__Electronic Design Project: Stage B.__

Measuring distortion in main supply.

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Abstract:
Inverters are used to store electrical energy and provide a sinusoidal wave output as close to supply voltage as possible in the case of a power failure. However, the basic circuit of an inverter (Figure 1) produces a distorted wave (which is between a sinusoid and square wave) of supply frequency rather than a perfect sinusoidal wave as desired. In this project, we have designed a circuit to find out the distortion in the inverter output quantitatively and this information can be used to determine the quality of inverters and then those devices which can tolerate this much distortion can be used effectively.

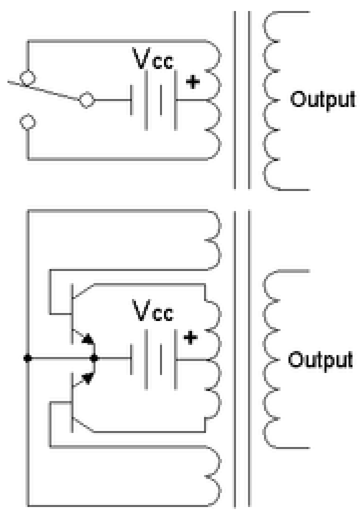

Figure 1

Introduction:
The main part of the project deals with programming a microcontroller (ATmega 32) to calculate the first few harmonic components (10) in the scaled down version of the supply voltage input wave. These harmonics are used to measure the total harmonic distortions and the results are displayed on an attached LCD screen.

Block diagram:

The key components of our project are:

1. A 230:9-0-9 transformer that scales down the supply voltage to a voltage range of 0-9 V , the half of which is compatible with the microcontroller ATmega 32.
2. Microcontroller ATmega32 which is programmed to calculate and display the required Fourier coefficients.
3. A LCD screen interfaced to the microcontroller that displays the percentage energy in the first few prominent harmonics of the sine wave.

The block diagram is as shown in Figure 2.

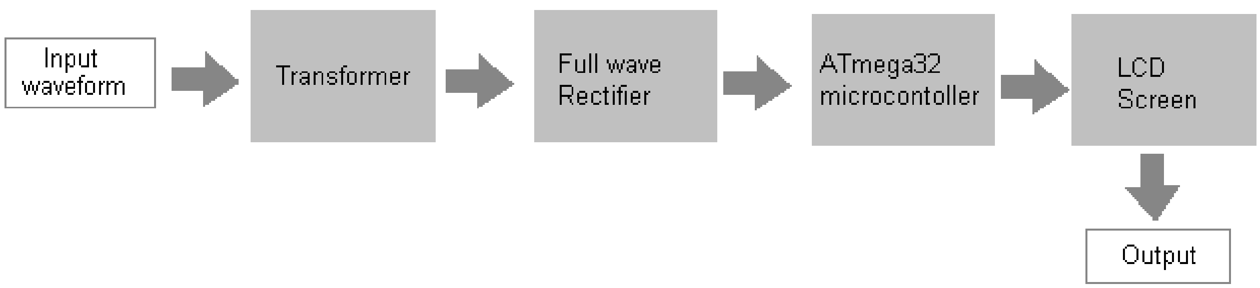

Figure 2.

Components used:

1. ATmega 32 microcontroller.
2. LCD screen
3. 230:9-0-9 transformer
4. 50 K potentiometers
5. 7805 to get a constant 5V dc supply
6. Diodes and Resistors

Analysis:

The assumptions taken are that the input waveform is of 50 Hz frequency. Since, the given wave is generally the output of a basic inverter circuit (Fig. 1), we can further assume that the input is mathematically between a perfect sine wave and a perfect square wave of 50 Hz frequency. This means the cause of distortion is the presence of extra components apart from the 50 Hz component. The Fourier series of a square wave is given as:

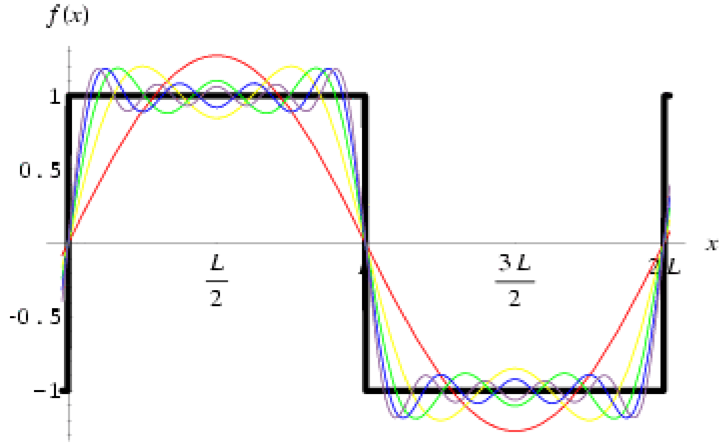

Figure 3.

$$f(x) = \frac{4}{\pi} \sum_{n=1,3,5,\ldots}^{\infty} \frac{1}{n} \sin\left(\frac{n\pi x}{L}\right). \quad \ldots\ldots\ldots\ldots\ldots (1)$$

It is observed that only the odd multiple frequencies of 50 Hz are present and V_m forms a converging series. Now, what we do is we compare a perfect sine wave with the input wave. The phases of two waves are matched, i.e, one common point of their zeroes is matched such that both waves have a sign change from negative to positive at the common zero.

