A NOVEL METHOD FOR POWER LINE INTERFERENCE SUPPRESSION

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ABSTRACT

In this paper, a novel method to suppress the power line interference induced into telephone lines that are in proximity to power conductors is proposed. A phase-locked loop is used to synchronize the interference signal with an anti-phase waveform, which is stored in a buffer. The anti-phase signal is injected on the line and the samples of the residual signal are used to update the anti-phase waveform. Computer simulations are used to compare the novel Adaptive Phase-Locked Buffer (APLB) approach with the traditional Least Mean Squares (LMS) algorithm in an adaptive noise cancellation configuration. The new technique achieves 15 dB further suppression compared to LMS and since it can be implemented on a single Digital Signal Processor (DSP) chip it proves to be a very efficient solution to the power line interference problem.

1. INTRODUCTION

The problem of power line interference is well known to the power and telecommunication industries [1–9]. The advent of new, fast and efficient electrified railroads has caused a significant increase of the 50/60 Hz induced voltages on both wires of the nearby telephony cables. This is called common-mode (CM) interference, and an actual signature is shown in Fig. 1 (50 Hz).

When the telephone pair is well balanced, it has a good common-mode rejection ratio and in such cases very little CM energy is transformed to differential-mode (DM) energy. However, in practice some lines are not well balanced, resulting in significant mains "hum" which is audible on a telephone handset (Fig. 2). The harmonics of this interference generally fall within the voice-frequency bandwidth (300–3400 Hz) of the telecommunication lines, degrading the quality of speech and data transmission and causing signaling problems [9].

The main problem is that the DM interference signal is superimposed on any speech or data signals that are present on the line. Any conventional attempt to filter out its frequency content would degrade telephony operation.
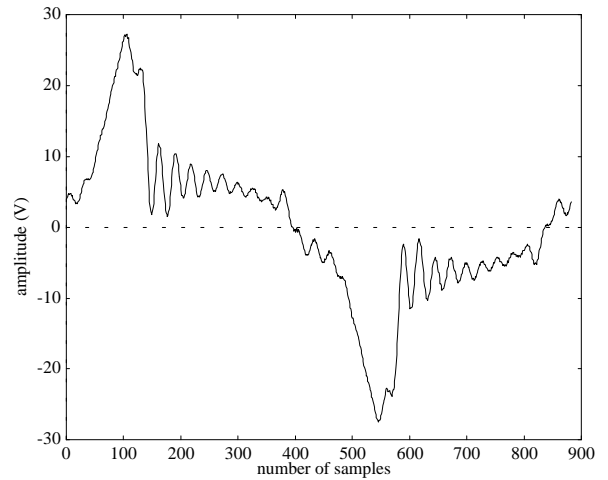

Figure 1. Actual CM signature recorded in Newcastle (UK) in October 1997. The sampling frequency was 44.1 kHz.
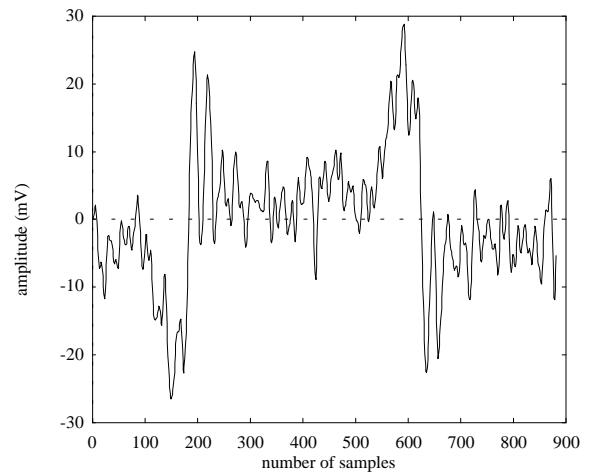

Figure 2. Corresponding DM signature recorded in Newcastle (UK), after handset filtering. The sampling frequency was 44.1 kHz.

Digital transmission systems may be employed as a solution to the problem but they have the disadvantages of being reach limited and expensive [10]. The necessity for a stand-alone noise canceller can thus be identified, to provide a more cost-effective solution to the interference problem.

Adaptive noise cancellation techniques have been widely employed in similar situations [11]. The configuration is shown in Fig. 3. An information signal S is transmitted over the line and the interference signal DM is superimposed on it. The Adaptive Noise Cancellation Unit (ANCU) receives the CM signal (reference signal) and the residual signal DE and produces an output that is as close as possible to the DM signal. This output is subtracted from the input S + DM to suppress the interference signal DM.
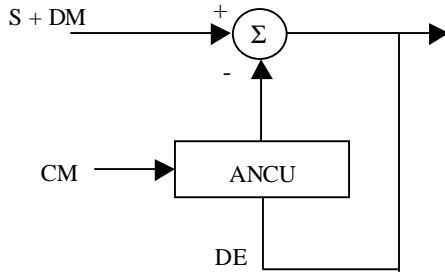

Figure 3. The adaptive noise canceling concept [11].

The particular characteristics of the case under consideration impose tight constraints on the possible adaptive algorithms that might be used, since there is a need for a real-time, robust and computationally efficient approach. The LMS algorithm has been successfully employed in similar cases, achieving significant interference suppression [12]. In this contribution, a novel adaptive system, the Adaptive Phase-Locked Buffer (APLB) is proposed, which outperforms the LMS approach. In Section 2, a complete description of the APLB approach is given. In Section 3, computer simulations are used to compare the performances of the two techniques. Finally, the conclusions are given in Section 4.

