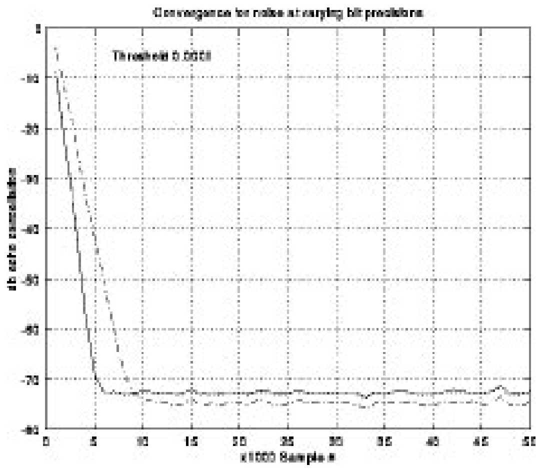

Figure 3. Convergence rate (simulation) of the modified FAP algorithm according to input signal bit widths. Solid, dotted and dashed lines are for 13, 11, and 8 bit inputs, respectively.

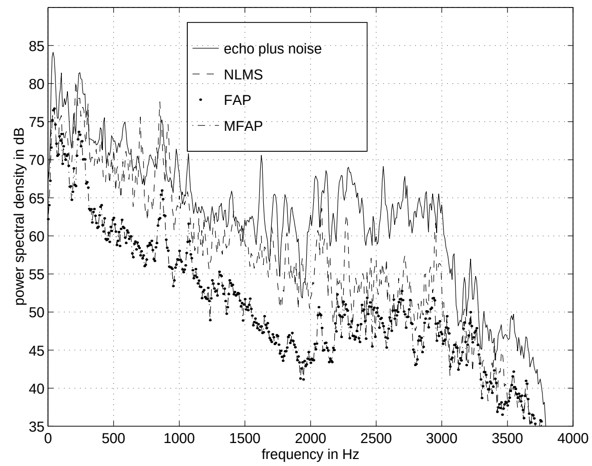

Figure 5. Power spectral densities for noisy echo signal and residuals of NLMS and two FAP algorithms. (highway case, after noise suppression)[7]

difference in performances of two FAP's that we have implemented. Even though we have observed error accumulating phenomena from the FAP of [5], it did not create any performance degradation with our collected files (the longest file lasts about 30 minutes).

As figure 4 and 5 depict, both FAP algorithms perform better than the LMS algorithm in the noisy environment. Note that the LMS algorithm sometimes results in jumps (no cancellation) in the presence of noise while we do not observe the same phenomena from the FAP algorithms. We confirmed this phenomena by listening the output files (i.e., from the LMS output files, we could hear some echo bumps.) We have found that FAP algorithms provide about 5 - 15 dB better in terms of residual energy calculation.

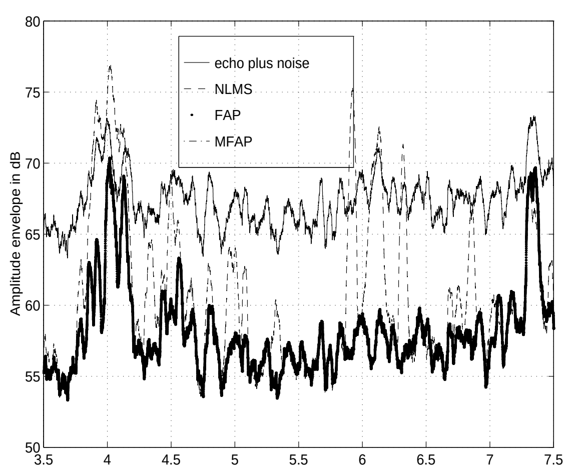

Figure 4. Envelops of noisy echo signal and residuals for NLMS, two different FAP's. (highway case, noise suppression was applied.)[7]

6. Conclusion

In this paper, we presented our new FAP algorithm and its empirical studies. The performance of the new FAP algorithm is better than that of NLMS algorithm with the small increase of implementational complexities. We believe that the new FAP algorithm is a better performing and implementable algorithm in real applications.

References

[1] CNN Report, *The hazards of driving and dialing,* Oct., 1995, located at "www.cnn.com/HEALTH/9510/cell phone".

[2] S. Oh, J. Crockett, and E. Arkan, "Development of an acoustic echo canceller in adverse environment with a fixed point DSP," DSPx95, 1995.

[3] M. Tanaka, Y. Kaneda, S. Makino, and J. Kojima, "Fast projection algorithm and its step size control," ICASSP-95, 1995

[4] S. Gay and S. Tavadia, "The fast affine projection algorithm," ICASSP-95, 1995.

