);
     magYw=abs(Yw);
     ad_magYw = magYw*2/length(y1);

     subplot(3,1,3)
     stem(F,ad_magYw);
     xlabel('Frequency (Hz)');
     ylabel('Magnitude');
     xlim([0 2000]),grid on;
     title('Discrete Fourier Transform With Windowing');
end
```

MATLAB Code for Current

```
%For Current
clc,clear,close all
Fs = 48000 ;
nBits = 16 ;
nChannels = 1 ;
ts = 1/Fs;
sec = 0.2;
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(3,1,1)
    plot(t,y1),xlim([0 0.2]);
    xlabel('Time(sec)');
    ylabel('Current');
    title('CURRENT WAVEFORM'),grid on;
    ylim([-0.5 0.5])

    n = (0:length(y)-1);
    disp([num2str(max(y))])
    t = n.*ts;
    Df = Fs/length(y);
    F = n.*Df;
    Y = fft(y1);
    magY = abs(Y);
    ad_magY = magY*2/length(y1);
    temp_var = max(ad_magY);

    subplot(3,1,2)
    stem(F,ad_magY);
    xlabel('Frequency (Hz)');
    ylabel('Magnitude');
    ylim([((-10/100)*temp_var) (temp_var+(temp_var*(10/100)))]),xlim([0 2000]),grid
on;
    title('Discrete Fourier Transform');
    w=window(@taylorwin,length(y1));
    w=w.';
```

```
    y1w=w.*y1;
    Yw=fft(y1w);
    magYw=abs(Yw);
    ad_magYw = magYw*2/length(y1);

    subplot(3,1,3)
    stem(F,ad_magYw);
    xlabel('Frequency (Hz)');
    ylabel('Magnitude');
    xlim([0 2000]),grid on;
    title('Discrete Fourier Transform With Windowing');
end
```