2. THE APLB APPROACH

In the APLB approach, the periodicity of the interference signal (50/60 Hz) is exploited. The basic idea is to store one period of the interference signal into a buffer, synchronize the buffer waveform with the incoming waveform and then subtract the replica from the noise signal [10]. To allow for the changes of the DM signal

from cycle to cycle, the replica of the interference is adaptively updated using the samples of the residual signal (DE). A detailed description of the basic stages of processing in the APLB technique follows.

2.1 Buffer updating

Suppose that complete synchronization has been achieved between the incoming DM signal and its replica, which is stored in a buffer. Then each tap in the buffer corresponds to a fixed location of the noise waveform. Let $w_m(n)$ be the value of the tap m at period n and M be the total number of taps used to store one period of the noise signal. As each tap is injected on the line, its effect on the error signal is used to modify it in the appropriate direction. Let $DE_m(n)$ be the value of the error signal which corresponds to the tap m at period n. During one noise period, all the taps in the buffer are updated once according to the following equation:

$$w_m(n+1) = w_m(n) + g \cdot DE_m(n) \quad (1)$$

The update gain g controls the speed of adaptation. During one noise period, eq. (1) is used for m = 1 to M, resulting in an update of one cycle of the noise replica.

2.2 Synchronization

The operation of the APLB technique critically depends on the synchronization between the incoming DM signal and its replica, which is stored in a buffer. The modeling algorithm relies on adapting the taps stored in the buffer such that the buffer waveform corresponds to one cycle of the fundamental frequency f_n (50/60 Hz) of the interference. In order to update all the taps of the buffer during one noise period, the updating rate must be $M \cdot f_n$ where M is the number of taps. It is therefore necessary to lock onto the noise repetition frequency f_n and update the taps at this rate. This can be achieved by controlling the number of CM arrivals between successive tap updates using a Digital Phase-Locked Loop (DPLL) [13, 14].

A diagram of the DPLL is shown in Fig. 4. A Phase Detector (PD) decides whether the phase of the noise replica needs to be advanced or retarded. Its output is then sent through a Loop Filter (LF) and then through a Delta Sigma Modulator (DSM), which has an integer output, determining the number of CM arrivals before the update of the next tap.
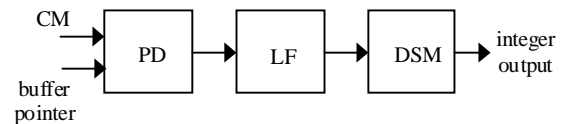

Figure 4. Digital Phase-Locked Loop [13, 14].

2.3 Gain control

A comb filter is used to measure the residual signal energy (DE) that is not related to the noise spectrum, i.e., speech or data. When this energy exceeds a certain threshold, the update gain is set to minimum to prevent false adaptation from taking place.

2.4 Computational complexity

The main processing of the APLB approach consists of two parts: the updating part and the DPLL part. As it can be seen from eq. (1), the updating part involves only 1 multiplication and 1 addition per tap and the DPLL part can be implemented with 3 multiplications, 5 additions and a few logical operations. This is a very small amount of processing compared with the (2M+1) multiplications and 2M additions per sample required for the implementation of an LMS filter with M taps.

3. COMPUTER SIMULATIONS

In this section results from computer simulations are presented. The APLB approach was applied on the Adaptive Noise Cancellation (ANC) setup shown in Fig. 3. Actual recordings of the CM and DM signals were used in these simulations. 320 taps were used to store one period of the interference signal. In each period, the Mean Square Error (MSE) of the residual signal was calculated and expressed in dB. Fig. 5 shows the performance of the APLB approach. The average interference suppression is 25 dB and this is achieved within 100 periods (2 seconds in the case of the 50 Hz interference). For comparison purposes, the LMS algorithm was also implemented. Its performance is shown in the same figure for 2 different filter lengths. The curve shows that the noise suppression achieved with the APLB technique is 15 dB higher than the one corresponding to an LMS filter with 800 taps. The performance of the LMS filter can be improved by using more taps, but this increases significantly the convergence time. The APLB method has excellent performance, combining very good interference suppression and fast convergence at the same time.

4. CONCLUSIONS

A novel method for power line interference suppression on telephone lines has been proposed in this paper. The Adaptive Phase-Locked Buffer (APLB) technique exploits the near-periodic nature of the interference in order to achieve noise suppression. Computer simulations have indicated that the APLB approach has superior performance when compared with the traditional LMS algorithm. In addition to this, its computational simplicity allows it to be implemented on a single DSP chip,

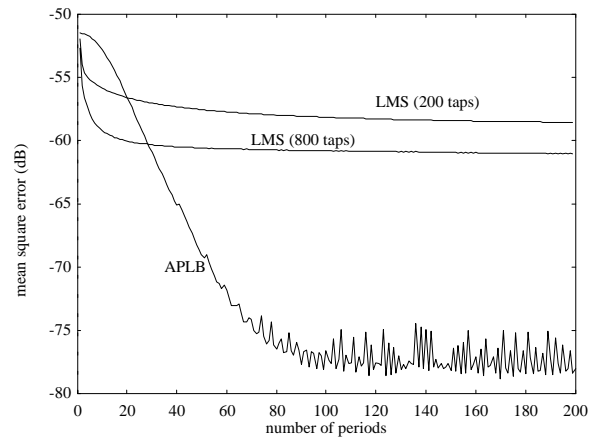providing a very efficient solution to the power line interference problem.

Figure 5. Comparison between the APLB and the LMS approaches

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