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A Novel Small Signal Power Line Quality Measurement System

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Abstract - **We investigate a technique to measure low level power line harmonics using adaptive cancellation of the fundamental component. The technique reduces the effect of quantization noise and extends the dynamic range of the analog-to-digital converter (ADC). It also enables good accuracy with a reduced bit ADC.**

Keywords - **Adaptive filtering, powerline quality measurements, power line harmonic measurements**.

I. INTRODUCTION

In order to characterize power line quality for a 60 Hz power transmission line it is necessary to accurately and precisely measure the voltage levels produced by the harmonic content of the 60 Hz signal. This is normally accomplished by digitizing the transmission line voltage via an analog-to-digital-converter (ADC) with the values fed to a computer system to be recorded and processed. While in theory this is readily done, in practice because the ADC has a limited *dynamic range*, it is relatively difficult to economically measure small signals in the presence of a strong 60 Hz component. Dynamic range is the ability of an ADC to digitize signals of various amplitudes so that large signals do not overload the ADC and neither do small signals get buried in the quantization noise. For example, a 12-bit ADC with +/- 10 volt input has 2^{12} = 4096 quantum levels and thus can accurately digitize signals that range from +/-10 volts to +/- 0.24 mV. A similar 16-bit ADC can digitize signals that range from +/- 10 volts to +/- 0.015 mV. However, the cost of the 16 bit ADC is much greater than the 12 bit ADC.

To understand the difficulties involved, let's consider measuring the harmonics of a 10,000 volt-peak-to-peak (V-p-p) 60 Hz transmission line that has been scaled down by a factor of 1000 and inputted to the above 10 volt ADC. With a 10 KV-p-p input, we can digitize harmonics as low as +/- 0.24 V-p-p. If we had a similar 16 bit ADC, we could digitize harmonics as low as +/- 0.015 V-p-p.

II. PROPOSED SOLUTIONS

There are several methods that could be employed to extend the ability of an ADC to accurately measure small voltage quantities in the presence of a large signal. The first is using a process known as *companding* whereby we first *compress* the input signal using a logarithmic amplifier prior to digitization and then later *expand* the signal [1]. Companding increases the significance of small signals and de-emphasizes the larger signals. However, good quality time-invariant logarithm circuits are relatively expensive and still may not have the precision we need.

Another approach is to minimize the 60 Hz signal altogether and prevent it from getting into the ADC and so cause the harmonics to occupy the full dynamic range of the ADC. Assuming the 60 Hz component is 1000 times, greater than the harmonics, its elimination would increase our dynamic range by a factor of 1000 and thus in the above 10 KV case, 12 and 16 bit ADCs could quantize signals as low as 0.24 mV p-p and 0.015 mV p-p respectively.

Removing the 60 Hz fundamental component could be done in two ways. The first is to use a notch filter where the 60 Hz signal is filtered out. This solution has several difficulties however. In our 10 KV example, removing the 60 Hz component so that the harmonics occupy the full ADC dynamic range would require a notch filter with an attenuation factor of 1000 or 30 dB. This is not easily done with discrete analog components and is virtually impossible to do on an integrated circuit (IC). Furthermore, its center frequency is not easily changed and therefore would not be easily adaptable to situations where the line frequency varies. Finally, to measure even the smallest higher order harmonic(s) may require us to also remove any significant 2nd, 3rd or higher order harmonic component. Adding additional notch filters to do this would be relatively cumbersome.

We propose using *adaptive cancellation* [2] to *eliminate* the 60 Hz component from getting into the ADC. This system has the following advantages over the notch filter: (a) the entire measurement system could be put on an IC, (b) it will work in cases where the fundamental frequency significantly deviates from 60 Hz, and (c) is easily expanded to cancel out other higher order harmonics.

