FAST COUPLED ADAPTATION FOR SPARSE CHANNELS USING A PARTIAL HAAR TRANSFORM

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ABSTRACT

This paper presents a novel scheme for identifying the impulse response of a sparse channel. The scheme consists of two adaptive filters operating sequentially. The first adaptive filter adapts using a partial Haar transform of the input and yields an estimate of the location of the peak of the sparse impulse response. The second adaptive filter is then centered about this estimate. Both filters are short in comparison to the delay uncertainty of the unknown channel. The principle advantage of this scheme is that two short adaptive filters can be used instead of one long adaptive filter, resulting in faster overall convergence and reduced computational complexity and storage.

I. INTRODUCTION

Echo Cancellers have been used in networks for voice quality enhancements for several years. The Network or Hybrid Echo on the Public Switched Telephone Network (PSTN) is caused by the four wire to two wire impedance mismatch. This mismatch results in unwanted reflection of transmitted energy back to the speaker or the source. Networks are equipped with Echo Cancellers, known as Network or Line Echo Cancellers, to remove these unwanted reflections. The International Telecommunication Union's (ITU) Recommendation ITU-T G.168 2002 [1] specifies the minimum requirements and test conditions for performance of Network Echo Cancellers in the PSTN. There are two main design problems: 1) choice of adaptation algorithm(s), and 2) control logic for adaptation. The latter is caused by double-talk. Basically, the algorithm weights should be frozen in the presence of double-talk and adapt quickly in the absence of double-talk. The control logic can be quite complicated [2,3] since it often not easy to discriminate between the far-end double- is talk and the near end-speaker.

One of the special characteristics of the echo channel is that of sparseness. The bulk delay of the channel is often much longer than the channel impulse response. Typically, the bulk delay can be on the order of 128 milliseconds whereas the actual impulse response can be as short as 4 milliseconds. Most real network hybrids have impulse responses in the range of 4-20 msec. ITU-T G.168 recommendation [1] describes a set of 7 hybrid impulse responses that model most hybrids used in real networks. Thus, most of the adaptive tap weights in a 128 millisecond delay line (1024 taps for an 8 kh sampling rate) will be zero. Now, long adaptive filters are both slow to adapt (speed is inversely proportional to the number of taps) and have noisy weights (weight noise is proportional to the number of taps). Thus, much effort has been expended towards reduction of the adaptive filter length while still being able to adapt to a delay uncertainty of 128 milliseconds. The following advantages accrue by reducing the adaptive filter length: (a) faster convergence and less residual tap noise, (b) a significant reduction in the required computations, and (c) a significant reduction in the overall state memory required for storing the coefficients.

Reference [4] first proposed using a separate adaptive filter for estimating the location of the peak of the channel impulse responses. The idea was to subsample band-limited versions of the adaptive filter input and desired signal, and then use an echo canceller (adaptive filter) to yield a delay estimate. With M denoting the decimation factor, the subsampled echo canceller has only 1/M as many taps as a full canceller might have. A second short adaptive filter is then centered about the peak delay estimate to obtain the actual echo cancellation. The bandlimiting has several disadvantages: a) processing delay, b) smearing of the peak of the unknown channel impulse response, and c) exclusion of many frequencies in the voice spectrum.

More recently, there has been significant interest in adapting in a transform domain using wavelets [5]. An especially useful wavelet transform for sparse channels has been the Haar transform. Reference [6] showed that much fewer taps are needed for the Haar transform based adaptive filter than for an FIR adaptive filter adapting in the time domain. Most recently [7] has proposed a two step procedure for a Haar-Basis adaptive echo cancellation algorithm. The first step involves adapting a subset of the Haar basis vectors which span the entire time range of the unknown impulse response. These coefficients are initially adapted and then used to subsequently identify the rest of the Haar coefficients needed to model the unknown channel. The hierarchal structure of the Haar transform allows this procedure to be performed very efficiently. The second step is to only adapt these coefficients. The primary difference between [6] and [7] is that [7] adaptively determines which coefficients to adapt whereas [6] needs to know a priori which coefficients to adapt. There are several drawbacks to this scheme: a) some time is required to identify which Haar coefficients to continue adapting, b) the possibility of freezing the wrong Haar coefficients, c) the number of resulting adaptive taps is random d) the canceller cannot converge to zero error if the wrong taps are initially frozen.