Now, the n^{th} Fourier coefficient of the input wave is found out by using the formula:

$$V_m = (2/T) \int_{-T/2}^{T/2} V(t) \sin(2m\pi t/T)\, dt$$

$$\cong \frac{4}{T} \sum_{k=0}^{n} V(t)[\sin(2m\pi\Delta t / T)]\Delta t$$

Now, $\Delta t = T/2n$ n being the number of samples taken in half loop

Therefore, $V_m \cong (2/n) \sum_{k=0}^{n} V(t) \sin(m\pi / n)$ ……………..(2)

V_m decreases in magnitude for a square wave as m is increased. For a perfect sine wave $V_m = 0$ for n $\neq 1$. However, since we assume our wave is 'between' a square and a sine wave, we can conclude that V magnitude decreases for increasing n for the input wave and V_m forms a convergent series. Since our analysis is in discrete time formula(2) is used to calculate a sufficient number of coefficients of V(t) or input wave. Then we can find out the Total harmonic distortion (THD) or the role of the secondary components in the wave by using the formula

$$\mathrm{THD} = \frac{\sqrt{V_2^2 + V_3^2 + V_4^2 + \cdots + V_n^2}}{V_1} \quad \ldots\ldots\ldots\ldots\ldots (3)$$

Software:

Programming the ATmega32 microcontroller to calculate and display the fraction of energy in various harmonics of the input wave was the key part of our project. For this purpose, we did all the programming in C language and used a cross compiler AVR Studio to get the corresponding hex file compatible with the microcontroller. The hex file thus obtained was burnt onto the microcontroller using the programmer circuit (Fig. 3) via the parallel port of the computer.

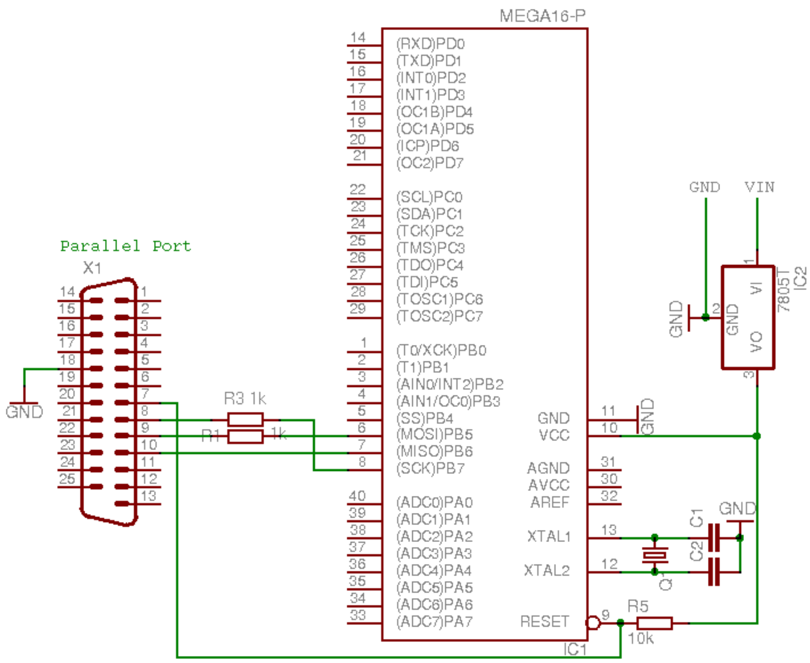

Fig.3

Logic for calculating Fourier Coefficients:

As we are taking only the sin terms, so we need to find the zero location to match the phase. So we are taking 250 samples from our ADC. Then we find out the local minima and assign it as our zero. This is because as ADC rectifies the input, zero is the only minima possible. Finally we take the two consecutive minimas to get the desired number of samples (n).

Calculation:

From series of experiments, it was found that 57 samples correspond to 50 Hz frequency.

By Nyquist theorem, Sampling Freq > 2 Band width

Here we are looking till 10th harmonic i.e. 500Hz, so Sampling Freq > 1K

i.e. Sampling rate of 10 samples is enough, which is true here.

Therefore:

Expected ADC conversion time = 10 ms/57 = 175 μs

Frequency = 1/ (2n*175 μs)

while finding local minima we are applying condition that our zero should be {<10 in binary (1024 is maximum)} so as to neglect the cases where local minima is there due to fluctuations.

Improvements done after first demo:
Earlier, our code was designed to measure distortion in 50 Hz waveforms only. Now we incorporate routines to determine the frequency of any given waveform and measure subsequent distortion of the input wave. Also now our project is on a pcb.

Experimental observations:

Providing main power supply we got the results-

Set 1
S=57
Freq = 50.00Hz
$A_0 = 1.3$
$A_1 = 0.11$
$A_2 = 0.266$
$A_3 = 0.208$
$A_4 = 0.148$
$A_5 = 0.108$
$A_6 = 0.085$
$A_7 = 0.067$
$A_8 = 0.054$
$A_9 = 0.043$
THD = 0.104

Set 2

S=58

Freq = 49.00Hz

A_0 =1.319

$A_1 = 0.108$

$A_2 = 0.289$

$A_3 = 0.221$

$A_4 = 0.154$

$A_5 = 0.109$

$A_6 = 0.084$

$A_7 = 0.066$

$A_8 = 0.052$

$A_9 = 0.043$

THD = 0.112

So we see harmonic distortion in the input wave to micro controller is of approx 0.1.

Sources of Error –

1. One major reason of error is transformer that we are using. It gets into saturation and there is a clear clipping in the output of transformer.
2. Phase matching- A small error is also caused because of the approximation that local minima is in phase with zero of ideal waveform.
3. Also calculating co-efficients using summation will lead to some error.

PCB Circuit Design: