PERFORMANCE LIMITATIONS OF A NEW SUBBAND ADAPTIVE SYSTEM FOR NOISE AND ECHO REDUCTION

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ABSTRACT

Oversampled Subband Adaptive Filters (OS-SAFs) have shown good performance in echo and noise cancellation. In order to understand the ultimate limitations, two issues must be considered: convergence rate reduction, and asymptotic performance reduction resulting from aliasing and other effects. When the Normalized Least Mean Square algorithm is employed for OS-SAFs, the convergence rate is considerably decelerated due to the coloring of subband signals. We have already proposed new techniques for improving the convergence rate based on spectral emphasis and decimation of the subband signals. In this paper, we theoretically analyze the effects of the proposed methods on the steady-state performance of the system. Also, simulation results are presented and compared to theoretical results. It is concluded that the introduced filter spectral images do not significantly contribute to the error. Rather, the performance is mainly limited by in-band aliasing.

1. INTRODUCTION

The Normalized Least Mean Square (NLMS) algorithm is a popular method used in adaptive filtering. It is a simple and stable adaptation technique of low complexity. However, NLMS convergence is sensitive to the spectral flatness of the reference input and may be slow when the input signal is colored.

In many adaptive applications, Over-Sampled Subband Adaptive Filters (OS-SAFs) have become a common practical solution [1], overcoming the spectral flatness problem by extending the NLMS to multiple bands. In addition to the well-known advantages of subband processing, oversampled systems offer a simplified implementation and much reduced distortion (aliasing) as compared to critical sampling implementations [2,3]. For many real-time applications requiring low processing delay, long analysis/synthesis time-windows cannot be employed. Consequently, high over-sampling factors (2 or more) are used to minimize the aliasing distortion that would occur in critical sampling or low over-sampling cases. When adaptive filters are used in these highly over-sampled subband structures, the over-sampled inputs to each subband adaptive filter are colored, leading to slow convergence of the NLMS.

We have already proposed two different techniques, based on spectral emphasis and decimation of the subband signals, to improve the convergence rate of the OS-SAFs [4,5]. In this paper, we analytically and experimentally evaluate the limits on the steady-state performance of the system due to the use of the proposed techniques. The employed spectral techniques offer a simplified and practical analysis of the performance limits [6].

The employed OS-SAF system is described in Section 2. Sections 3 and 4 introduce the convergence improvement techniques. In Sections 5 and 6, the performance limits are presented and discussed. Finally, conclusions are discussed in Section 7.

2. THE OS-SAF SYSTEM

Fig. 1 shows an SAF system in an echo cancellation framework. The system employs a highly over-sampled GDFT uniform filterbank. Through the DFT, a single prototype filter (an analysis window of length L samples) is modulated into K complex filters ($K/2$ real bands due to Hermitian symmetry). Data frames of length L samples are multiplied by the analysis window and shifted by R samples. To achieve low processing delay, a high oversampling factor $OS = K/R$ is usually employed. The added computation cost due to over-sampling is partly compensated by the use of shorter analysis prototype filters, and the efficiency of the hardware structure [3]. Referring to Fig. 1, each adaptive processing block is generally an adaptive filter working on a specific frequency band thus modeling a narrow frequency band of the echo plant. As a result of oversampling, even for white reference noise, input signals of the adaptive filter are no longer white, as depicted in Fig. 1 for an oversampling factor of 4.

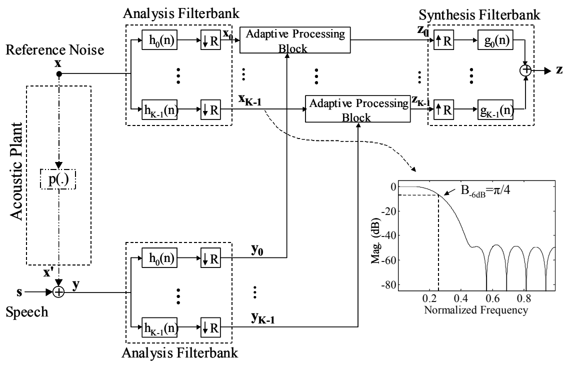

Fig. 1: Block diagram of SAF system

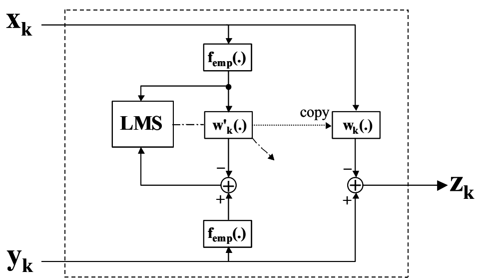

Fig. 2: Adaptive processing block employing whitening spectral emphasis.

This significantly degrades the convergence properties of the LMS algorithm compared to the critical sampling case, where all subband signals are almost white.

3. WHITENING BY SPECTRAL EMPHASIS

To cope with the slow convergence problem of OS-SAF systems, we proposed whitening of the oversampled subband signals through spectral emphasis [4,5].

