Development of a Simple, Cost-effective µP based Harmonic Analyzer

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Abstract – The present era has witnessed rapid and wide spread advancement in the field of science and technology. With the advent of science and technology, human life has been made much simpler and faster. The same development has given rise to numerous problems among which harmonic distortion of voltage and current signals plays predominant role. To control this signal pollution IEC, IEEE and almost all national bodies have specified standards for each electrical and electronic equipments. This advancement has lead to the manufacture of highly sophisticated instruments that have excelled in their accuracy in measurement of harmonic contents. However, these instruments are very costly. This paper presents the details of the development of a cost effective, simple µP based harmonic analyzer along with its performance results.

Index Terms: Harmonic Analysis, Harmonic analyzers, Microprocessor based systems.

1 Introduction

Growth of science and technology has not only eased the human life, but also introduced some man made problems. One such problem is harmonic distortion. Harmonics are generated with the use of devices such as circuit breakers (high speed operation), induction furnaces, arc furnaces, rectifiers, inverters, thyristor controlled variable speed drives, slip energy recovery systems, saturable reactors, fluorescent lamps, computers and televisions. The reasons for the generation of harmonic currents in the system may be due to (a) Variable impedance offered by the device for each half cycle or (b) Generation of impulsive or non-sinusoidal back emf.

Sinusoidal signal is preferred in alternating current circuits, owing to its advantages such as differentiation & integration of a sinusoidal signal does not distort the signal, but only shift the signal backward or forward by \( \pi/2 \) radians. Because of this advantage, it is possible to tune the inductors and capacitors to give maximum system efficiencies. As in cases like series/shunt capacitor compensation in transmission lines, inductor tuning in series resonant circuits, IFT tuning in radio circuits etc. The other advantage of using the sinusoidal signals is that, it is possible to rebuild any irregular, haphazard wave using number of sinusoidal signals. However, all alternating waveforms observed on site, deviate to a greater or lesser degree from the ideal sinusoidal shape. Such waveforms are referred to as non-sinusoidal or complex waveforms.

With the increasing use of non-linear devices, high speed switching circuits, harmonic distortion of the voltage waveform is introduced. This distortion in voltage waveform is a problem that is receiving considerable attention from the site engineers. Thyristor converters and ac regulators are widely used due to their characteristics such as compact size, high efficiency, reliability, static and salient operation, less maintenance and quick response. However, they draw a non-sinusoidal current from the supply. Thus, serious draw back of thyristorised systems are the non-sinusoidal line current and poor power factor. As such, these line current harmonics may interfere with other loads connected to the same line and may cause disturbance on the neighboring communication lines. If low order harmonics are to be filtered, a large rating filter has to be used. This gives raise to considerable loss of efficiency and increase in the cost. Therefore, low order harmonics deserve more attention than the higher ones. Amongst the low order harmonics, third harmonics are given more importance, as they are the main cause for faulty operation of relays, more heating in rotating machines etc.

Power factor in a system with a non-sinusoidal current waveform is defined as \( \mu \cos \phi \) where \( \mu \) is the distortion factor and \( \cos \phi \) is the displacement factor. Distortion factor is given by the ratio \( \{I_{\text{fundamental RMS}} / I_{\text{RMS (Total)}}\} \) and the angle between the fundamental component of line current and line voltage is represented as \( \phi \). In case of line current being sinusoidal \( \{I_{\text{RMS (F) }} / I_{\text{RMS (Total)}}\} = 1 \). As such, since the beginning of the use of thyristors, attempts are being made to improve the power factor and to reduce the line harmonics. The power factor can be improved by improving both \( \mu \) and \( \cos \phi \). The induction of harmonics in the system has led to the degradation of the performance of the devices in the system and also reduced effective life. It is required to eliminate at least some of the harmful harmonics that obviously demand one to know their levels. Harmonic distortion of voltage waveform is introduced with the increasing use of non-linear devices and high speed switching circuits. Before going into the details of development of harmonic analyzer, the various causes of harmonic generation in the system are discussed in brief.

II Analysis of a complex wave

The sinusoidal components of a complex wave are called harmonics. Thus the signal with twice the fundamental is called second harmonic and third harmonic signal will have thrice the fundamental frequency and so on. In general, a wave may contain an infinite number of harmonics. However, for the purpose of analysis, only the first few harmonics are considered to simplify the complexity and also because of relatively smaller effect of frequencies. Based on this principle, harmonic analysis is derived.

Any complex wave can be obtained by the summation of all the contents of a particular wave. In general, any unsymmetrical wave may contain infinite number of harmonics. Fourier series are used for this purpose. Fourier series is one of the most easily comprehensible forms of harmonic analysis. Fourier’s theorem states that any continuous, single valued, periodic function may be split into a series of simple sine functions, this being true, however complex the original function [1]. Mathematically, this is represented as
\[ Y = f(x) = A_0 + A_1 \sin(x) + B_1 \cos(x) + A_2 \sin(2x) + B_2 \cos(2x) + \ldots + A_n \sin(nx) + B_n \cos(nx) \]  

Here, \( A_0 \) represents the dc component. \( A_1, B_1, \ldots, A_n, B_n \) are Fourier coefficients. It is apparent that each term in the above equation other than the dc term are sinusoidal in nature. Thus, the harmonic levels can be determined by expanding the function as a Fourier series. System based on this technique has been built and are fairly accurate. One such example is illustrated in Fig. 1. For illustration purpose a square pulse is considered, as majority of the current pulses are of this type. Similar techniques are adopted to design the harmonic analyzer.

