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HYBRID SUBBAND/FREQUENCY–DOMAIN ADAPTIVE SYSTEMS FOR ACOUSTIC ECHO CANCELLATION

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ABSTRACT

Subband filter implementations have many desirable properties. However, when used to implement adaptive filters, various effects (such as residual errors and slow convergence due to aliasing) occur and reduce performance. On the other hand, it is known that frequency–domain adaptive filters do not suffer from these problems despite being (nearly) equivalent to subband adaptive filters with 'poor' filter banks (modulated sinc frequency responses). In this paper, we formulate design criteria for subband adaptive systems which should lead to adaptive filters exhibiting better properties, similar to what can be obtained with frequency–domain techniques. The aim is to focus on an alternative adaptation scheme, which feeds back the fullband error instead of the subband errors as is done in a standard subband approach. This is an attempt to generalise frequency–domain alias–compensation techniques to subband adaptive systems.

1. INTRODUCTION

For high quality echo cancellation long acoustic echoes need to be suppressed. This easily leads to adaptive filters with several thousands of filter taps. Classical LMS based echo cancellers are unattractive for real–time processing as they are suboptimal from a computational point of view. Moreover, speech signals have a coloured spectrum and it is well known that the performance of classical adaptive algorithms is not optimal in that case. Alternative solutions have been proposed and they are mainly based on either subband or frequency–domain techniques.

Multirate adaptive echo cancellation schemes have been a topic of interest for many years now. Still, with the available techniques, it is difficult to meet all the required specifications. *Frequency–domain techniques* are well understood [1]. They are based on an exact translation of the Block–(N)LMS algorithm to the frequency domain, and hence their performance is very tractable. On the other hand, *subband adaptive filters* —at first sight— may have lower complexity and better performance . Unfortunately, this picture of the subband approach is certainly too optimistic.

Subband adaptive filters and frequency–domain based techniques are mostly considered as being different approaches. In this paper we will consider the frequency–domain algorithm as a special case of subband adaptive filtering and point out why frequency–domain techniques are better from certain perspectives. The ultimate goal is then to transplant their desirable properties to the subband approach.

2. SUBBAND ADAPTIVE FILTERING

The standard setup for a subband acoustic echo canceller is shown in figure 1. The loudspeaker and microphone signal

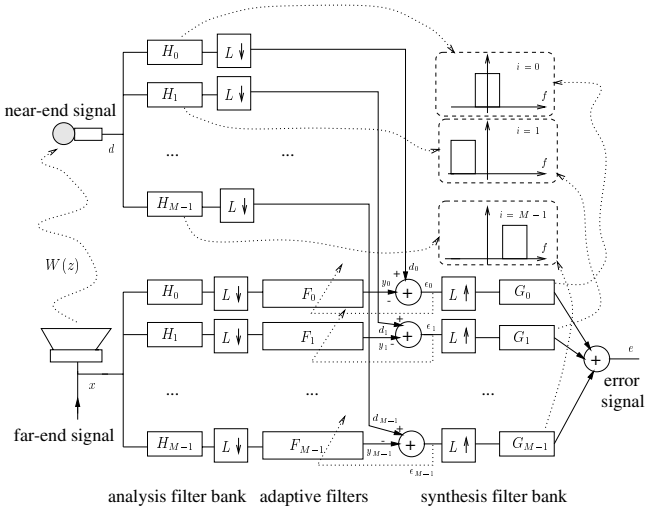


Figure 1: Subband adaptive filter with ideal filter banks

are fed into identical M–band analysis filter banks. After subsampling with a factor L, adaptive filtering is done in each subband and finally the outputs of the subband adaptive filters are recombined in the synthesis filter bank. Due to aliasing effects, this setup will only work for $M \geqslant L$. The ideal frequency response of the analysis bank filters H_i and synthesis bank filters G_i are shown (ideal bandpass filters). If L is chosen equal to M a critically downsampled

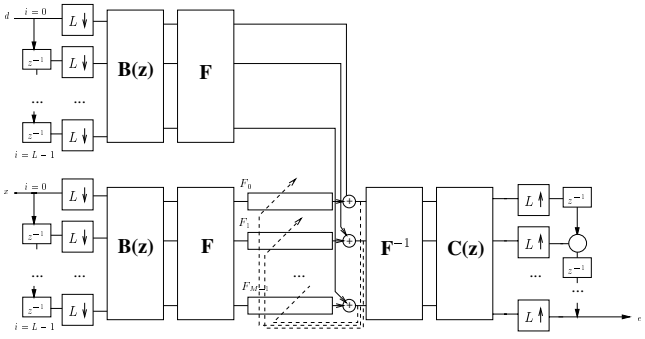


Figure 2: DFT modulated subband adaptive echo canceller

subband adaptive filter is implemented. In this paper we will consider the more general class of oversampled subband schemes for which $M > L$.