[5] Y. Kaneda, M. Tanaka, and J. Kojima, "An adaptive algorithm with fast convergence for multi-point sound control," ACTIVE95, 1995.

[6] S. Douglas, "Efficient approximate implementations of the fast affine projection algorithm using orthogonal transforms," ICASSP-96, 1996.

[7] M. Ali and S. Oh, "An adaptive filtering method using a modified fast affine projection algorithm," Texas Instruments Internal Report, Aug., 1996.

[8] D. Linebarger, "Redundancy Averaging with large arrays," IEEE Trans. SP, vol. 41, no.4, pp1707-1710, Apr, 1993.

4.1 Regularization parameter

We have found that the regularization parameter plays an important role to determine the performance of FAP algorithm. Note that in the regularization parameter dominant case (i.e., the correlation matrix are near zeros), the reciprocal value of the regularization parameters will determine the maximum values which can be represented by DSP (Q point). Therefore, special care is needed to determine the value of regularization parameter to optimize the performance and numerical stability. Refer to Figure 1 for the performance differences of the FAP algorithm according to different regularization parameter values. We have decided to observe the performance degradation due to the regularization even though some research papers addressed this issue from the theoretical point of view since it is easier to determine the regulazation value for the fixed point implementation.

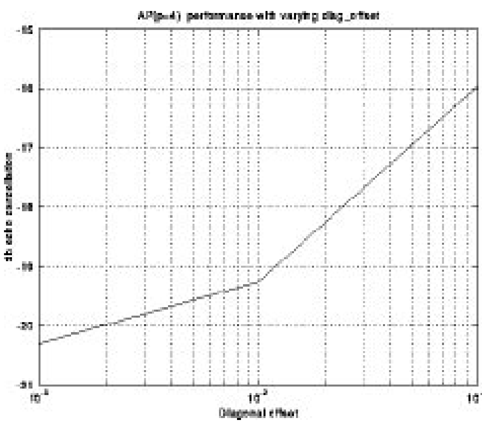

Figure 1. Performance degradation of FAP according to the regularization parameter values

What we found was the fact that the regularization parameter should be large enough to reduce the fixed point implementation stress. However, if the value of the parameter is too big, it will degrade the performance.

4.2 Dynamic range

The FAP performance can be also affected significantly by the dynamic range of the input signal. For the real application, the A/D converter can be slightly mal-adjusted; it results in not utilizing full dynamic range of the converter. For example, Texas Instruments' VBAP can sample 13 bit linear at 8 kHz sampling rate provided the analog signal level is 2.5 $V_{p\text{-}p}$. The maximum analog voltage level can be achieved by precisely adjusting potentiometers. However, due to the nature of the analog circuits and components, the signal level does not achieve 2.5 $V_{p\text{-}p}$ always. For this case, the VBAP cannot utilize 13 bit precision fully for the input samples. Therefore, it is critical to understand the behavior of an algorithm when the dynamic range of the input signal is not utilized fully.

We have found that our FAP implementation is fairly robust against the deficiency of input signal bit precision. The performance of the FAP maintains even when the input signal bit width is only 11 bits. However, when the maximum bit width of the input signal becomes 8 bit, the performance of the FAP algorithm degrades by 15 dB. Refer to Figure 2 for the performance degradation of FAP and LMS algorithms according to the input signal bit width.

The input signal bit precision also affects the convergence rate of the algorithm. To see the convergence behavior, we did a simulation study of the FAP algorithm with white Gaussian noise since it is difficult to see the convergence rate and the steady state behavior with speech signal. The FAP algorithm converges more slowly when the bit precision of the input signal is 10 bits compared to 13 bits. However, the LMS algorithm suffers more compared to the FAP algorithm (LMS algorithm failed to converge with 8 bit input), which is a surprising result since it is known that LMS algorithm is a fairly robust algorithm against the bit width variation. Refer to figure 3 for the convergence difference of the FAP algorithm due to the input bit width differences.

5. EXPERIMENTAL EVALUATION

The modified FAP algorithm with the Toeplitz constraint has been also evaluated in the real environment. To present the result from the real environmental test in this paper, we have collected speech files inside a car. The speech files were processed using the FAP proposed by [5], our FAP, and NLMS algorithms. For our evaluation, we have used $p=4$, $\mu=0.3$, and $N=320$ taps. We have found no