III. ADAPTIVE CANCELLATION SOLUTION

The idea of adaptive cancellation is to eliminate the 60 Hz component and leave us the harmonic function to be digitized and measured. Let Eq. (1) describe a power line signal with harmonic content:

$$x(t) = A_1 \cos(2\pi 60 t + \theta_1) + \sum_{n=2}^{N} B_n \cos(2\pi 60 n t + \phi_n) \qquad (1)$$

where the first term represents the 60 Hz signal and with A_1 and θ_1 the respective magnitude and phase values. The second term represents the power line harmonic content with B_n and ϕ_n the respective magnitude and phase of the n_{th} harmonic. It should be noted that we are ultimately interested in digitizing and measuring only harmonics. It is possible for some deviation in the 60 Hz frequency to occur and so Eq (1) can be modified to be

$$x(t) = A_1 \cos(2\pi f_1 t + \theta_1) + \sum_{n=2}^{N} B_n \cos(2\pi 60 n t + \phi_n) \qquad (2)$$

where $f_1 = 60 \pm \Delta$ is the fundamental line frequency and Δ is the amount of deviation from 60 Hz. The underlying assumption for this project is that the overall power quality is somewhat reasonable and that we are really trying to accurately measure trace amounts of harmonic content. We can therefore say that $B_n \ll A_n$.

Now let's consider the adaptive cancellation system shown in Fig 1. As indicated, we synthesize a replica of the 60 Hz term in Eq. (2) giving

$$y(t) = M_1 \cos(2\pi f_1^{'} t + \theta_1^{'}) \qquad (3)$$

The $y(t)$ and $x(t)$ signals are then subtracted giving us

$$\begin{aligned} \varepsilon(t) &= x(t) - y(t) \qquad (4) \\ &= \left[A_1 \cos(2\pi f_1 t + \theta_1) + \sum_{n=2}^{N} B_2 \cos(2\pi 60 n t + \phi_n) \right] \\ &\quad - M_1 \cos(2\pi f_1^{'} t + \theta_1^{'}) \end{aligned}$$

By adjusting the values of M_1, $f_1^{'}$, and $\theta_1^{'}$ to be equal to A_1, f_1, and θ_1 respectively we get

$$\varepsilon(t) = x(t) - y(t) = \sum_{n=2}^{N} B_2 \cos(2\pi 60 n t + \phi_n) \qquad (5)$$

Function $\varepsilon(t)$ represents the harmonic content of the power line signal we are ultimately interested in digitizing. It is then fed to the ADC and computer system.

It should be noted that generating $\varepsilon(t)$ involves minimizing function $|\varepsilon(t)|$. This is done by adjusting the values $M_1, f_1^{'}$, and $\theta_1^{'}$ to minimize function $\varepsilon^2(t) = [x(t) - y(t)]^2$. There are several to perform this minimization process. If the respective functions are smooth and do not have local minima, we can use a gradient algorithm to solve for the respective values of $M_1, f_1^{'}$, and $\theta_1^{'}$ that minimize $\varepsilon^2(t)$. If the functions are more complicated we could resort to more sophisticated algorithms such as linear and nonlinear programming, genetic algorithms or simulated annealing [3].

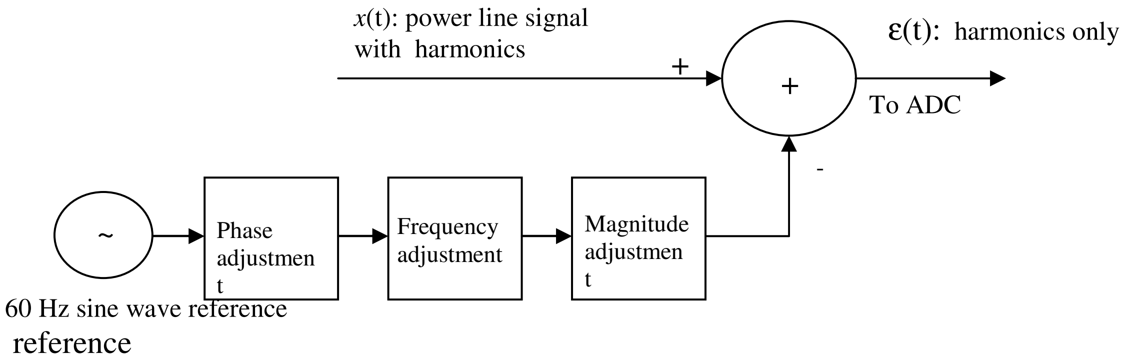

Figure 1. Adaptive cancellation filter to remove 60 Hz component.