2.1. DFT modulated subband schemes

Oversampled subband acoustic echo cancellers are often based on DFT modulated filter banks. The M subband filters are derived by frequency shifting a well–designed prototype lowpass filter $h_0[k]$. DFT modulated filter banks may be implemented efficiently by means of a polyphase decomposition and fast signal transforms. In [2] a general framework for oversampled DFT modulated subband systems was proposed. An M–band DFT modulated filter bank with L–fold downsampling can be implemented as a tapped delay line of size L followed by a structured $M \times L$–matrix $\mathbf{B}(z)$, containing polyphase components of the prototype h_0, and an $M \times M$–DFT matrix $\mathbf{F}$. In case of DFT modulated filter banks, figure 1 can be redrawn resulting in figure 2. The synthesis bank is constructed in a similar fashion with an $L \times M$–matrix $\mathbf{C}(z)$.

2.2. Subband adaptive filters : performance

Splitting signals into subbands seems very promising, since for coloured input spectra the convergence of classical adaptation algorithms is too slow. In the subband case, each subband signal will have a flatter spectrum, leading to improved convergence. Unfortunately, this picture of the subband approach is certainly too optimistic. Optimal computational savings will be obtained by critical downsampling, but critical subband schemes introduce aliasing and lead to an undesired residual error [3]. In oversampled subband adaptive systems reduced aliasing distortion is traded off for extra computational cost and slower steady–state convergence. Moreover, the assumption that in the standard subband approach, a full–band echo path of length N can be identified by M subband filters with reduced length $\frac{N}{L}$ (L is the downsampling factor) seems to be quite wrong. The adaptive identification process has to track (many) more than $\frac{N}{L}$ samples per subband and an extra delay has to be inserted in the near–end signal path [4].

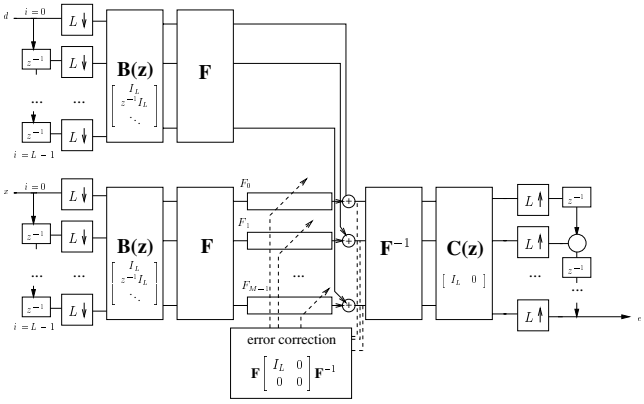


Figure 3: PBFDAF as a DFT modulated oversampled subband scheme

2.3. Frequency–domain techniques

On the other hand, many frequency–domain based techniques [1] have a better overall performance thanks to dedicated projection operations. These operations lead to aliasing–free gradient estimates and increased convergence behaviour without residual error. Furthermore, using frequency–domain techniques echo paths of finite length can be modelled exactly using a finite number of coefficients.
Many frequency–domain based techniques can be considered as a special case of subband adaptive filtering. As an example we consider the partitioned block frequency–domain adaptive filter (PBFDAF) [5]. It appears that the PBFDAF is just a special case of an oversampled DFT modulated filter bank as illustrated in figure 3, having a simple filter bank with low frequency selectivity. This easily admits re–transformation of subband errors into time domain where a repair operation restores aliasing errors, without introducing extra delay (see "error correction" block).