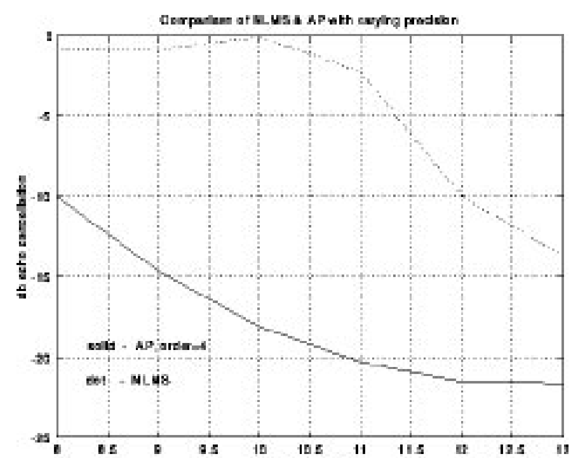

Figure 2. The performance degradation of FAP and LMS algorithms according to input signal bit widths

$$W_{k+1} = W_k + \mu X_k (X_k^T X_k + \delta I)^{-1} e_k \qquad (1)$$

where X_k is a $p \times N$ data matrix, e_k is a error vector, and δ is called a regularization parameter.

The FAP algorithm can be viewed as the generalization of LMS algorithm. We rewrite equation (1) as

$$W_{k+1} = W_k + \mu X_k g(k) \qquad (2)$$

where

$$g(k) = (X_k^T X_k + \delta I)^{-1} e_k. \qquad (3)$$

Note that similar to the RLS algorithm, the FAP algorithm contains the matrix inversion operation, which is considered numerically not desirable for the fixed point implementation even though the dimension of the correlation matrix of the FAP is much smaller than that of the RLS algorithm. Since the previously proposed FAP algorithms involves many variables, it is hard to follow in implementation unless special care is not taken. Most of the variables of the FAP algorithms are, however, necessary only for matrix inversion in equation (3).

Let's set the numerical complexity aside while we are analyzing the FAP algorithm for the moment. We can simplify the algorithm with a direct matrix inversion method in equation (3) as List 1.

The numerical complexity of the simplified FAP algorithm is about $2N+O(p^3)$. The simplified FAP algorithm motivates us to develop a new FAP algorithm. The new FAP algorithm will be described in the next section.

3. A New FAP Algorithm

From step 4 of List 1, the main difficulty of implementing an FAP algorithm is how to invert the correlation matrix (R_p) effectively. Both [3] and [4] suggested to use a sliding window fast RLS type algorithm and [5] suggested use of the matrix inversion lemma twice. As we pointed out, the FTF (Fast Transversal Filter) type of algorithm is too complicate to implement and numerically unstable, and, hence, it is difficult to use in the real environment. The use of the matrix inversion lemma is also not guaranteed to be stable. It is worthwhile to mention that Douglas [6] proposed to implement the FAP using approximation in transform-domain. This method also has an inherit problem as [6] proposed to use either periodic restart or leaky integration to reduce the error accumulation in the fixed point implementation.

List 1. Simplified FAP algorithm with a direct matrix inversion (partially taken from [3])

1. $r_{p-1}(k) = r_{p-1} + x(k)\bar{x}_{p-1}(k-1)$
   $-x(k-N)\bar{x}_{p-1}(k-(N-1))$
2. $e(k) = y(k) - \bar{x}_N(k)^T z_N(k) - r_{p-1}(k)^T s_{p-1}(k-1)$
3. $\begin{bmatrix} \bar{e}_p(k) \\ * \end{bmatrix} = \begin{bmatrix} e(k) \\ (1-\mu)\bar{e}_p(k-1) \end{bmatrix}$
4. $g_p(k) = R_p^{-1}(k)\bar{e}_p(k)$
5. $s_p(k) = \begin{bmatrix} 0 \\ s_{p-1}(k-1) \end{bmatrix} + \mu g_p(k)$
6. $z(k+1) = z(k) + \overline{x_N}(k-p-1)s_p(k)$

To develop a new algorithm, consider the matrix R_p as

$$R_p = \begin{bmatrix} \hat{r}_0(n) & \hat{r}_1(n) & \dots & \hat{r}_{p-1}(n) \\ \hat{r}_1(n-1) & \hat{r}_0(n-1) & \dots & \hat{r}_{p-2}(n-1) \\ \hat{r}_2(n-2) & \hat{r}_1(n-2) & \dots & \hat{r}_{p-3}(n-2) \\ . & . & . & . \\ . & . & . & . \\ \hat{r}_{p-1}(n-p+1) & \hat{r}_{p-2}(n-p+1) & \dots & \hat{r}_0(n-p+1) \end{bmatrix} \qquad (5)$$

where

$$\hat{r}_\tau(n) = \sum_{i=0}^{N-1} x(n-i)x(n-i-\tau)$$

is the estimate of the autocorrelation at lag τ and at time instant n based on the past N input data.