This paper proposes a solution to the sparse channel echo cancellation problem which combines the favorable characteristics of [4] and [7]. The scheme involves a partial Haar transform of the input and two short adaptive filters as shown in Figure 1. The partial Haar transform uses an appropriate fixed subset of the Haar basis vectors. The dimensionality of the subset determines the size (number of taps) of the upper adaptive filter. Thus, the upper adaptive filter is unable to exactly model any impulse response. However, this is not required since the upper adaptive filter is trying only to estimate the delay of the channel impulse response peak. The lower short adaptive filter is centered in time about the peak. The error signal e(n) of the lower adaptive filter can converge to zero in the absence of doubletalk and background noise.

For a 128 millisecond delay uncertainty (1024 taps for a single adaptive filter), the number of basis functions considered here for the partial Haar transform (and the number of taps for the upper filter) are 256 or 128 or 64. The lower filter has 128 taps to match the longest expected impulse response. Thus, the scheme offers a reduction in the total number of adaptive filter weights on the order of four or five. There is one disadvantage to using fewer taps in the upper adaptive filter. The partial Haar transform of the input z(n) causes some smearing of the system identification signal x(n). A specific example is presented in Section III which shows the typical losses incurred using 256, 128 or 64 Haar basis functions.

II. HAAR TRANSFORM PROPERTIES

A. The Haar Transform

The Haar transform is based on the theory of wavelets [8] The Haar wavelets are discrete-time orthonormal sequences [6 , 9] defined by the relations

$$\psi_{mn}(t) = \psi_{m0}(t - 2^m n) \qquad (1)$$

where

$$\begin{aligned} \psi_{m0}(t) &= 2^{-m/2}, \quad \mathit{for} \quad 0 \le t \le 2^{m-1} - 1, \\ &= -2^{-m/2}, \quad \mathit{for} \quad 2^{m-1} \le t \le 2^m - 1, \qquad (2) \\ &= 0 \quad \mathit{otherwise} \end{aligned}$$

The indices *m* and *n* correspond to the scale and translation respectively. *m* is a natural number assumed to go from 1 to M and *n* is an integer. These conditions imply that the Haar transform can be represented by an N x N orthogonal matrix H_M where $N = 2^M$ and $H_M^{-1} = H_M^T$.

B. Effect of the Haar Transform on the Wiener Solution

It is well-known that the Wiener filter W_O for the minimization of the Mean-Square Error (MSE) is $W_O = R_{XX}^{-1} R_{dX}$ where X(n) = [x(n) x(n-1) ......x(n-N-1)]^T and d(n) is the desired signal. It is easy to show that the Wiener solution W_{HO} for the Haar transform is given by $W_{HO} = H_M W_O$. Thus, the Wiener solution for the Haar transformed input is the Haar transform of the original Wiener solution.

III. THE PARTIAL HAAR TRANSFORM

A. Choice of the Partial Haar Transform

The choice of the subset of the basis vectors of the Haar transform for a partial Haar transform depends upon the number of adaptive coefficients desired for the upper filter. The subset needs to span the time axis to locate the of unknown channel impulse response peak.

The data block length is 1024 in the real world case. Thus, M = $\log_2(1024)$ = 10. In this case, the four shortest basic pulses in the H_{10} matrix are

$1/\sqrt{2}$ x [1 -1] , $1/\sqrt{4}$ x [1 1 -1 -1]
$1/\sqrt{8}$ x [1 1 1 1 -1 -1 -1 -1] ,
$1/\sqrt{16}$ x [1 1 1 1 1 1 1 1 -1 -1 -1 -1 -1 -1 -1 -1] for 512, 256, 128 and 64 adaptive weights, respectively. These vectors correspond to $\psi_{10}(t), \psi_{20}(t), \psi_{30}(t)$ *and* $\psi_{40}(t)$ in equation (2). 512 adaptive weights will be discarded because this choice does not yield much of a complexity or computational reduction. Thus, the chosen partial Haar transform corresponds to three of the four shortest pulses in the H_{10} matrix.

