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DEVELOPMENT OF A POWER LINE TRANSCEIVER WITH DSP BASED ADAPTIVE HYBRID ECHO FILTERING FOR SIGNAL TRANSFER

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ABSTRACT

Suitably designed transceivers would enable signal transfer in power line which may be an alternate low cost option in monitoring plants and Small Office Home Office setting. In this paper, a transceiver is designed to enable full duplex analog/digital signal communication through the power line using amplitude modulation. The same having a digital signal processor based echo canceller as its hybrid component is optimized for adaptive filter parameters like filter length, step size and convergence time. Convergence time decreases with increase of filter order and step size (for Least Mean Square implementation). The Normalized Least Mean Square Algorithm showed better results than Least Mean Square Algorithm. Using the arrangement, analog signals like that from signal generators and audio source and data files from computers are transferred through a small room over a maximum length of 10m. The design of the transceiver and the study of the adaptive filter is presented in detail in the paper.

KEYWORDS

Power Line Communication, Full Duplex, Digital Signal Processor, Adaptive Filtering.

I. INTRODUCTION

Power Line Carrier Communication commonly known as PLCC [1] provides a low cost infrastructure for signal transfer in data monitoring plants, industries and Small Office/Home Office (SOHO) setting. This however necessitates adequate solutions to associated problems like high noise, attenuation and unpredictable channel conditions [2]. Literature survey shows that a large number of transceivers are designed for different applications like automatic meter reading, resource sharing and monitoring [3-5]. Moreover equipments like computers, telephone receivers, data monitoring systems etc can with slight modification enable data/signal to be transferred through a pair of power line wires in suitable phase-neutral, phase-ground or neutral-ground combinations. The equipments however cannot be connected to the line directly due to usual low power of the message signals and high noise of the channel. Attempts have been made to design a workable transceiver for communication in indoor power line using single channel [6], dual channel [7] and multi-channel modes [2]. Multi-channel modes using OFDMA techniques are efficient but are difficult to implement [2]. Channels having a single bandwidth for bidirectional transmission are easier to implement but has to take into account advanced mechanisms for reducing the echo in the hybrids [3]. As such, in this paper, we present the development of a transceiver using analog modulation scheme for full duplex communication in a SOHO setting. Although this modulation is the least used due non-immunity to noise, the scheme is chosen as it provides a worst case scenario to test modulation schemes. In Section 2, an initial study of the power line noise necessary for channel selection is presented. Section 3 and 4 describes the working of the transceiver and the necessity of an echo canceller (EC) in the hybrid. In Section 5, an EC using adaptive filtering is implemented on a digital signal processor (DSP) and the system is tested using signal from signal generators, audio source and computers (Section 6). Least mean square (LMS) and normalized least mean square (NLMS) algorithm are implemented and optimized for filter parameters like filter length (FL), step size (μ) and convergence time. A brief discussion on the hardware setup and results obtained are presented in the paper.

II. POWER LINE NOISE ANALYSIS AND CHANNEL SELECTION

Power line noise is reported to be Non-Gaussian and has a much localized behaviour [6-9]. As such the preliminary requirement for the development of any transceiver is make an estimate of the noise scenario prevalent in the channel. Extensive studies shows that the noise in the experimental place is a combination of background noise, impulses both asynchronous noise and synchronous to the AC mains [8] and narrowband interferences at the transmitting frequencies of the nearby radio station (730 kHz and 1.30 MHz). The Power Spectral Density (PSD) decreases with frequency and is very high of the order of -70dB/Hz in the CENELAC band (<150 kHz used in USA) [9], -100dB/Hz in the FCC band (<530 kHz used in Europe) and as low as -130dB/Hz at 10MHz [8]. As such, we choose the band of frequency to be in the FCC band to reduce the necessity of high transmitting power. Although PLCC Standard are not extensively developed in India, the power line wiring here is more like that of Europe than USA[8].