Shown in Fig. 2 is the adaptive processing block diagram employing whitening by spectral emphasis. Considering the subband signal spectrum (for $OS = 4$ case), we have designed and employed an emphasis filter $f_{emp}(.)$ amplifying the high three quarters of the spectrum (in each subband) while leaving the low quarter band intact. The emphasized signals are used only to improve the convergence characteristics of the adaptive filter. As shown in Fig. 2, the adaptive filtering is done in the main branch (right branch) while the side branch does the spectral emphasis and LMS weight adaptation. In each iteration, the updated weights are copied from the side branch to the main branch. The use of spectral emphasis has no effect on the modeling behavior of the adaptive filter. Since spectral emphasis has been applied to both reference and noisy inputs, its effect will be cancelled [5]. This is also verified by the simulation results. The spectral emphasis basically improves the convergence rate through amplification of small eigenvalues at the cost of additional computations for the spectral emphasis filtering.

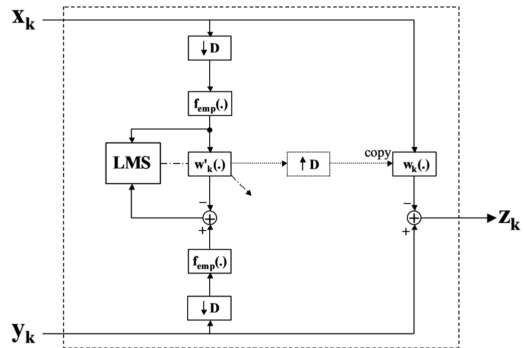

Fig. 3. Adaptive processing block employing whitening by decimation.

4. WHITENING BY DECIMATION AND SPECTRAL EMPHASIS

Since the subband reference signal has a limited bandwidth, we further decimate the subband signals, consequently generating whiter signals that will ultimately increase the convergence rate. Fig. 3 shows the adaptive processing block employing the whitening by decimation technique [5]. In this research for $OS = 8$, $D = 1,2,4$ are employed to limit the in-band aliasing. After each LMS weight update, adaptive filter coefficients are upsampled and copied to the mirror filter in the main branch. Although the upsampling creates in-band images in the filter spectrum, they are filtered out by the input signal in the main branch. As verified by the simulation results, the filter spectral images do not limit the performance.

Whitening by decimation improves the convergence rate by increasing the effective bandwidth of the reference input. Nevertheless, it cannot deal with the stop-band region of the prototype filter. Therefore, we employ a spectral emphasis filter following the whitening by decimation method. The block diagram of the combined adaptive processor is shown in Fig. 3. A low-cost second-order IIR spectral emphasis filter has been employed here.

5. STEADY-STATE PERFORMANCE LIMIT

Even in the absence of uncorrelated observation noise, the asymptotic performance of the SAF system is limited by the noncausality of the adaptive filters, and the aliasing of the subband signals due to nonideal analysis filters [7,6]. Following the approach of [6], we employ the Signal-to-Alias Ratio (SAR) defined as,

$$SAR = \int_0^{\pi} |H(e^{j\omega})|^2 \, d\omega \Bigg/ \int_{\pi/M}^{\pi} |H(e^{j\omega})|^2 \, d\omega \qquad (1)$$

where $H(e^{j\omega})$ represents the filterbank prototype filter, and M is the subband decimation factor. The SAR provides a theoretical limit to the steady-state system performance measured by the Echo Return Loss Enhancement (ERLE).

5.1. System setup and the SAR limit

The OS-SAF system is employed in an echo cancellation setup. In order to take advantage of the over-sampling properties of the system and to observe the effects of the convergence improvement techniques, the OS-SAF system was configured for $R = 4$ and $K = 32$ subbands, so the over-sampling factor was $OS = K / R = 8$. Two different analysis filters were employed with time durations of L=512 and L=1024 samples. Three different subband decimation rates were chosen, D=1,2,4. To calculate the SAR limit, the overall subband decimation rate in (1) was set to $M = R \times D$ since the subband signal goes through two stages of decimation as shown in Fig. 3. Table 1 depicts the calculated SAR limit for the two employed prototype filters. As expected, for each window, the in-band aliasing increases as D varies from 1 to 4.

5.2. Performance limit measurements

To practically evaluate the performance limit, we used a very long sequence of white noise as input. The subband adaptive filters were chosen sufficiently long to achieve the best steady-state performance. For the echo transfer function, we employed echo-path models suggested by the ITU-T Recommendation G.168 [8], as well as a typical measured acoustic transfer function.

We measured the ERLE in frequency domain as:

$$\mathrm{ERLE} = 20\mathrm{Log}10 \frac{\sum_{k=1}^{K/2} | y_k |}{\sum_{k=1}^{K/2} | z_k |}, \qquad (2)$$

where, y_k and z_k are the subband signals (shown in Figures 2 and 3). The ERLE is calculated after full convergence of the adaptive system. Measuring the ERLE directly in the frequency domain avoids the inclusion of reconstruction errors after time synthesis. However, since the employed filterbank achieves a near-perfect reconstruction performance, the time-domain input/output measured ERLE's were very close to the ERLE calculated by (2).

