An Adaptive filter for Harmonic Elimination

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Summary
In this brief, we propose a novel filter to eliminate undesirable harmonic from composite signal using RLS algorithm. The proposed approach eliminates all harmonics component from selected variable (current or voltage) and it requires only knowledge of the frequency of the fundamental component. Using frequency of fundamental component of composite signal, the outputs of the adaptive filter is a harmonics replica and is subtracted from the original composite waveform to eliminate them. The bipolar waveforms are roughly analyzed and considered case of square wave pattern which contain all odd harmonics. The simulation results show that the method can effectively eliminate undesirable harmonics and sine wave is produced.
Keywords:
Harmonics Elimination, RLS Algorithm, Square wave generator

I. INTRODUCTION

A harmonic is a signal or wave whose frequency is an integral (whole-number) multiple of the frequency of some reference signal or wave [1]. For a signal whose fundamental frequency is f, the second harmonic has a frequency 2f, the third harmonic has frequency of 3f, and so on. Signal occurring at frequencies of 2f, 4f, 6f, etc. are called even harmonics; the frequencies 3f, 5f, 7f etc. are called odd harmonics. If all the energy in a signal is contained at the fundamental frequency, then that signal is a perfect sin wave. If the signal is not a perfect sine wave, then some energy is contained in the harmonics. Examples are square wave, saw tooth wave and triangular wave.
It is well know that any periodic waveform such as that mentioned previously can be represented by Fourier series, an infinite sequence of sine and cosine waves, at the fundamental frequency of the waveforms and its harmonics. These harmonics can cause trouble in several areas particularly in motors and sensitive application.
The coefficients of the Fourier series are computed with a pair of integral's that produces the coefficients of the sine and cosine terms in series. For a signal f(x) with a zero dc component, the integrals are

$$an=(1/\pi)\int f(x)\cos(nx)\,dx \qquad n>0$$
$$bn=(1/\pi)\, f(x)\sin(nx)\,dx \qquad n>0$$

where the an and bn terms are the coefficients of the cosine and sine terms, respectively, in the series. The Fourier series is then:

$$f(x)=a_1\cos x + a_2\cos 2x+a_3\cos 3x\text{-----}+b_1\sin x + b_2\sin 2x+b_3\sin 3x+\text{----}$$

In conventional square wave have both half-wave symmetry and quarter-wave symmetry, integration is required only over one-quarter of the waveform, and further that only the sine terms and odd harmonics are required. Thus, the integral used to compute the coefficients for the conventional square wave becomes

$$bn=(4/\pi)\ f(x)\sin(nx)\,dx$$
$$(4/n\pi) \quad \text{for odd values of n only.}$$

The series is then $(4/\pi)\sin(x)+(4/3\pi)\sin(3x)+(4/5\pi)\sin(5x)+$----------

The standard measure for distortion is Total Harmonic Distortion (THD). Numerical evaluation of the coefficient for the square wave indicates that if the square wave is to be considered a sine wave with distortion, the THD is in the range of 45%(-7dB). The third harmonic, the hardest to filter out, is one-third the magnitude of the fundamental (-10dB).
A modulation –based method for generating pulse waveforms with selective harmonics elimination is proposed in [2]. Another method to eliminate harmonics in multilevel converters is highlighted in [3].The main challenge is associated with elimination of all harmonics.
This manuscript proposes an approach for harmonic elimination based an adaptive filter technology is developed. The task is accomplished by generating harmonics replica using fundamental frequency of composite signal. The output of adaptive filter is a harmonics replica and is subtracted from the composite signal of fundamental frequency to eliminate them and the total harmonic distortion (THD) is greatly reduced. The weights of filter are adjusted on-line by using RLS adaptive filtering algorithm. The above course is used as a basis for the development of adaptive harmonics elimination algorithm. From the comparison result, it is

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seen clearly that this method has better harmonic elimination efficiency and faster weight convergence.