Therefore, if $N>>p$ (and it is true for most applications), the following holds:

$$\hat{r}_\tau(n) \approx \hat{r}_\tau(n-1) \approx \dots \approx \hat{r}_\tau(n-p+1). \qquad (6)$$

Thus, equation (5) can be approximated as

$$\tilde{R}_p = \begin{bmatrix} \hat{r}_0(n) & \hat{r}_1(n) & \dots & \hat{r}_{p-1}(n) \\ \hat{r}_1(n) & \hat{r}_0(n) & \dots & \hat{r}_{p-2}(n) \\ \vdots & \vdots & \dots & \vdots \\ \hat{r}_{p-1}(n) & \hat{r}_{p-2}(n) & \dots & \hat{r}_0(n) \end{bmatrix} \qquad (7)$$

and the correlation matrix becomes a Toeplitz matrix.

The inversion of a Toeplitz matrix is dominated by $O(2p^2)$ operations [7]. Forcing a Toeplitz structure on correlation matrices that are only asymptotically Toeplitz has been done before [8]; the more efficient algorithm, therefore, exists to inverse matrix defined in equation (7). The new algorithm uses updates for the autocorrelations $\hat{r}_\tau(n)$. These updates require adding a new element corresponding to the time instant n and subtracting that corresponding to n-N+1. It can be done in step 1 of List 1. There is, therefore, no propagation of error in calculation of the updates for the autocorrelations.

4. Empirical analysis of FAP algorithm

To understand the FAP algorithm in the real environment, we have conducted a study to analyze the behavior of the FAP algorithm empirically. We have implemented our new FAP algorithm and evaluated it empirically.

A FAST AFFINE PROJECTION ALGORITHM FOR AN ACOUSTIC ECHO CANCELLER USING A FIXED-POINT DSP PROCESSOR

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ABSTRACT

We present an empirical analysis of the fast affine projection (FAP) algorithm to be used in an acoustic echo cancellation application using a fixed point DSP processor. We also introduce a modified FAP algorithm that was developed based on our FAP study. Our analysis study shows that the modified FAP algorithm is more robust and provides more consistent performance than the LMS algorithm. The new FAP algorithm is also numerically efficient and easy to implement with a fixed point DSP processor.

1. INTRODUCTION

A recent report shows that people with a cellular phone in a car run a 34 percent higher risk of having an accident [1]. Due to these safety concerns, many countries have already considered banning the usage of hand-held cellular phones in an automobile. Hands-free phone conversation in a car has thus received much attention in answer to these concerns. As a consequence, many ongoing studies are dedicated to development of an acoustic echo canceller that can be used in a car (i.e., adverse environment). We have developed and implemented an acoustic echo canceller with an adaptive noise suppression algorithm using Texas Instruments' fixed point DSP processor (TMS320C5x) [2].

Recently, many researchers reported several adaptive algorithms for acoustic echo cancellation applications. One of the most interesting developments to us was the fast affine projection (FAP) algorithm [3-4]. The authors of the FAP algorithm indicated that FAP can achieve the performance level of the RLS (Recursive Least Square) algorithm for the computational price of the LMS (Least Mean Square) algorithm. The numerical complexity of the FAP algorithm is about $2N + 20p$ where N is the number of Taps and p is the projection order ($p<<N$) whereas the fast RLS algorithm's complexity is about *8N*. The FAP algorithm, however, contains five division operations, which result in requiring more operational DSP cycles than it appears. This is because one division requires more than 20 cycles whereas multiplication-and-addition (MAC) operation requires 1 cycle.

It was also pointed out [6] that the sliding window fast RLS algorithm (which is used in [3-4]) is difficult to implement, memory intensive, and potentially numerically unstable unless special care is taken in its implementation. It is worthwhile to note that two alternate approaches for the FAP have been proposed in [5] and [6]. We have found that the FAP algorithm proposed by [5] is easier to implement than any other proposed FAP algorithms for the price of slightly higher computational expense ($5p^2$ instead of *20p*).

The FAP algorithm proposed by [5], however, has an inherit problem specifically for fixed point DSP implementation; error accumulation. Since at each iteration, it tracks the correlation matrix ($X_k^T X_k$) in the inverse domain, it is possible that the matrix values do not reflect the true inverse of the correlation matrix with finite precision computation.

This paper thus deals with developing an FAP algorithm by analyzing the existing FAP algorithms so that an algorithm can be used in the real environment. We first analyzed the existing FAP algorithms and developed a modified FAP algorithm. We have designed an evaluation board using TMS320C5x DSP processor with 13 bit linear A/D converters (VBAP: voice band audio processor) and interfaces with variety of cellular phones [2]. This board was used to develop and test algorithms in the real hands-free environment in a car so that we can develop the acoustic echo canceller that performs in actual usage.

2. FAP ALGORITHMS

The FAP algorithm is a fast realization of the affine projection algorithm. The coefficient (W_k) update equation of the affine projection algorithm is