B. Effects of Partial Haar Transform on the Wiener Solution

Let H_{Mp} denote the partial Haar transform. H_{Mp} is not a square matrix. For M =10, $H_{10\,p}$ is p x 1024 for p=256, 128 or 64 weights. When $R_{XX} = \sigma_x^2 I_N$, [10] shows $W_{HpO} = H_{Mp} W_O$. The partial Haar Wiener filter is the partial Haar transform of the optimum filter for the un-transformed input.

C. The Effect of the Partial Haar Transform on W_O

The purpose of this subsection is to demonstrate how the partial Haar transform changes the system identification problem for W_O into that for W_{HpO} and the possible losses incurred. Consider a symmetric exponential channel impulse response of the form

$W_O = [a^r \;\; a^{r-1} ....a \quad 1 \quad a............a^{r-1} \;\; a^r]^T$.

Table I shows the significant partial Haar coefficients of W_{HpO}. Table I was generated as the channel bulk delay was varied from zero to 8 taps for r =32 and a = .5. The variable bulk delay represents the random delay of the channel with respect to the 1024 tap adaptive filter. The best case (worst case) peak amplitude degradation is .7126 for 128 taps (.2419 for 64 taps). Although the worst case amplitude loss is significant, it is not catastrophic for locating the peak of the channel impulse response as is shown below.

IV. MONTE CARLO SIMULATIONS AND COMPARISON WITH THEORY

An analytical model for the stochastic behavior of the Partial Haar Dual Adaptive filter has been developed in [10] for LMS-LMS algorithms. This theoretical model cannot be presented here for reasons of space. Suffice to say that there was very good agreement between the theoretical predictions of the weight behavior and Monte Carlo simulations for both filters.

A. Mean Time to Accurately Estimate Delay

Monte Carlo simulations (200 for each case) were run to obtain the mean and standard deviation of the number of iterations of the upper adaptive filter for correct estimates by the peak delay estimator ($\mu = .1/(q+2)$ and $\sigma_x^2 = 1$). The results are shown in Table II.

B. Lower LMS Adaptive Filter.

Tapped delay-line signal model Monte Carlo simulations for the MSE (10 for each case with smoothing with a uniform time average of 100 samples) with zero-mean unit variance Gaussian white input data samples are compared with the theoretical predictions in Figure 2. The various parameters are q=256 ($W_{HpO\,\max}$=.375), q=128 ($W_{HpO\,\max}$=.7126) and q=64 ($W_{HpO\,\max}$=.249) and $\mu = .1/(q+2)$ and $\sigma_x^2 = 1$. The second and third cases correspond to the largest and smallest maximum of W_{HpO}. The theoretical behavior was obtained using [10] for the maximum of W_{HpO} equal to unity and q =128. Note there is little difference in MSE performance between the three MC simulations and the theoretical curve. The only difference in the MC simulations is due to the time required to correctly estimate the location of the impulse response peak using the upper adaptive filter. Table I indicates that, at worst, less than 200 iterations are needed. Thus, a total of 1024 + 200 = 1224 data samples are required to correctly estimate the location of the peak. Hence, the upper adaptive filter requires about 15 milliseconds to converge.

V. CONCLUSIONS

This paper has presented a novel scheme for identifying the impulse response of a sparse channel. The scheme consisted of two adaptive filters operating sequentially. The first adaptive filter adapts using a partial Haar transform of the input and yields an estimate of the location of the peak of the sparse impulse response. The second adaptive filter is then centered about this estimate. Both filters are short in comparison to the delay uncertainty of the unknown channel. The principle advantage of this scheme is that two short adaptive filters can be used instead of one long adaptive filter, resulting in faster overall convergence and reduced computational complexity and storage.