II.PROPSED HARMONIC ELIMINATION FILTER

The task of eliminating an undesirable harmonic component from a signal can be done by proposed approach. The circuit consists of a summing point and recursive least square (RLS) adaptation Algorithm. it operates in the following way:

a) The signal of frequency wo=2 π fo eliminated from composite signal. Elimination of fundamental frequency fo is obtained using RLS algorithm.
b) Adaptation process adjusts weights to exactly match amplitude and phase of fundamental frequency component.
c) The signal created by a filter circuit is subtracted from the primary input, such that the output harmonics are cancelled leaving the desired fundamental frequency signal alone.

The RLS adaptation algorithm as developed in [4] will discussed as Steps in calculation of filter coefficients are

i) $K(n)=\lambda^{-1}P(n-1)u(n)\{1+\lambda^{-1}H(n)P(n-1)u(n)\}^{-1}$

ii) $y(n)=\hat{w}H(n-1)w(n)$

iii) $e(n)=d(n)-y(n)$

iv) $\hat{w}(n)=\hat{w}(n-1)+k(n)e^*(n)$

v) $P(n)=\lambda^{-1}P(n-1)-\lambda^{-1}k(n)u\,H(n)P(n-1)$

The RLS algorithm uses the information contained in all the previous input data to estimate the inverse of the autocorrelation matrix of the input vector. It uses this estimate to properly adjust the tap weights of the filter. In the above equation, P corresponds to the inverse of the autocorrelation matrix of the input; K is a quantity called the gain vector. Lambda is the forgetting factor, which tells the filter to forget earlier inputs. In general, it seems to converge within 2M iteration, where M is the number of tap weights. This is well over an order of magnitude increase in performance versus LMS algorithm [5].RLS algorithm has significantly faster convergence behavior than LMS algorithm. For the RLS algorithm the behavior of the filter coefficients is much more stable. They adhere much closer to the correct values.

III. ANALYZING HARMONICS

Harmonics are AC voltages and currents with frequencies that are integer multiples of fundamental frequency. On a 50Hz system, this could include 2nd order harmonic100Hz, 3rd order harmonic (150Hz), 4th order harmonics (200Hz), and so on.

The flattened and dimpled sinusoid in fig.1 has the mathematical equation y=2sin (2π50) +0.5sin (3*2π 50). This means a 50Hz sinusoid(the fundamental frequency) added to a second sinusoid with a frequency three times greater than the fundamental (150Hz) and an amplitude ¼(0.5times) of the fundamental frequency.

Similarly the peaky sinusoid in fig. 2 has the mathematical equation y=2sin (2π50)-0.5sin (3*2π50). This waveform has the same composition as the first waveform except the third harmonics component is out of phase with the fundamental frequency, as indicated by the negative sign preceding the "0.5sin(3*2π50)"term. The waveform in fig. 3 contains third, fifth, seven, nine and eleven harmonics in phase with the fundamental frequency. The waveform in fig. 4 contains several other harmonics in addition to the second harmonic some are in phase with the fundamental frequency and others out of phase