If the system has other significant components that are preventing us from accurately digitizing trace harmonic content, we could extend the adaptive cancellation filter to also remove these high level signals. We merely generate a replica(s) of the offending signal(s), subtract it from the power line signal, and adjust the magnitude(s), phase(s) and frequency(s) values to minimize the absolute value of the summer's output.

IV. EXPERIMENTAL

Figure 2 describes the practical implementation of the adaptive cancellation system. It works as follows. Function x(t) is the 60 Hz input signal with the harmonics that we wish to measure. The reference signal used to cancel out the 60 Hz component in x(t) is generated via a digital source and then converted to analog form using a digital-to-analog converter (DAC) and a Bessel low pass filter. The relatively pure 60 Hz reference output of the Bessel filter is then subtracted from x(t) via a summing module giving us an error function, ε(t). Function ε(t) is then digitized giving us ε(n) which used by the least-mean-squared (LMS) module to adjust the reference's magnitude and phase causing the 60 Hz component in x(t) to be mostly canceled out. Thus, function ε(n) contains only the harmonic content which is what we wish to accurately measure. In this particular case, it was assumed that there was only a global minima and therefore the LMS module uses the conventional gradient and minimum mean-squared error algorithm to adjust the reference signal's magnitude and phase in order to minimize error term $|\varepsilon(n)|^2$. Note that in this particular case, we assumed that the 60 Hz line frequency is a given and therefore, only the magnitude and phase will be adjusted to cancel the 60 Hz component.

Test signal x(t)'s spectrum is shown in Figure 3. Note that the input signal has frequencies of 60 Hz, 120 Hz, 180 Hz and 240 Hz with the harmonic amplitudes being 1/64 of the 60 Hz amplitude. As Figure 3 indicates, the harmonics are barely observable in the frequency domain. Figures 4 is the ideal harmonic spectrum with Figure 5 actual result using the system of Figure 2. Figure 6 is a multidimensional plot showing error versus reference amplitude and phase.

