A Generalised Sidelobe Canceller Architecture Based on Oversampled Subband Decompositions

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Abstract. Adaptive broadband beamforming can be performed in oversampled subband signals, whereby an independent beamformer is operated in each frequency band. This has been shown to result in a considerably reduced computational complexity. In this paper, we primarily investigate the convergence behaviour of the generalised sidelobe canceller (GSC) based on normalised least mean squares algorithm (NLMS) when operated in subbands. The minimum mean squared error can be limited, amongst other factors, by the aliasing present in the subbands. With regard to convergence speed, there is strong indication that the subband-GSC converges faster than a fullband counterpart of similar modelling capabilities. Simulations are presented.

1 Introduction

Broadband beamforming is a technique used for numerous applications ranging from sonar to mobile communications. Generally, the task is to focus a sensor array onto a signal impinging from a certain direction, while interfering signals from other spatial angles of incident are suppressed [3, 9], as indicated in Fig. 1. To perform beamforming at high resolution, generally beamforming filters with a considerable temporal length are required to accurately match fractional delays [6] which are necessary to align the differently delayed signals in the BF for constructive or destructive interference. To reduced the resulting computational complexity associated with an adaptive broadband beamformer, a promising approach is to apply subband decompositions [13], whereby the structure given in Fig. 3 has been proposed.

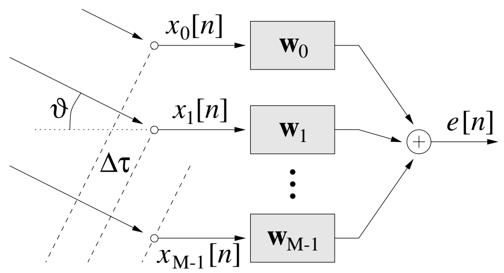


Figure 1. Broadband beamformer.

The subband adaptive beamformer developed in [13] relies on an underlying generalised sidelobe canceller (GSC) architecture. If the explicit signal of interest impinging on the array is unavailable for training but spatial information (angle of incident, direction of jammers, etc.) accessible, a linearly constrained minimum variance (LCMV) beamformer is a popular choice [9]. Amongst the LCMVs, the GSC translates the constrained problem defined by the spatial information into an unconstrained problem of lower order, which can be solved for by standard adaptive algorithms [5].

While the problem of constraint projection into the subbands and the reduction of computational complexity by the subband approach have been addressed in [13], this paper is mainly concerned with investigating the convergence rate of the proposed subband adaptive beamformer. For this we address two different scenarios, whereby a filter bank with increased bandwidth for the analysis filters is compared to both standard paraunitary oversampled filter banks, and a fullband implementation. This is motivated by other subband adaptive filter applications such as acoustic echo control, where slow modes of convergence arising at band-edges with low spectral energy are overcome by modified analysis filter banks [7].

In Sec. 2, we will briefly review the subband adaptive beamformer (SAFB) structure of [13], and the different filter banks used for subband decompositions. The convergence of the SAFB is discussed in Sec. 3 and illustrated by simulations and results in Sec. 4.

2 Subband Adaptive Beamforming

The proposed subband adaptive beamformer (SAB) structure decomposes each sensor signal $x_m[n]$ by means of an analysis filter bank, and applies an independent beamformer to each subband, as shown in Fig. 3 [13]. In this case, the subband beamforming algorithms are GSCs as given in Fig. 2, but could as well be replaced by other LCMV beamformers. The outputs of the beamformers working in different frequency bands are recombined to a fullband output by a synthesis filter bank.

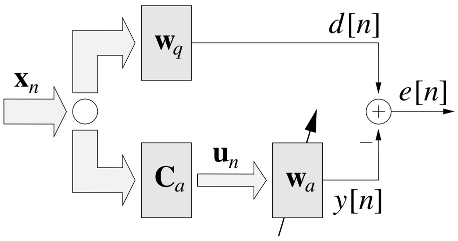
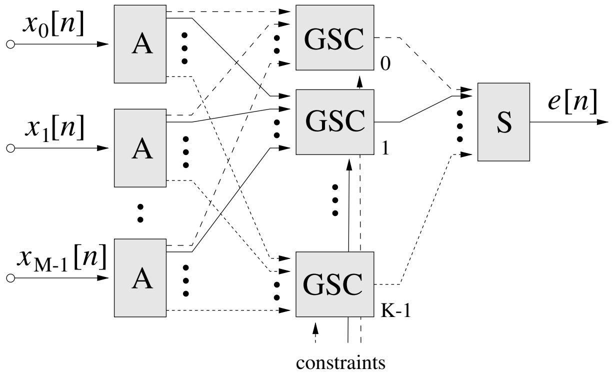


Figure 2. (top) generalized sidelobe canceller.

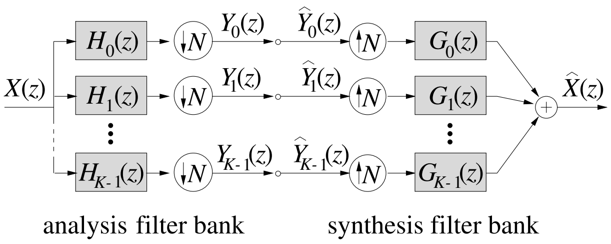
Figure 3. (right) subband adaptive beamformer.



