Sub-Band IIR Acoustic Echo Canceller

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***Abstract* - In teleconferencing systems the echo can be very long. A direct FIR implementation of the echo canceller often requires more than 2000 taps. One approach is to process the signal in sub-bands providing a separate FIR echo canceller for each band.**

In this paper we propose the use of an IIR filter in each sub-band adapted by a variant of the Steiglitz-McBride identification scheme instead of the classical adaptive FIR filters. Experimental results and computer simulations show that the proposed IIR sub-band echo canceller performs better then the equivalent FIR counterpart.

I. INTRODUCTION

The basic structure of a sub-band acoustic echo canceller is shown in Fig.1.

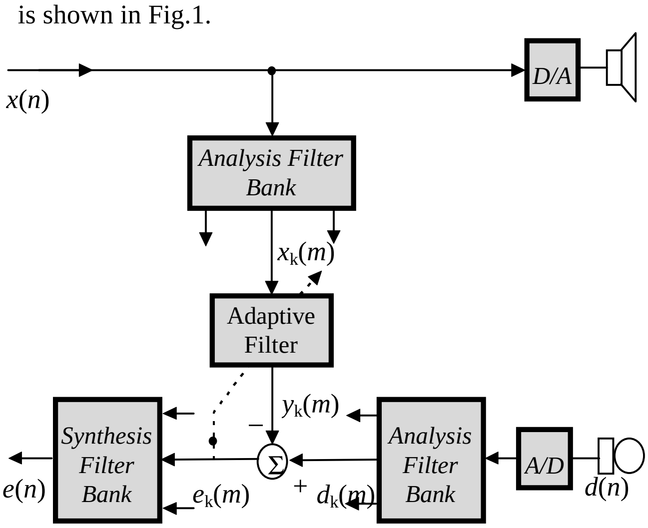

Fig.1 Sub-band echo canceller

The signal $x(n)$ is the received speech from the far-end talker and $d(n)$ is the echo produced by the coupling between the loudspeaker and microphone: both signals are decomposed in sub-bands by analysis QMF (quadrature mirror filter) [4] banks. In this way, the number of taps and the coefficient adaptation rate, required for each band, can be reduced due to the decimation.

Hence, as shown in [2], the total computational requirement decreases with the number of sub-bands. Moreover, the adaptation speed can be modified in each sub-band.

The classical sub-band acoustic echo cancellers use an FIR adaptive filter for each band [2]. Instead, we propose the use of an adaptive IIR filter for each band. The primary advantages of an IIR filter is that generates an infinite impulse response with only a finite number of parameters. In this way, a desired long response as the echo path can be more effectively matched by the output of such filters. Experimental results and computer simulations show, in fact, that the proposed IIR split-band echo canceller performs better then the equivalent FIR counterpart.

II. THE PROPOSED IIR SUB-BAND ECHO CANCELLER

The classical FIR sub-band echo canceller uses the normalized LMS algorithm to adapt each filter shown in Fig.1. The basic idea of our structure is to use this algorithm to separately adapt the IIR filter of each band. Fundamentally, there are two approaches to adaptive IIR filtering corresponding to the formulation of the error criterion used for minimization. These are the *output error* method and the *equation error* method [1]. The method of minimization used is the LMS algorithm which is an approximation of the stochastic gradient method. The *output error* adaptive *IIR* filter is characterized by the following recursive difference equation:

$$y(n)=\sum_{m=1}^{N}\hat{a}_m(n)y(n-m)+\sum_{m=0}^{M}\hat{b}_m(n)x(n-m) \qquad (1)$$

where $\hat{a}_m(n)$ and $\hat{b}_m(n)$ are the coefficients of the denominator and numerator polynomials, respectively.