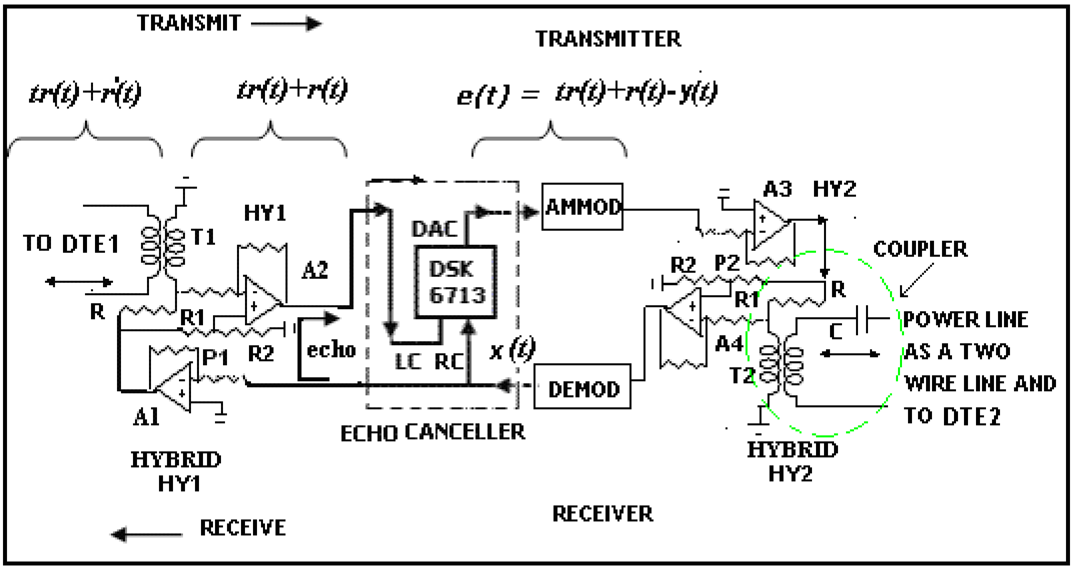

Figure 1: The design of the transceiver system with the echo canceller

III. DESIGN OF THE TRANSCEIVER

The system or transceiver designed for signal transfer requires two hybrids (HY1 AND HY2), a transmitter, a receiver and a coupler (Figure 1). The purpose of HY1 is to convert the two wire line acting as the output line of the first *data terminal equipment* (DTE1) to be networked through the power line to a four wire line thus separating the transmit $tr(t)$ from the receive signal $x(t)$. The transmitter is a simple amplitude modulator (AMMOD) that does the necessary signal conditioning like modulation (carrier frequency of 200 kHz), amplification etc as required. The amplified signal is transmitted to the power line through another hybrid circuit (HY2) that converts the four wire line back to a two wire line through a coupler. The coupler is a combination of a the secondary of the transformer (T2) and a capacitor (C) that acts as a high pass filter which filters off the 230V and 50Hz power line signal from the transceiver [6] and enables easy flow to the high frequency message signal. DTE1 is connected to the power line through the transceiver by plugging on the coupler to the same. The second *data terminal equipment* (DTE2) to be networked is connected to the other end of the power line through a similar transceiver. The receiver of the first transceiver (DEMOD) demodulates/ detects/filters the receive signal after receiving the same from the power line through HY2. The

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receiver passes the message signal $x(t)$ on to DTE1 through HY1. The construction of the hybrid and the necessity of an echo canceller are given in the following section.