### III Development of \( \mu \)P based Harmonic Analyser

#### Hardware:

The hardware section is responsible for the complete data acquisition operation. The input signal, being analog is sampled and quantized for use in the \( \mu \)P system. Figure 2 depicts the necessary block diagram of the hardware modules of a harmonic analyzer.

#### Signal reduction using potential and current transformers:

The signal is taken from any system as input to the analyzer using appropriate potential and current transformers. The ratio and phase angle errors of the instrument transformers must be known before they are used in the measurement.

#### Anti-aliasing filter:

Filter is an essential block in the development of the proposed harmonic analyzer. Here, filter is used to eliminate unwanted harmonic components of the input signal without affecting the measurement accuracy. A second order, low pass Butter Worth filter is most suited for this purpose.

#### Sample and hold circuit:

The sample and hold circuit is a digital controlled analog circuit that tracks the analog input signal during the hold mode to the instantaneous value of the signal, as the system is switched from sampled mode to hold mode.

#### Analog to digital converter:

The basic task of an analog to digital converter is to convert a continuous range of input amplitude into a discrete set of digital code words. AD574A is used in the present work as it matches with the frequency requirement (1600 Hz). The AD574A is a complete 12-bit successive-approximation analog-to-digital converter with 3-state output buffer circuitry for direct interface to an 8- or 16-bit microprocessor bus with maximum conversion time 35 \( \mu \)S. A high precision voltage reference and clock are included on-chip, and the circuit guarantees full-rated performance without external circuitry or clock signals.

#### Software:

Software implementation of digital algorithm is realized using the DSP computation. Software module includes data shuffling routine, two point FFT routines; twiddle factor determination routine, complex multiplication routine, data acquisition routine and butterfly computation. Fig. 3 shows the flow chart used for one subroutine in the computation of FFT.

In the present work the input data is assumed to be real and each data is of 8-bit length. The data shuffling operation is done according to the bit reversal rule. The 2-point FFT of every two contiguous data points is computed. The various program parameters are initialized. A dual node pair number counter is initialized to 1. The value of the variable \( P \) is determined so that an offset addressed to access the table of complex constants is computed. The actual computation upon which the successive block increments the node count is performed, signifying the progress of computation down the computational array. Depending on the dual node...
counter I, nodes to be skipped are checked. If the computation
in the array is complete, necessary parameters for the next
array computation are reinitialized and it is checked if all the
arrays have been computed.
For the system under design, the highest frequency in the
signal is limited to 1.6 kHz since the design is limited to
measure signals up to the 32nd harmonic of the fundamental
50 Hz signal. This means
\[ T_{<}=\frac{1}{2f_0} \text{ or } \frac{1}{2f_0} = \frac{1}{(2 \times 1.6 \times 1000)} = 0.3125 \text{ mS} \]
The sample time \( T \) is constrained by
\[ T_{<}=0.3125 \text{ mS} \]
For a frequency resolution of 50 Hz, record length (in
seconds)
\[ t_p=\frac{1}{F}=\frac{1}{50}=0.02 \text{ S} \]
Therefore the number of points must satisfy
\[ N_{\geq} \frac{0.02}{0.3125 \text{ mS}} = 64 \]
Satisfying the requirement \( N=2\gamma, N = 64 \) is an appropriate
choice.

**IV Results and Validation**

First the designed Butter worth filter output was tested in the
laboratory for different types of input waveforms. The output
obtained by the filter is depicted in Figures 4-6. It is recalled
here that the filter is designed for 1600Hz and the output
obtained is satisfactory. Fig.7 shows the frequency response
of BWF.

For testing purpose, a symmetrical square wave (which is an
ideally infinite source of harmonics of varying peak
magnitudes in V), is considered. The first ten harmonics are
evaluated by this system using a 32-point DFT and its values.
These values are then compared with the computer results for
validation. Table I shows the comparison between the results
obtained in proposed scheme (magnitude) and the computer
results. From the Table I it is evident that the results agree
fairly well.

**Table I: Comparison Table**

<table>
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<tr>
<th>Harmonic number</th>
<th>HEX Real</th>
<th>HEX Imaginary</th>
<th>Results obtained in proposed scheme (magnitude (V))</th>
<th>Computer results Magnitude (V)</th>
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<td>EBCC</td>
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**V Conclusions**

The fact that only an 8085 based system has been used to
achieve this kind of result speaks volumes on both the
programmable capabilities of 8085 and cost effectiveness
of the system. The 8085 have been used to control
operations ranging from control of the data acquisition to
the FFT computation and display of the harmonic
waveforms. Some of the other merits of this system are:
- A 32-bit square root routine has been developed based
  on Newton-Raphson algorithm for use in connection with
  the magnitude and phase angle calculations. This
  32 bit operation has been implemented on the 8085,
  contrastingly, an 8 bit processor.
- System could be used as a total for computing the
  discrete Fourier transform of the input sequence.
Small size and the rigidity of 8085 based microcomputer used to achieve the results, make it very much reliable. There is also the advantage due to standardization of the hardware on the board of the system. Thus, upgrading and fixing the system is made simple. All these advantages stress the fact that such a microprocessor based system is indeed a low cost solution to the problem of harmonic level measurement.

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VII Bibliography