Then the *output error*

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

which can be written in vector form as

$$e(n)=d(n)-\ \mathbf{q}(n)\,\mathbf{f}(n) \quad (2)$$

where

$$\mathbf{q}(n)=\left[\hat{a}_1(n) \quad \dots \quad \hat{a}_N(n) \quad \hat{b}_0(n) \quad \dots \quad \hat{b}_M(n)\right]^T$$

and

$$\mathbf{f}(n)=\left[y(n-1) \quad \dots \quad y(n-N) \quad x(n) \quad \dots \quad x(n-M)\right]^T$$

Using the LMS algorithm, the updating equation becomes

$$\mathbf{q}(n+1)=\mathbf{q}(n)-\alpha\hat{\nabla}(n) \qquad (3)$$

where $\hat{\nabla}(n)$ is the estimate of the gradient vector given by

$$\hat{\nabla}(n)=-2\left[\frac{\partial y(n)}{\partial \hat{a}_1(n)} \quad \dots \quad \frac{\partial y(n)}{\partial \hat{a}_N(n)} \quad \frac{\partial y(n)}{\partial \hat{b}_0(n)} \quad \dots \quad \frac{\partial y(n)}{\partial \hat{b}_M(n)}\right]^T e(n)=$$

$$=-2\left[\alpha_1(n) \quad \dots \quad \alpha_N(n) \quad \beta_0(n) \quad \dots \quad \beta_M(n)\right]^T e(n)$$

where $\alpha_i(n)$, $\beta_i(n)$ can be calculated recursively using:

$$\alpha_i(n) = y(n-i) + \sum_{m=1}^{N} \hat{a}_m(n)\alpha_i(n-m) \quad 1 \le i \le N$$
$$\beta_i(n) = x(n-i) + \sum_{m=1}^{N} \hat{a}_m(n)\beta_i(n-m) \quad 0 \le i \le M \tag{4}$$

The recursive form of the equations in (4) is the result of the recursive nature $y(n)$. From (1) it is clear that the error $e(n)$ is nonlinear function of the filter coefficients, resulting in an error surface which is nonquadratic function of the filter coefficients. This property can result in convergence to a local minimum. The other problem with IIR filters is that the adaptive algorithm can become instable if the poles of the filter move outside the unit circle during adaptation. The *equation error* approach avoids stability and convergence problems of the *output error* method just described.

The *equation error* $e_e(n)$, where the subscript e is used to distinguish this output from that of the *output error* formulation, is:

$$e_e(n) = d(n) - \sum_{m=1}^{N} \hat{a}_m(n)d(n-m) - \sum_{m=0}^{M} \hat{b}_m(n)x(n-m) \tag{5}$$

The parameter $\hat{a}_m(n)$, $\hat{b}_m(n)$ are updated minimizing the *equation error* criteria using the well known LMS algorithm.

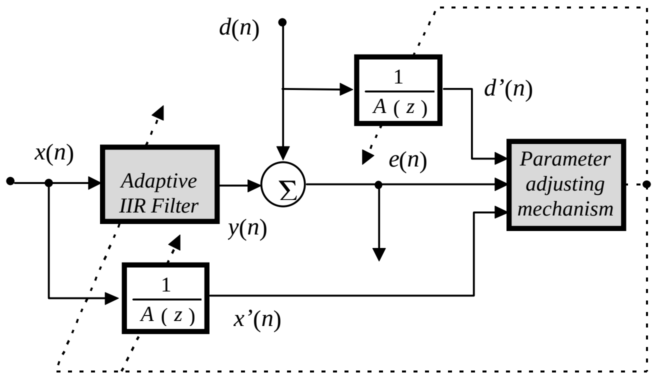

Fig.2. Diagram for *AFM* algorithm

$$A(z) = 1 - \sum_m \hat{a}_m(n)z^{-1}$$

From (5) it is noted that the *equation error* $e_e(n)$ is a linear function of the coefficients, which results in global convergence. Also, $e_e(n)$ does not contain feedbacks, therefore there is no stability problem. Although *equation error* approach for adaptive IIR filters exibits fast convergence behaviour, its performance may be completely unsatisfactory if the bias is significant. In [3] a variant of the Steiglitz-McBride identification scheme (AFM algorithm [7]) is used to design a full band adaptive IIR echo canceller which is closely related to the *equation error* approach just described.