IV. CONSTRUCTION OF THE HYBRID AND NECESSARY ECHO CANCELLATION

The hybrids (HY1/ HY2) consists of 1:1 transformers (T1/T2), buffer amplifiers (A1/A3) and differential amplifiers (A2 / A4).The primary of T1 and the secondary of T2 are connected to the output of DTE1 and to the coupling capacitor C respectively. The windings of the transformer are bifilar windings on a ferrite EE core to minimize the effect of capacitance between the windings for high frequency operation. HY1 combines the receive signal $x(t)$ from the receiver and the transmit signal $tr(t)$ from the DTE1 to provide a two wire line for bi-directional flow. In the same way, HY2 converts the two wire line (provided by the power line) into four wire line and separates them into the transmit and the receive signals. Due to impedance mismatch at the output of the DTE1, a part of $x(t)$ is reflected ($r(t)$) in the transmit line of HY1 and combines with the same forming the modulating signal $tr(t)+r(t)$ to the AMMOD. Similarly, impedance mismatch caused by the power line causes a part of the modulated signal to be reflected in the receive line in HY2 which combines with $x(t)$. The hybrid echo, if of considerable magnitude will act like a noise and cause distortion of the transmit/ receive signal and decrease the *SNR* of the signal. The potentiometer P1/ P2 connected to the differential amplifiers A2/A4 are adjusted to minimize the echo. The echo is completely cancelled if the hybrid gain *G* of A2/A4 is equal to zero which is possible only when T1/T1 is resistive equal to R and $A = R_2/(R_1 + R_2)$ is 0.25 [10]. However, in the presence of T1/T2, the minimized echo is a non zero quantity and decreases with the increase of the number of winding and equal to zero if the impedance of the primary is infinite which is however not possible [6]. This limitation permits only ~ 50% of the echo (with an echo return loss (ERL) of 6dB) to be cancelled in the transmit/receive path considering the input impedance of DTE1/ power line to be high. As such, the hybrid echo noise is much larger than the power line noise thereby decreasing the transmit *SNR* (equation 1) to as low as 3.65dB (Table 1, Sl 1) values compared to *SNR* of 56.84dB in the presence of power line noise only (*echo*=0, in the absence of hybrid noise).

$$SNR = 20\log\left(\frac{S}{Noise + Echo}\right) \qquad (1)$$

Table 1. Echo Cancellation Parameters for signal from signal generators (Sl1 to 19) and audio source (Sl 20 to 26)

Sl No	Setup	Algorithm	N	µ	ERL	ERLE	Residual Echo(dB)	T(sec)	SNR (dB)
1	No EC	NIL	NIL	NIL	6dB	NIL	-1.2	NIL	3.65
2	With EC	LMS	46	5	do	<-57.8	<-57.8	No output	-
3		LMS	do	4	do	do	do	<1	53.24
4		LMS	do	1e-2	do	do	do	<1	do
5		LMS	do	1e-3	do	do	do	3.36	do
6		LMS	do	8e-4	do	do	do	4.15	do
7		LMS	do	5e-4	do	do	do	5.70	do
8		LMS	do	3e-4	do	do	do	10.24	do
9		LMS	do	1e-4	do	do	do	28.77	do
10		LMS	35	3e-4	do	do	do	No output	-
11		LMS	40	do	do	do	do	12.34	do
12		LMS	46	do	do	do	do	10.35	do
13		LMS	50	do	do	do	do	8.97	do
14		LMS	56	do	do	do	do	7.81	do
15		LMS	60	do	do	do	do	6.69	do
16		LMS	100	do	do	do	do	4.41	do
17		NLMS	2	-	do	-27.8	-33.8	<1	37.6

18		NLMS	4	-	do	-31.0	-45	<1	47.62
19		NLMS	6	-	do	<-57.8	<-57.8	<1	53.24
20		NLMS	46	-	do	<-57.8	<-57.8	<1	53.24
21		LMS	46	e-1	do	-13	-19.0	<1	23.61
22		LMS	46	e-3	do	-11	-17.0	<1	21.65
23		LMS	46	e-6	do	-8.9	-14.9	<1	19.60
24		LMS	46	e-7	do	-8.5	-14.5	<1	19.2
25		LMS	46	e-8	do	-7.5	-13.5	<1	18.21
26		NLMS	6	-	do	-12.3	-18.3	<1	22.93
27		NLMS	46	-	do	-14.4	-20.4	<1	25.0

NB: $SNR = 20\log\left(S/\left(noise + echo\right)\right)$ where *S* is the signal at the output to the coupler (~1.74V_{rms}), *noise* is the noise in the power line (~130dB/Hz) and *echo* is the hybrid echo

Here, *S* is the signal strength, *Noise* is the power line noise (-130dB/Hz [8]) and *echo* is the echo caused by the hybrid. A comparatively high echo (-3.2dB) is expected if the input impedance decreases to 50 ohm which is usually the case for many devices causing signal distortion to over 70%. In the transceiver designed, additional echo cancellation is done for HY1 by implementation of EC in DSP. The echo in HY2 is dependent on the impedance of a power line which may go from 1K to a few ohms and is highly variable and unpredictable and hence not done in the work.