2.4. Realization conditions

In this paragraph some design criteria for DFT modulated subband adaptive systems are formulated, which should lead to subband adaptive filters exhibiting better properties, similar to what can be obtained with frequency–domain techniques. Frequency–domain subband algorithms such as the PBFDAF typically use simple filter banks and have a subband oversampling factor equal to 2. Most subband adaptive systems perform better on this part as their oversampling factors are normally less than 2 and more advanced filter banks are employed. Bringing together both ideas into a 'hybrid' approach, should therefore ultimately lead to better overall performance.
An analysis/synthesis filter bank set is therefore (preferably) designed such that the following 3 conditions are met [6] :

1. the analysis filters are frequency selective. This prevents inserting too much inter–subband aliasing com-

ponents as it has an inhibitive effect on the convergence of the adaptive filters.

2. the analysis/synthesis filter bank set is (nearly) perfect reconstructing i.e. a near–end source signal is not distorted by the analysis/synthesis system.

3. the acoustic path can be modelled by finite–length adaptive subband filters. In case of DFT modulated filter banks (see figure 3) the lower branch models a time–invariant path only when

$$\mathbf{C}(z)\mathbf{F}^{-1}\text{diag}\{F_m(z)\}\mathbf{FB}(z)$$

is pseudo–circulant.

It appears that for the PBFDAF the above mentioned conditions about perfect reconstruction and perfect path modelling are indeed fulfilled [6].
Furthermore, one can observe (figure 3) that the subband filters are adapted based on corrected subband errors. This in fact corresponds to the overlap–save or overlap–add procedure upon which frequency–domain techniques are based. The correction implies passing the subband errors through the synthesis filter bank in order to obtain a fullband, i.e. an alias–free, error signal.

3. FULLBAND ADAPTATION SCHEME

The aim is to focus on an adaptation scheme for subband adaptive systems, which adjusts the subband adaptive filters F_m (see figure 4) based on the fullband error e instead of using the subband error signals $\epsilon_m = d_m - y_m$ as is normally done in a standard subband scheme (see figure 1). This is

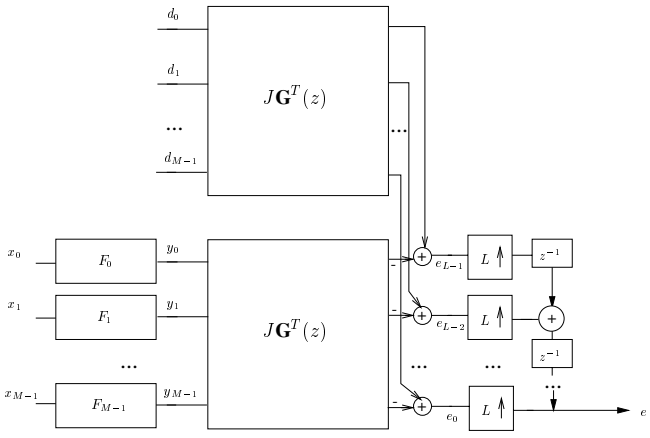


Figure 4: Oversampled subband adaptive filter : synthesis part

an attempt to generalise and extend the frequency–domain alias–compensation to subband adaptive systems. $\mathbf{G}(z)$ in figure 4 is the so–called synthesis polyphase matrix and is defined as

$$\mathbf{G}_{ij}(z) = G_{i_{j:L}}(z) \qquad \begin{cases} i = 0 \rightarrow M-1 \\ j = 0 \rightarrow L-1 \end{cases} \tag{1}$$

$G_{i_{j:L}}(z)$ is the j–th L–th order polyphase component of the i-th synthesis filter $g_i[k]$. Now define :

$$\underbrace{\begin{bmatrix} Y_0^{k-1} \\ \vdots \\ Y_{M-1}^{k-1} \end{bmatrix}}_{\mathbf{Y}_{k-1}} = \underbrace{\begin{bmatrix} X[k] & \dots & X[k-L_f] \end{bmatrix}}_{\mathbf{X}_{k-1}} \underbrace{\begin{bmatrix} F_0^{k-1} \\ \vdots \\ F_{L_f}^{k-1} \end{bmatrix}}_{\mathbf{w}_{k-1}}, \tag{2}$$

in which

$$X[n] = \underbrace{\left.\begin{bmatrix} \chi_0[n] & \dots & 0 \\ \vdots & \ddots & \vdots \\ 0 & \dots & \chi_{M-1}[n] \end{bmatrix}\right\}}_{M} M(L_s+1) \tag{3}$$

$$\chi_m[n] = \begin{bmatrix} x_m[n-L_s] \\ \vdots \\ x_m[n] \end{bmatrix}, \tag{4}$$

$$F_l^{k-1} = \begin{bmatrix} f_0^{k-1}[l] \\ \vdots \\ f_{M-1}^{k-1}[l] \end{bmatrix}, \quad Y_m^{k-1} = \begin{bmatrix} y_m[k-L_s] \\ \vdots \\ y_m[k] \end{bmatrix} \tag{5}$$