We practically verified that the spectral emphasis of the subband signals does not affect the steady-state performance. As the use of spectral emphasis accelerates the convergence rate, we employed it in all of the tests.

Table 1: The SAR limit (dB) for different D and L parameters

D	1	2	4
L=1024	83.1	74.7	65.3
L=512	77.0	68.9	59.5

Table 2: The steady-state ERLE and the SAR limit, for L=512 prototype filter

D	1	2	4
ERLE (dB)	63.2	61.4	51.1
SAR (dB)	77.0	68.9	59.5

Table 3: The steady-state ERLE and the SAR limit, for L=1024 prototype filter

D	1	2	4
ERLE (dB)	64.9	62.7	56.0
SAR (dB)	83.1	74.7	65.3

Tables 2 and 3 show the steady-state ERLE of the OS-SAF system, employing the convergence improvement techniques, and one of the ITU-T G.168 echo transfer functions.

The employed noise was a long sequence of 16-bit samples of white noise, sampled at 8 kHz. Allowing for sufficient headroom, the input dynamic range was limited to around 75 dB. Thus, an ERLE of more than 75 dB implies a time-domain output of magnitude less than 1. Simple integer truncation of the output then leads to infinite time-domain ERLE. As a result, in our setup, an ERLE of 75 dB should be used as a higher bound on the SAR limits. To test this, we simply employed echo plants consisting of integer delays (R*D samples of delay). As expected, this leads to perfect cancellation even in the presence of aliasing errors [6]. Measured frequency domain ERLE's were around 75 dB and as a result input/output ERLE's were infinite.

6. DISCUSSION

As noted by many researchers, there are many factors limiting the SAF performance, most agreeing that aliasing is the dominant one. The employed SAR measure only takes the aliasing into account. Most notably, however, it fails to consider the modeling errors due to truncation and noncausality of the optimal adaptive filter [6,7].

Referring to Tables 2 and 3, obviously the SAF system cannot reach the SAR limit (assuming a maximum cap of 75 dB for the SAR) in all conditions. Starting from the simpler case of $D = 1$, there are performance gaps of 10 and 12 dB (for L=1024 and L=512, respectively). The most probable justification for this could be a very slow final convergence due to the eigenvalue spread problem [9]. In fact, due to the high oversampling rate of 8, small eigenvalues of the autocorrelation matrix hinder and disrupt the long-term convergence. To further investigate this effect, we performed a set of tests using the RLS algorithm (without the convergence improvement techniques) with oversampling factors of OS=2, 4, 8 (L=512, K=32, D=1). Table 4 depicts the performance gap between the achieved steady-state performance (ERLE) and the theoretical limit (SAR). The results clearly demonstrate that as the oversampling factor decreases, the performance gap ($\text{SAR} - \text{ERLE}$) closes.

To compare the results of this research with the previously reported results in the literature, it is noteworthy that none of the previously reported steady-state performance tests use high oversampling factors of 2, 4 and 8 as we employ. Instead, often oversampling factors close to 1 is used (as in [6]). However low over-sampling leads to disadvantages mentioned in the introduction.

Modeling errors (due to signal decimation) could be another cause of the performance gap. Although, current tests cannot discriminate between the effects of various causes.

To verify that adaptive filter spectral images (due to filter interpolation in Fig. 3) do not contribute to additional errors, we compared the time-domain ERLE and the frequency-domain ERLE values obtained through Eq. (2). In all tests, the two ERLE values were very close.

7. CONCLUSION

OS-SAF is a common practical choice for many adaptive systems. However, over-sampling leads to coloration of the signals at the adaptive filter input. This decelerates the convergence rate of the NLMS technique which is sensitive to the whiteness of the input signal. Two different techniques (spectral emphasis and whitening by decimation) have already been proposed to improve the convergence rate [4,5]. It is been shown that a combination of the two methods is very effective in improving the convergence rate [5].

In this research, we are mainly concerned with the effects of the proposed convergence improvement techniques on the steady-state performance of the system. Simulation results show that the spectral emphasis has no considerable effect on the steady-state performance as already predicted in [5]. On the other hand, whitening by decimation increases the aliasing error and limits the performance of the system.

In the final stage, the adaptive filter in the side branch is interpolated to obtain the adaptive filter of the main branch (Fig. 3). Although this adds spectral images to the adaptive filter in the main branch, our simulation results show that the images do not contribute to any extra errors. This is because the signal in the main branch of the adaptive filter does not contain considerable energy in spectral regions where the filter images are located [5].

Table 4: The Performance gap ($\text{SAR} - \text{ERLE}$ in dB) versus the oversampling factor, for L=512 prototype filter, using the RLS algorithm.

OS	$\text{SAR} - \text{ERLE}$ (dB)
8	11.6
4	6.7
2	3.7

Our future research is directed towards a deeper understanding of the steady-state behavior of the system, including conducting analysis and tests to better identify the role of modeling and truncation errors.

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