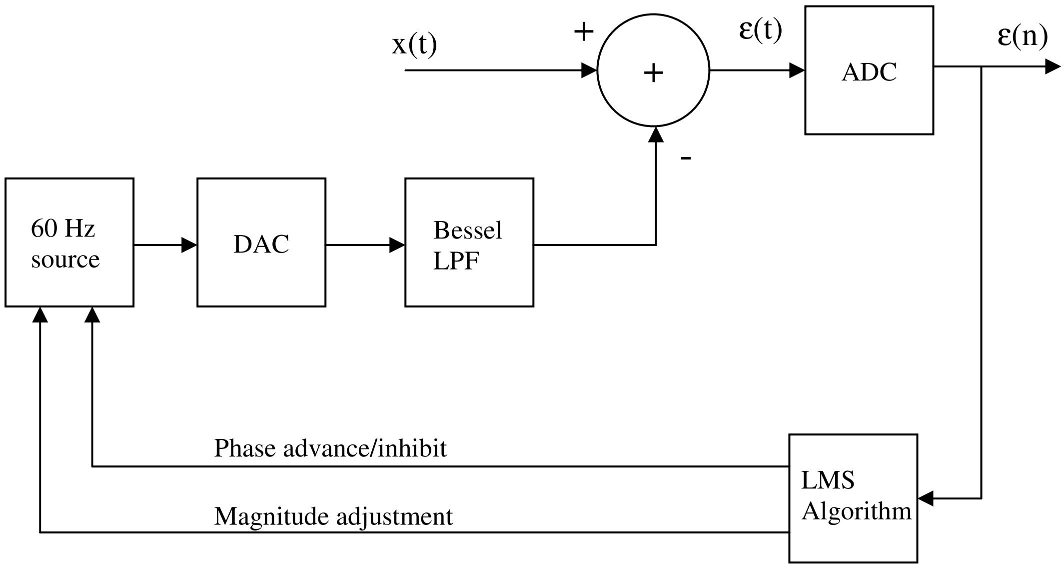

Figure 2. Test system for adaptive cancellation filter to remove 60 Hz component.

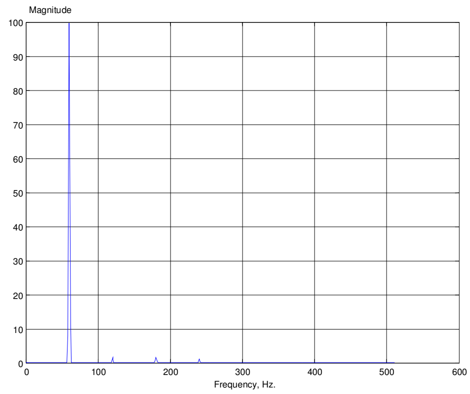

Figure 3. Spectrum of input signal

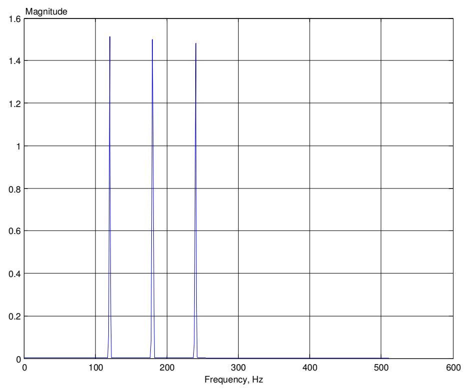

Figure 4. Expected spectrum of harmonic content.

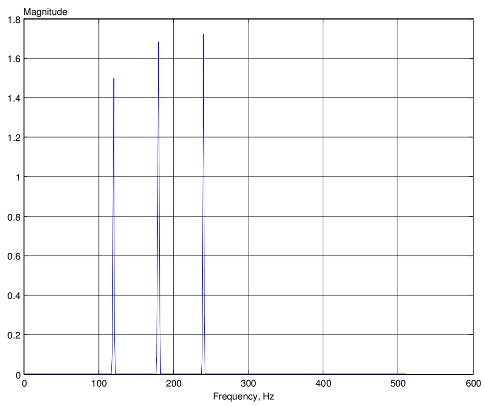

Figure 5. Actual spectrum obtained by system.

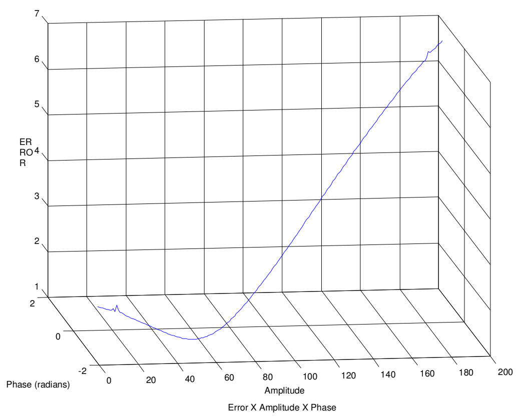

Figure 6. LMS error versus phase and magnitude.

V. CONCLUSIONS

We have described a system that causes the relatively large 60 Hz fundamental component to be canceled from a power line signal and thereby maximizing the dynamic range of the ADC. This in turn enables us to accurately measure these harmonics. This of course is a preliminary investigation and more testing of the method has to be done. We have also not considered signals that have been corrupted by random noise.

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