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256 tap partial Haar transform W_{HpO}

[-0.0001 -0.0011 -0.0176 -0.2812 0.5625 0.0352 0.0022 0.0001]
[0.0005 -0.0088 -0.1406 0.3750 0.0703 0.0044 0.0003]
[-0.0003 -0.0044 -0.0703 -0.3750 0.1406 0.0088 0.0005]
[-0.0001 -0.0022 -0.0352 -0.5625 0.2812 0.0176 0.0011 0.0001]

128 tap partial Haar transform W_{HpO}

[-0.0012 -0.3107 0.6215 0.0024]
[-0.0006 -0.1554 0.7126 0.0049]
[-0.0003 -0.0777 0.6298 0.0097]
[-0.0002 -0.0388 0.3315 0.0194 0.0001]
[-0.0001 -0.0194 -0.3315 0.0388 0.0002]
[-0.0097 -0.6298 0.0777 0.0003]
[-0.0049 -0.7126 0.1554 0.0006]
[-0.0024 -0.6215 0.3107 0.0012]

64 tap partial Haar transform W_{HpO}

[-0.2481 0.4961]
[-0.1240 0.6172]
[-0.0620 0.6719]
[-0.0310 0.6876 0.0001]
[-0.0155 0.6720 0.0001]
[-0.0078 0.6174 0.0002]
[-0.0039 0.4966 0.0005]
[-0.0019 0.2490 0.0010]

Table I - Significant coefficients of the partial Haar Transform of a two-sided exponential channel for r = 32 and a = .5 for different bulk delays.

			DELAY LINE		INDEPENDENT SIGNAL	
q	W_{HpO} max	$J_{\min}$	mean	std. dev.	mean	std. dev.
256	.375	1.5016	88.5	27.0	122.0	66.
256	.5625	1.26	44.1	19.9	64.2	34.4
128	3315	1.5553	140.0	64.9	124.7	67.9
128	.6215	1.184	75.9	31.9	41.9	22.1
128	.7126	1.1347	62.0	24.1	38.0	19.3
64	.249	1.6047	176	117	149	79.0
64	.4961	1.36	147	81.5	58.6	34.6
64	.6172	1.27	111	60.7	41.5	25.7
64	.6876	1.193	99.9	52.2	33.8	17.4

Table II. Mean and standard deviation of the number of iterations of the upper adaptive filter for correct estimates by the peak delay estimator.

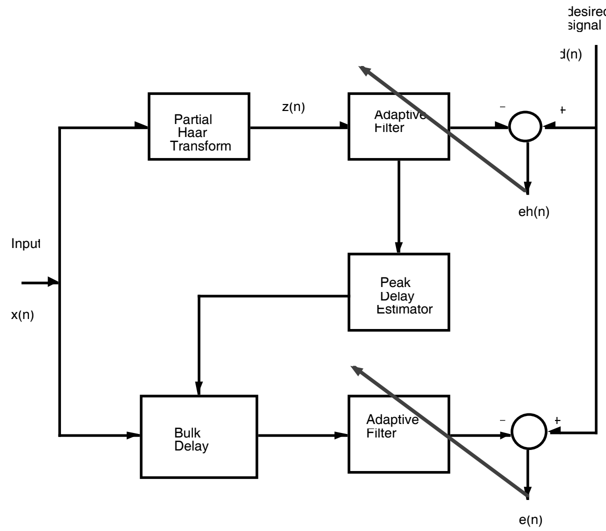

Figure 1. Partial Haar - Dual Adaptive Filter for Sparse Channels

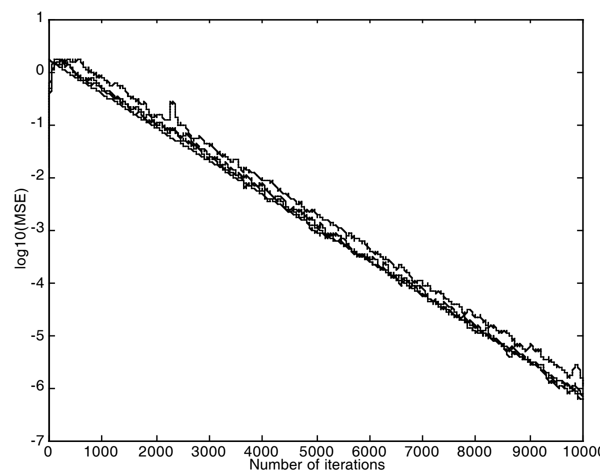

Figure 2 $\log_{10}$(MSE) vs. number of iterations for the lower adaptive filter for four cases:
i) theory (bottom straight line),
ii) partial Haar 64 (.249) - top curve,
iii) partial Haar 128 (.7126) , and
iv) partial Haar 256 (.375) (middle curves).