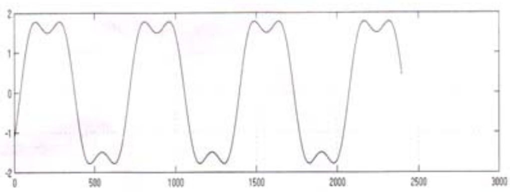


Fig.1. flattened and dimpled sinusoid waveform

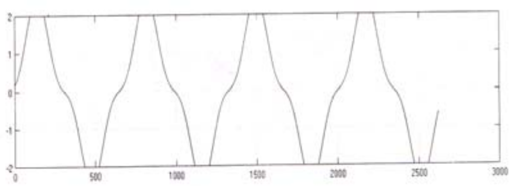


Fig.2 peaky sinusoid waveform

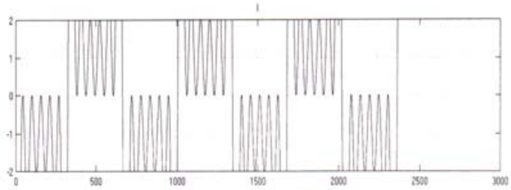


Fig.3. The wave contains third, fifth, seven, nine and eleven harmonics

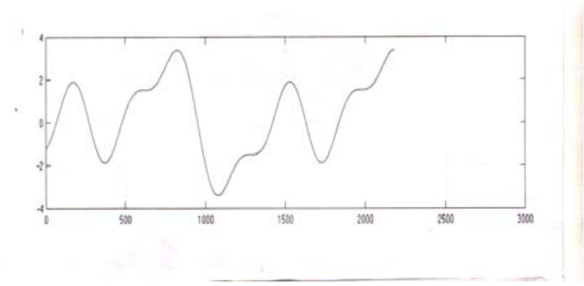


Fig. 4. Contain several other harmonics in addition to the second harmonic

As the harmonic spectrum becomes richer in harmonics the waveform takes on more complex appearance, indicating more deviation from the ideal sinusoid. A rich harmonic spectrum may completely obscure the fundamental frequency sinusoid, making a sine wave unrecognizable. When the magnitude and order of harmonics are known, reconstructing distorted waveform is simple.

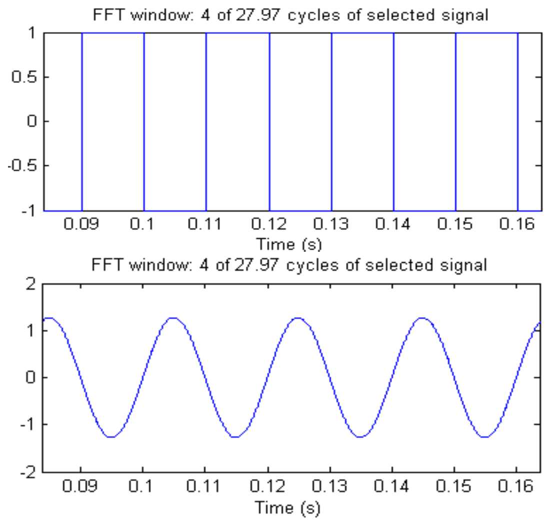
Decomposing a distorted waveform into its harmonic component is considerably more difficult. This process requires, Fourier analysis, which involves a fair amount of calculus. However, electronic equipment has been developed to perform this analysis on a real time basis.

One of the strategy is to reduce the magnitude of the harmonic waveform, usually by filtering. The other method is to use system components that can handle the harmonics more effectively, such as K-factor transformers. Harmonic filters can be constructed by adding an inductance (L) in series with a power factor correction capacitor(C). The series L-C circuit can be tuned for a frequency close to that of the troublesome harmonic, which is often the 5th by tuning the filter in this way, you can attenuate the unwanted harmonic. This can be a very cost effective means of reducing harmonics. Because of the computation difficulty of the resultant method to eliminate all harmonics an approach is propose using RLS Algorithm.