The semplicity and the proved convergence make it attractive for this application. The AFM algorithm is shown in Fig.2.

In this method, the input and the output are pre-filtered before being used in the adaptive algorithm. For each adaptive filter the adaptation coefficients $\hat{a}_m(n)$ e $\hat{b}_m(n)$ (denominator and numerator polynomial, respectively) are modified as in Table 1.

AFM algorithm (Table 1)
Inizialization: $\hat{a}_m(0) = 0$, $\hat{b}_m(0) = 0$
For each new input $x(n)$, $d(n)$; $n \ge 0$
$\hat{a}_m(n+1) = \hat{a}_m(n) + \tau\, e(n)d'(n-m) \quad m = 1,2,..,N$
$\hat{b}_m(n+1) = \hat{b}_m(n) + \tau\, e(n)x'(n-m) \quad m = 0,1,..,M$
$d'(n) = d(n) + \sum_{m=1}^{N} \hat{a}_m(n)d'(n-m)$
$x'(n) = x(n) + \sum_{m=1}^{N} \hat{a}_m(n)x'(n-m)$
$e(n) = d(n) - y(n)$
$y(n) = \sum_{m=1}^{N} \hat{a}_m(n)y(n-m) + \sum_{m=0}^{M} \hat{b}_m(n)x(n-m)$

where M and N are respectively the order of numerator and denominator polynomials of the IIR filter.

Three kinds of echo cancellers have been implemented and tested in this paper: the first, proposed by us and named *AFMBNK*, uses in each sub-band a IIR filter with the AFM adaptation algorithm; the second, named *FIRBNK*, uses in each sub-band a FIR filter with the normalized LMS adaptation algorithm as in [2], and the third is a full band IIR echo canceller with the AFM algorithm [7], here called only AFM. The QMF bank is designed using type 32D filters [4], both for analysis and synthesis.

III. SIMULATION RESULTS

Our experiments use two different echo path models $H_1(z)$ and $H_2(z)$. The first is built considering that in a teleconference system, a typical echo path trasfer function has poles very near to the unit circle and hence an amplitude spectrum characterized by peaks and valleys. A good example of this, which we used for the experiments, is the echo path reported in [5].

The second echo path shown in Fig.3, is modelled by the well-known image method for simulating small rectangular rooms as described in [6]. We use a room dimensions of 3mx4.5mx3.8m using 2048 points.