V. IMPLEMENTATION OF THE ECHO CANCELLER ON THE DSP

5.1. The TMS320C6713 based DSK6713

The TMS320C6713 DSP based DSK6713 Starter kit [11] is provided with onboard AIC23 stereo codec with adjustable sampling rates from 8 and 96 kHz, a 16MB synchronous dynamic RAM, 128 KB flash ROM and works at a clock frequency of 225MHz. The board is connected to a PC via a universal serial bus whence the Code Composer Studio Integrated Development Environment (CCS IDE) enables creation and compilation of program files which can then be loaded onto the DSK for implementation and executing. Adaptive echo cancellation is performed by the DSK using Least Mean Square (LMS) and Normalized Least Mean Square Algorithm (NLMS) [12].

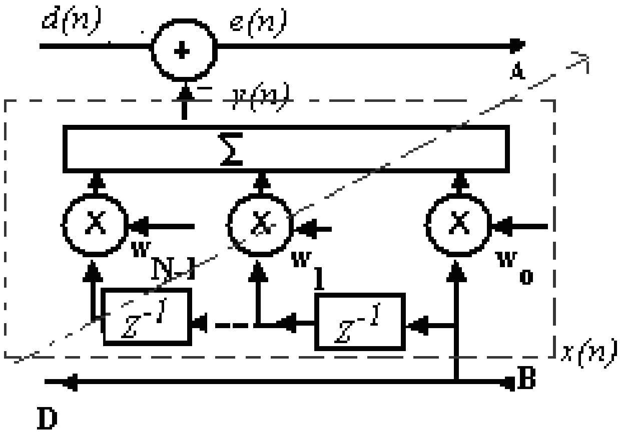

Figure 2: Adaptive filters

5.2. Echo Cancellation using Adaptive Filtration

Adaptive filter is a transversal filter (Figure 2)[13], the characteristics of which can be expressed as a vector consisting of values known as the tap weights or the co-efficient of the finite impulse filter (FIR) $w(n)$. If $x(n)$ is the input vector samples, z^{-1} is the delay of the sample period and *N* is the number of tap weights known as filter order or filter length, then the output of the adaptive filter is given by $y(n)=\sum_{i=0}^{N-1} w(n)x(n-i)$. The aim of an adaptive filter is to calculate the difference between the desired signal $d(n)$ and the adaptive filter output $y(n)$ known as the error signal $e(n)$ and to change the coefficients algorithmically aiming to minimize a function of this difference, known as the *cost* function. In the LMS algorithm, $w(n)$ are updated at every iteration according to the following formula.

$$\mathbf{w}(n+1)=\mathbf{w}(n)+2\mu e(n)\mathbf{x}(n) \tag{2}$$

The small positive constant μ known as the step function controls the influence of the updating factor. In the NLMS algorithm, μ is proportional to the inverse of the total expected energy of the instantaneous values of the coefficients of the input vector $\boldsymbol{x}(n)$ and is a variable in each iteration. The recursion formula for the NLMS algorithm is stated in the following equation.

$$\mathbf{w}(n+1)=\mathbf{w}(n)+\frac{1}{\mathbf{x}^T(n)\mathbf{x}(n)}e(n)\mathbf{x}(n) \tag{3}$$

In the work, the efficiency of both LMS and NLMS are tested in the transceiver.

5.3. Hybrid Echo Cancellation on the DSK

In the system designed, $tr(t)$ from DTE1 is passed to the input of the modulator through HY1 unchanged. A part of the received signal $x(t)$ from the demodulator (denoted by $r'(t)$) is passed on to the two wire line connected to DTE1 through HY1 and the other part is reflected as echo($r(t)$) in the transmit line. The signal $tr(t)+r(t)$ is inserted to the left channel (LC) of the CODEC of the DSK and acts as the desired signal $d(n)$ of the Section 6.2. The receive signal $x(t)$ from the receiver is applied to the right channel (RC) of the same and acts as the input to the adaptive filter implemented. The output of the digital to analog converter (DAC) of the CODEC is the difference between $tr(t)+r(t)$ and the minimized echo $y(t)$ obtained after echo cancellation and constitutes $e(t)$. Both the LMS and NLMS algorithm are written using C language with GEL files to give options to the output. During implementation, the EC is optimized for N, μ and convergence time (the time required by the error signal to reach -57.8 dB which is the noise floor of the output signal).