$\mathbf{w}_{k-1}$ is the adaptive filter vector. L_f is the length of the adaptive filters (assuming that all filters are FIR), L_s is the length of the synthesis bank polyphase filters. Further define

$$\underbrace{\begin{bmatrix} e_0[k] \\ \vdots \\ e_{L-1}[k] \end{bmatrix}}_{\mathbf{E}_{k-1}} = \underbrace{\begin{bmatrix} S_{0_{0:L}} & \dots & S_{M-1_{0:L}} \\ \vdots & \ddots & \vdots \\ S_{0_{L-1:L}} & \dots & S_{M-1_{L-1:L}} \end{bmatrix}}_{\mathbf{S}^T} \underbrace{\begin{bmatrix} \epsilon_0^{k-1} \\ \vdots \\ \epsilon_{M-1}^{k-1} \end{bmatrix}}_{\mathbf{E}_{s_{k-1}}} \tag{6}$$

in which (see figure 4)

$$\underbrace{\begin{bmatrix} \epsilon_0^{k-1} \\ \vdots \\ \epsilon_{M-1}^{k-1} \end{bmatrix}}_{\mathbf{E}_{s_{k-1}}} = \underbrace{\begin{bmatrix} D_0^{k-1} \\ \vdots \\ D_{M-1}^{k-1} \end{bmatrix}}_{\mathbf{D}_{k-1}} - \underbrace{\begin{bmatrix} Y_0^{k-1} \\ \vdots \\ Y_{M-1}^{k-1} \end{bmatrix}}_{\mathbf{Y}_{k-1}}, \tag{7}$$

$$D_m^{k-1} = \begin{bmatrix} d_m[k-L_s] \\ \vdots \\ d_m[k] \end{bmatrix}, \quad \epsilon_m^{k-1} = \begin{bmatrix} \epsilon_m[k-L_s] \\ \vdots \\ \epsilon_m[k] \end{bmatrix}, \tag{8}$$

$$S_{m_{l:L}} = \underbrace{\begin{bmatrix} g_{m_{l:L}}[L_s] & \dots & g_{m_{l:L}}[0] \end{bmatrix}}_{L_s+1} \tag{9}$$

$\mathbf{E}_{k-1}$ is the fullband error vector, $\mathbf{E}_{s_{k-1}}$ contains the subband error signals, $\mathbf{S}$ represents the synthesis filter bank and

$g_{m_{l:L}}[k] = g_m[kL + l]$ is the l–th polyphase component of the m-th synthesis filter $g_m[k]$.
Optimal error suppression is obtained when Eq. 6 is as small as possible :

$$\min_{\mathbf{w}_k} ||\mathbf{E}_k||_2^2 \tag{10}$$

The optimal $\mathbf{w}$ is

$$\mathbf{w}_{\text{opt}} = \arg\min_{\mathbf{w}_k} ||\mathbf{S}^T\mathbf{D}_k - \mathbf{S}^T\mathbf{X}_k\mathbf{w}_k||_2^2. \tag{11}$$

In practice the optimal subband filters are estimated adaptively using a steepest descent algorithm. The gradient with respect to $\mathbf{w}_k$ is

$$\nabla\mathbf{w}_k = 2\mathbf{X}_k^H\mathbf{S}^*\mathbf{S}^T(\mathbf{X}_k\mathbf{w}_k - \mathbf{D}_k). \tag{12}$$

Therefore, $\mathbf{w}_{\text{opt}}$ can be found in an iterative way using fullband errors :

$$\mathbf{w}_{k+1} = \mathbf{w}_k - 2\mu\mathbf{X}_k^H\mathbf{S}^*\mathbf{S}^T(\mathbf{X}_k\mathbf{w}_k - \mathbf{D}_k). \tag{13}$$