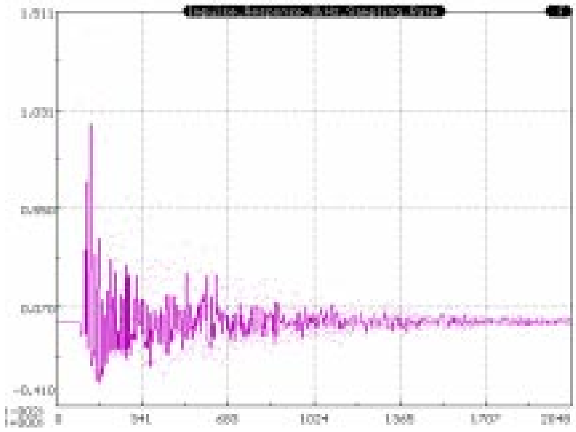

Fig.3 Impulse Response for a room
$3x4.5x3.8m^3$

In the first set of experiments we use the echo path $H_1(z)$ driven by uniform white noise with zero mean and upper and lower bounds of +1,-1 respectively . The proposed sub-band echo canceller (*AFMBNK*) uses two IIR adaptive filters, with a decimation factor of $M=2$ (sub-sampling at 4 kHz in each sub-band). Since the path model has eight zeros and ten poles [5], we use, in each sub-band, an IIR adaptive filter with 4 zeros and 5 poles with an adaptation gain $\tau=0.001$. The results are reported in terms of *ERLE,* defined as $10\mathrm{LOG}_{10}\{E[e^2(n)]/E[d^2(n)]\}$ where $E[.]$ indicates the expectation operator, vs. the number of input samples. Fig.4 shows the performance obtained by the *AFMBNK* on an average of 20 independent computer runs: as we can see, a value of about -20 dB is achieved. In order to compare this result with the *ERLE* obtained in the same experimental conditions by the *AFM* algorithm [3,7] in full band, Fig.5 reports, in the same plot, the *ERLE* performances of both algorithms.

It can be seen that the final values are very similar for both algorithms (-20dB for *AFMBNK*, -22dB for *AFM*), but a slower convergence rate is exibited by the *AFMBNK*. However, the proposed algorithm requires a smaller computational burden than the classical *AFM*, due to the reduced filters order and the decimation effect.

In the second set of experiments we compare the performance of the *AFMBNK* echo canceller with the *FIR* sub-band echo canceller (*FIRBNK*) using the *normalized LMS* algorithm. First we use the transfer function $H_1(z)$ as an echo path driven by the same input noise applied above. In each sub-band the *FIRBNK* structure has an adaptive filter of 32 taps with adaptation rate $\mu=0.001$. Fig.6 reports the *ERLE* obtained by the *FIRBNK* and that obtained by the *AFMBNK*. Here an *ERLE* of -12.5dB is achieved by the *FIRBNK* against the -20dB performance of the *AFMBNK*. We expect this result because the echo canceller, that uses an *IIR* adaptive filter in each sub-band, can better match poles as well as zeros of the echo model, whereas an *FIR* adaptive filter may give only rough approximations to them.

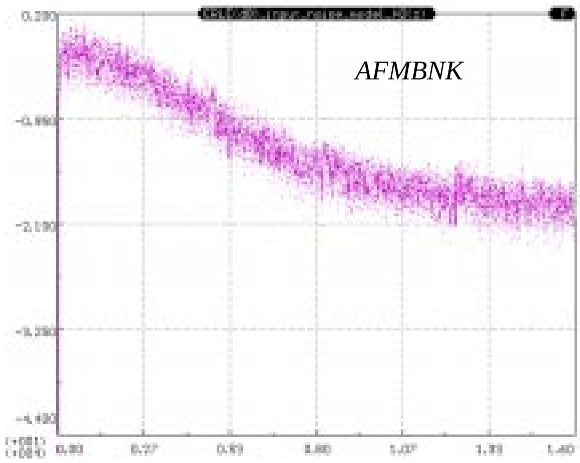

Fig.4. *ERLE (dB)* vs. input samples with echo path $H_1(z)$, obtained with *AFMBNK*

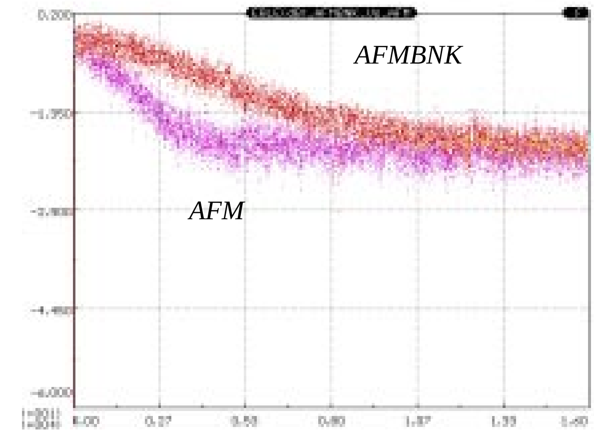

Fig.5. Comparison of the *ERLE*'s obtained with *AFMBNK* and the *AFM* full-band echo canceller.