VI. RESULT

6.1. Analog signal communication

To test the transceiver for analog signal transmission, signal generators and audio source are taken as the DTE1 and signal transmitted through the power line.

6.1.1. Signal from signal generators

To optimize the adaptive filter for filter parameters, the transmit is constituted of a 1Vp-p sinusoidal signal of frequency 540 Hz from a signal generator acting as DTE1 and receive, a 2.5 kHz signal of the same amplitude from another signal generator acting as the output of the DEMOD. The former together with the echo of the later is connected to the LC of the DSK and the later at the RC of the same. N and μ (for LMS) are varied and the Echo Return Loss Enhancement (ERLE), convergence time and residual echo is noted for each cases. ERLE is defined by G.168 as the attenuation of the echo signal as it passes through the send path of an echo canceller. It is seen that for LMS algorithm, the convergence time decreases with an increase of step size (Sl. 3- Sl. 9 in Table 1) but failed to give any output for $\mu \geq 5$ (Sl.2 in Table 1). This is because μ must so be chosen that $0<\mu<2/\lambda_{\max}$ where

$\lambda_{\max}$ is the maximum Eigen value of the autocorrelation matrix $R = E\{x(n)^T x(n)\}$ of the input sequence and provides divergence if not obeyed [14]. The convergence time decreased with the increase of N (at fixed μ) (Sl. 11-Sl. 16 in Table 1) and failed to give any output for $N \leq 35$ (Sl.10 in Table 1) due to failure of the adaptive filter to represent the impulse response of the hybrid effectively. The NLMS showed better performance than the LMS with the inputs remaining same. For low filter lengths, it is impossible to cancel the error completely due to reasons stated earlier. However, the EC worked with low value of filter taps compared to the LMS. With this arrangement, analog signal in both the directions could be transmitted in a SOHO with a transmit power of only 0.003W and a *SNR* of 53.24dB which is high compared to 3.65 in case of absence of the EC. However, the signal in the bi-directional way should be uncorrelated otherwise the EC will not work.

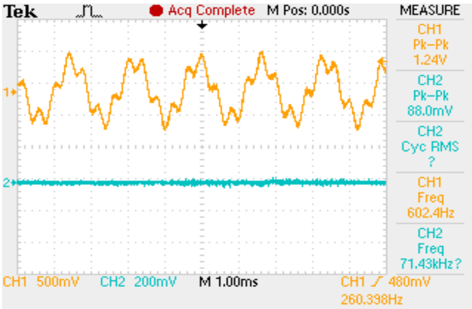

Figure 3(a)

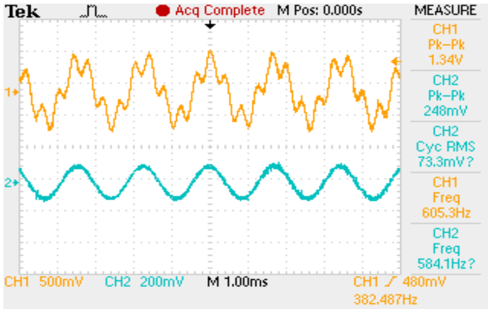

Figure 3(b)

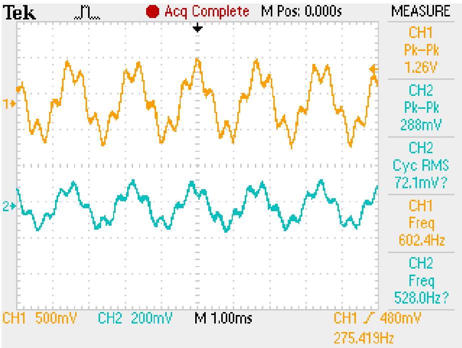

Figure 3(c)