In a standard subband adaptation scheme there is usually no compensation matrix $\mathbf{S}^*\mathbf{S}^T$ which transforms the subband errors to fullband errors.
$\mathbf{S}$ as defined in Eq. 6 is closely related to the synthesis bank polyphase matrix $\mathbf{G}(z)$. In case of a DFT modulated filter bank the synthesis bank polyphase matrix $\mathbf{G}(z)$ can be written as[1]

$$\mathbf{F}^{-T}\mathbf{C}^T(z)J. \tag{14}$$

Frequency-domain adaptive filters such as the PBFDAF can be considered as a special case of DFT modulated subband systems (see figure 3). For PBFDAF [6]

$$\mathbf{C}(z) = \underbrace{\left[\begin{array}{cccc}\mathbf{I}_L & \mathbf{0}_L & \dots & \mathbf{0}_L \quad \mathbf{0}\end{array}\right]}_{M} = \mathbf{C} \tag{15}$$

is independent of z and hence also $\mathbf{G}(z)$ is independent of z. This means that $L_s = 0$ for PBFDAF. The PBFDAF is indeed a subband system with very 'poor' filter banks (modulated sinc frequency responses). $\mathbf{S}$ as defined in Eq. 6 then equals $\mathbf{G}(z) = \mathbf{S} = \mathbf{F}^{-T}\mathbf{C}^T J$. For the PBFDAF the update equation 13 comes down to

$$\begin{aligned}\mathbf{w}_{k+1} &= \mathbf{w}_k + 2\frac{\mu}{M}\mathbf{X}_k^H\mathbf{F}\mathbf{C}^T\mathbf{C}\mathbf{F}^{-1}(\mathbf{D}_k - \mathbf{X}_k\mathbf{w}_k) &(16)\\ &= \mathbf{w}_k + 2\frac{\mu}{M}\mathbf{X}_k^H\mathbf{F}\left[\begin{array}{cc}\mathbf{I}_L & \mathbf{0}\\ \mathbf{0} & \mathbf{0}_{M-L}\end{array}\right]\mathbf{F}^{-1}\mathbf{E}_{s_k} &(17)\end{aligned}$$

$\mathbf{E}_{s_k}$ are the subband errors, $\mathbf{F}\mathbf{C}^T\mathbf{C}\mathbf{F}^{-1}$ does the error correction. It can be proven that the weight update equation of a so–called unconstrained PBFDAF [6][5] corresponds to equation 17, derived here based on the fullband error feedback. Therefore, it is hoped for that employing the fullband adaptation scheme for subband schemes will lead to improved performance.

[1] J is the anti–diagonal matrix.

4. CONCLUSIONS

In this paper, we formulated design criteria for subband adaptive systems. An adaptation scheme was presented that feeds back the fullband error. It can be considered as a generalisation and extension of frequency–domain alias–compensation techniques to subband adaptive systems.

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6. REFERENCES

[1] J. Shynk, "Frequency-Domain and Multirate Adaptive Filtering," *IEEE Signal Processing Magazine*, pp. 15–37, January 1992.

[2] Z. Cvetković, "Oversampled Modulated Filter Banks and Tight Gabor Frames in $l^2(\mathbb{Z})$," in *Proceedings of the 1995 IEEE Int. Conf. on Acoust., Speech and Signal Processing*, (Detroit, Michigan, USA), pp. 1456–1459, May 1995.

[3] A. Gilloire and M. Vetterli, "Adaptive Filtering in Subbands with Critical Sampling : Analysis, Experiments and Application to Acoustic Echo Cancellation," *IEEE Trans. Signal Processing*, vol. 40, pp. 1862–1875, August 1992.

[4] K. Eneman and M. Moonen, "Real-Time Implementation of an Acoustic Echo Canceller on DSP," in *Proceedings of the ProRISC/IEEE Benelux workshop on Circuits, Systems and Signal Processing*, (Mierlo, The Netherlands), pp. 163–170, November 1997.

[5] J. Páez Borrallo and M. García Otero, "On the implementation of a partitioned block frequency domain adaptive filter (PBFDAF) for long acoustic echo cancellation," *Signal Processing*, vol. 27, pp. 301–315, June 1992.

[6] K. Eneman and M. Moonen, "Filter Bank Constraints for Subband and Frequency-Domain Adaptive Filters," in *Proceedings of the 1997 IEEE Workshop on Applications of Signal Processing to Audio and Acoustics*, (New Paltz, New York, USA), October 19-22 1997.