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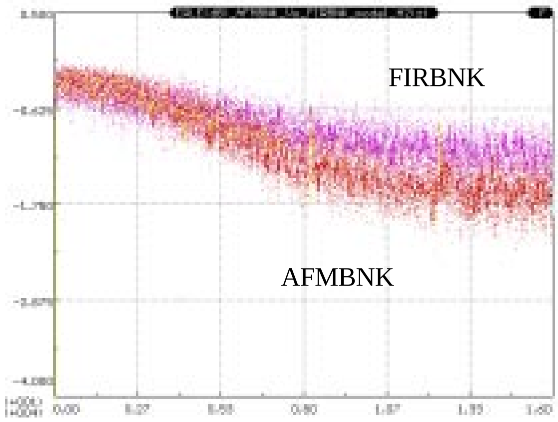

Fig.6. Comparison of the *ERLE*'s obtained by the *AFMBNK* and the *FIRBNK* echo cancellers.The echo path is $H_1(z)$.

Next the echo path model $H_2(z)$ with a 2048 taps response is used. In this case, the *AFMBNK* has 256 poles and 256 zeros. On the other hand, the *FIR* sub-band echo canceller uses, in each sub-band, an adaptive transversal filter with 1024 coefficients (zeros). Here the adaptation step for the *FIRBNK* is μ=0.001 while for the *AFMBNK* an adaptation gain of τ=0.01 is used. The input signal for identification is a white noise as above for both the algorithms. The *ERLE*'s (dB) of both algorithms are plotted in Fig.7 as below.

Also in this case, the *AFMBNK* performs better than the FIR equivalent sub-band echo canceller although its computational complexity is consinstently lower.

All the algorithms have been implemented on a *Gateway 2000 P5-60* machine equipped with a *Pentium* Processor using *BORLAND Turbo Pascal 7.0*. We can summarize the results comparing algorithms in terms of speed, convergence, stability, and *ERLE*. In this case we use the echo path $H_1(z)$ with input white

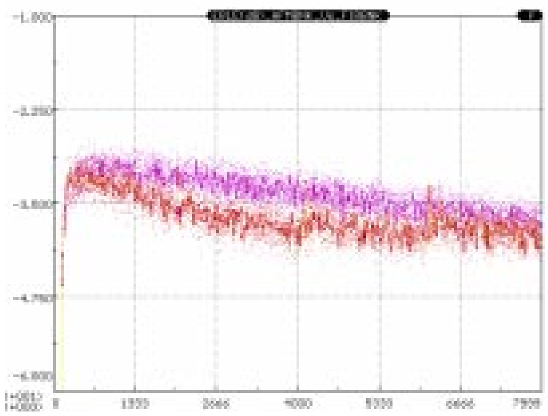

Fig.7. Comparison of the *ERLE*'s obtained by the *AFMBNK* and the *FIRBNK* echo cancellers.The echo path is $H_2(z)$.

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In this Table convergence implies that the algorithm ensures that the coefficients move to a fixed point in the parameter space, whereas stability specifies that this point is a stable filter point. The complexity is given in terms of number of multiplications vs. numerator (N) and denominator (M) orders. Note that due to the QMF filter banks, N_2 M_2 orders are smaller than N_1 M_1 and much smaller than N_3, thus the AMFBNK seems to have the best trade-off between performance and complexity.

Table 2. Algorithm properties and implementation results

Algorithm	Order	Time	τ
AFM	8/10	1s:87hs	1e-3
AFMBNK	4/5	7s:25hs	1e-3
FIRBNK	32	8s:24hs	1e-3

Alg.	**Conver.**	**Stability**	**ERLE**	**Multipliers**
AFM	not yet	reached only with low noise	-22dB	$5N_1+3M_1+1$
AFMBNK	not yet	reached only with low noise	-20dB	$5N_2+3M_2+1$+QMF
FIRBNK	yes	sure	-12.5dB	$8N_3+6$+QMF

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