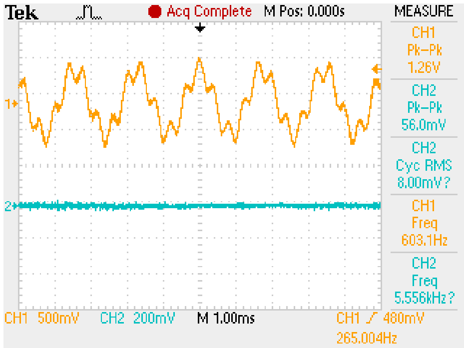

Figure 3(d)

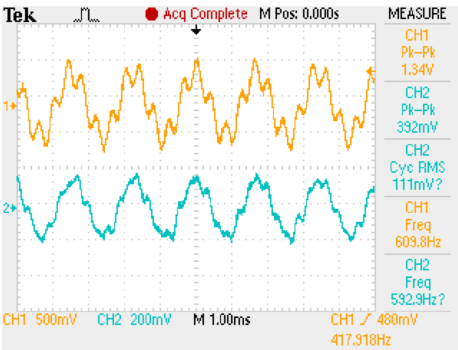

Figure 3(e)

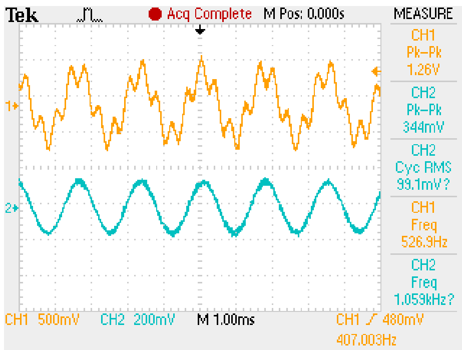

Figure 3(e)

Figure 3: $tr(t)+r(t)$ (top) and $e(t)$ (bottom) for (a) LMS , FL:46, μ=5 (Sl. 2, Table 1), (b) LMS, FL:46, μ=1.e-2 (Sl. 4, Table 1) , (c) LMS, FL:46, μ=1.e-4 (Sl. 9, Table 1)(d) LMS, FL:35, μ=3e-4(Sl. 10, Table 1)(e)NLMS, FL: 4, (Sl. 18, Table 1)(f) NLMS, FL: 46, (Sl. 20, Table 1)

6.1.2. Audio Signal

To optimize for adaptive filter parameters, audio signal from a source acting as the output of the DEMOD is applied at the RC of the DSK and the reflected echo from HY1 at the LC of the same. Here too, the residual echo is very much dependent on the μ of the LMS algorithm and hence unsuitable for the job. However with an *N* of 46 a very low echo (-20.4dB) could be obtained using the NLMS algorithm. Using this arrangement, audio signal could be transfer through the power line through a distance of 10m.The *SNR* (25.0dB) is however less than that due to orthogonal signals. However correlated speech signal in a bidirectional ways expected to reduce the performance of the transceiver due to the EC.

6.2. Data Communication

For data communication two computers equipped with 56kbps internal modems act as the DTE. The data signal from both the modems having a base band bandwidth of 4 kHz digital signal is modulated by AMMOD and transmitted to the power line. To make the implementation simpler, the adaptive is optimized as in the case of audio signal and a *N* of 46 is used. As modem signal has a flat PSD, the same requires faster convergence speed than that predicted by the audio signal which has a decreasing one. Also bidirectional modem signal are uncorrelated unlike audio signals and cancels the risk of decreasing the performance of the EC. With this arrangement, data rates of 33.3kbps were obtained although through a small distance of 10m.

VII. CONCLUSION

In this work, a transceiver has been designed to enable analog/ data transmission through an indoor power line setting in a SOHO. Higher power would be required for greater distances and would require modifying of the hybrids to decrease the echo in the receive line. Keeping in mind that the power line noise is unpredictable; a multi-channel frequency hopping technique would be a good option for adaptive signal transfer. The second hybrid can also be made more robust over power line channel impedance by using adaptive impedance matching in power line [15]. However, use of the transceiver in a practical SOHO setting requires further study of the S/N of the signal transmitted and the error caused by the same. Further study is expected to lead to the development of plug in systems that can be simply connected to the channel for data monitoring, transfer and control.

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