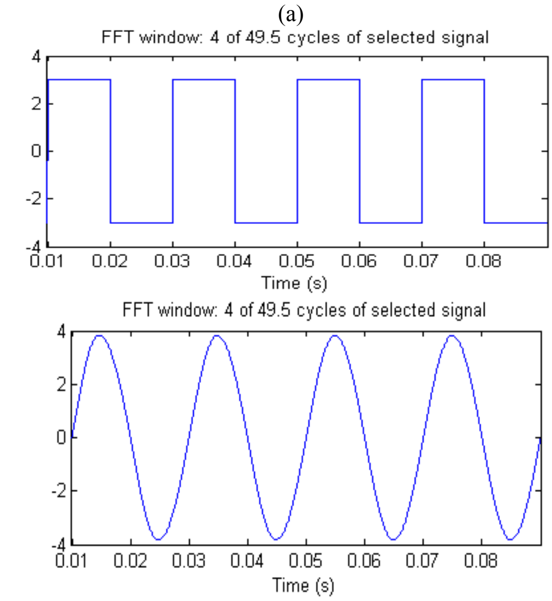
IV. PROPOSED SCHEME

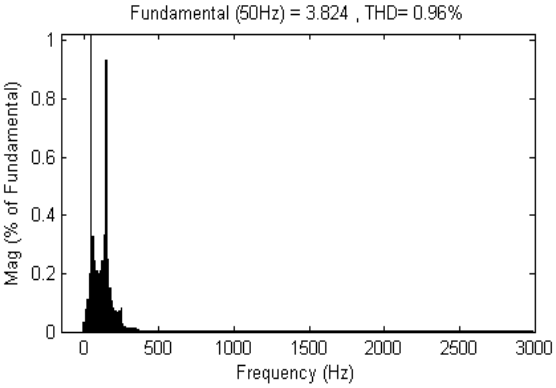
A single-phase square wave inverter has the least desirable output waveform type; a square wave [6],[7] is sort of a "flattened-out" version of a sine wave. Generally, square wave output inverter available in market is simple in design and low cost. Such inverter has some disadvantages. For starters, the peak voltage of a square wave is substantially lower than the peak voltage of a sine wave. In addition, a square wave contains many higher frequencies as well, called harmonics, which can cause buzzing or other problems The proposed system can be used to eliminate higher order harmonics from a periodic wave form waveform effectively and result in low total harmonic distortion (THD). This system is also useful to detect wheather the specific harmonic is present in composite signal or not.

A novel approach is proposed in this paper to eliminate all harmonics from square wave waveform. A square wave output waveform which contain many odd harmonics (ex.3rd, 5th, 7th etc.)First, a fundamental frequency of square wave is applied to RLS algorithm and RLS adaptive filter gives output of harmonics replica present in square wave output wave. Next, harmonics replica is subtracted from square wave output. Therefore, the output voltage wave is sinusoidal waveform of fundamental frequency. The proposed scheme is validated under no load condition. Input and output waveforms are shown in fig 5 using matlab/Simulink.



(a)





(b)

Fig. 5. Performance of the proposed scheme.
(a) Input and output waveforms for 1V, 50Hz
(b) Input and output waveforms for 3V, 50Hz

V. CONCLUSION AND WORK IN PROGRESS

In this brief, we proposed and developed a filter to eliminate undesirable harmonics from composite signal. The simulation results show that the proposed filter can be used to eliminate higher order harmonics effectively and result in low total harmonic distortion (THD). This filter is also useful to detect the specific harmonic present in composite signal or not. The simulation result show that the method can effectively eliminate all odd harmonics from square wave and sine wave is produced. The advantages of this approach are simple design, low THD and cost effective. We are currently working on further analysis by considering different load (inductive and capacitive) and hardware implementation of proposed scheme.

REFRENCES

[1] Cyril W. Lander,Power Electronics, Singapore, McGRAW-HILL. International limited, 1993.pp 295-333.
[2] Brett M. Nee, Jason R. Wells, Xin Geng, Patrick L. Chapman and Philip T. Krein, “Modulation-Based Harmonic Elimination,” IEEE Trans. Power Electron. vol. 22 no.1, pp.336-339, Jan 2007.
[3] Zhong Du, Leon M. Tolbert and John N Chiasson, “Active Harmonic elimination for Multilevel Converters,” IEEE Trans. Power Electron. vol. 21 no.2, pp.459-469, March 2006.
[4] Emmannel C. Ifeachor and Barrie W. Jervis, Digital signal processing, Singapore,Pearson Education, pte. Ltd., 2002.
[5] B. Window and S. D. Stearns, Adaptive signal processing. Englewood cliffs, NJ: Prentice-Hall, 1985.
[6] Ashfaq Ahmed, Power Electronics for Technology,Singapore, Pearson Education Pte. Ltd., 2003. pp. 304-312.
[7] Philip T. Krein, Elements of power Electronics,New York, Oxford University Press, Inc.,2004, pp. 206 -212.

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