The filter banks performing the subband decomposition into K subbands decimated by $N < K$ are depicted in Fig. 4. We consider analysis and synthesis filters which are derived from a prototype lowpass filter $p[n]$ such that the overall system in Fig. 4 is near perfectly reconstructing. This allows for two important cases: (i) the filter banks are paraunitary, i.e. analysis and synthesis filter bank are both derived from the same prototype; (ii) analysis and synthesis filter banks are derived from two different prototype filters. While the latter choice is the general case of a frame decomposition [8], a paraunitary filter bank based on a single prototype implements a tight frame expansion [1]. Such an expansion guarantees energy conservation for projections between subband and fullband domain, which offers advantages when e.g. assessing steady-state limitations of subband adaptive systems [14].

An example for a frame filter bank with a wider analysis prototype filter is given in Fig. 5. While the standard prototype filter ($P(e^{j\Omega}) = H(e^{j\Omega}) = G(e^{j\Omega})$ has a stopband starting at π/N, where N is the maximum decimation factor, the frame analysis prototype has a higher magnitude response in the transition band $\Omega \in [\pi/K; \pi/N]$, which avoids guard band gaps in the decimated subband spectrum, and permits faster convergence if signal components are lying within this spectral region [2]. Note however that a moderate level of aliasing is produced, which will be filtered out in the synthesis filter bank due to the sharper synthesis prototype filter in Fig. 5.

In either case of filter banks, the SAFB constraints can be determined by optimal subband projections derived in [12] for the general frame filter bank case. After adaptation of the SABs, an equivalent fullband beamformer can be calculated by exciting in turn each sensor signal $x_m[n]$ by an impulse while feeding zeros to all other inputs. The impulse response observed at the reconstructed beamformer output $e[n]$ then represents the equivalent filter $\mathbf{w}_m$ in Fig. 1.



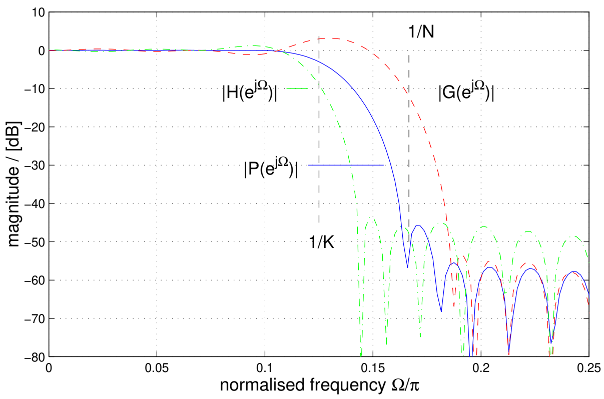


Figure 4. (top) filter bank system for decomposition into K subbands decimated by $N \leq K$.

Figure 5. (right) prototype lowpass filter characteristics for frame and tight frame filter banks with $K = 8$, $N = 6$; the frame decomposition permits wider analysis filters, which is compensated by the synthesis prototype.

3 Subband Adaptive Beamformer Convergence

In comparable subband adaptive filter applications, the minimum mean-squared error (MMSE) is limited by the amount of aliasing produced in the subband signals [10]. Although for the exact MMSE limit, knowledge of the input signal and the transmission path characteristics is required, a good approximation for the amount by which the error power can be reduced in the adaptation process is given by the stopband energy of the analysis filters. If the filter banks are derived from a prototype filter by modulation, the MMSE due to aliasing can be formulated in terms of the prototype filter, $P(e^{j\Omega})$,

$$\xi_{\mathrm{MMSE}} \propto \int_{\pi/N}^{\pi} P(e^{j\Omega}) d\Omega \tag{3.1}$$

where the stopband edge is defined by the decimation ratio N. Additionally, here the number of sensors M (and therefore analysis filter bank operations) contributes to the amount of aliasing brought into the adaptive process.

The convergence speed of LMS-type adaptive filters is proportional to the eigenvalue spread of the input signal to the GSC [5]. Due to Fig. 2, the input signal, $\mathbf{x}_n$, prior to the projection with the blocking matrix $\mathbf{C}_a$ is a concatenation of all tap delay lines of the beamforming filters in Fig. 1. Therefore, both temporal and spatial correlation will increase the eigenvalue spread and therefore the required convergence time. Although the subband approach performs an implicit prewhitening of the sensor signals, this only affects the temporal, but not the spatial dimension. Additionally, for a white input to a sensor, the analysis filters introduces a spectral gap, and therefore a correlation in the subband domain, which has been recognised to cause slow modes of adaptation [7]. Although the wider analysis filters discussed in Sec. 2 can bring some benefit here, the question arises whether the subband approach can bring any benefit at all. For this reason, we are investigating and comparing subband approaches with different types of analysis and synthesis filter banks in the following section.

4 Simulations and Results

The proposed subband GSC scheme is demonstrated for a beamformer with $M = 12$ sensors and 180 coefficients in temporal dimension (fullband). The GSC is constrained to receive a signal of interest from broadside ($\vartheta = 0°$), which is white Gaussian with unit variance. The BF should adaptively suppress an interferer consisting of several narrowband signals from $\vartheta = 20°$ at around -40 dB SINR and iid Gaussian additive noise at 10dB SINR using a normalised LMS algorithm [5]. The standard fullband adaptive BF is compared to two subband adaptive BFs operating in $K = 8$ frequency bands decimated by $N = 6$ with a temporal dimension of $180/N = 30$ filter coefficients. The filter banks are derived from prototype lowpass filters shown in Fig. 5, i.e. one SABF employs a paraunitary, and the second a frame filter bank with wider analysis filters. The resulting residual error MSE curves are given in Fig. 6.

In a first simulation, the narrowband interferers are placed in the centre of the subband signals, such that

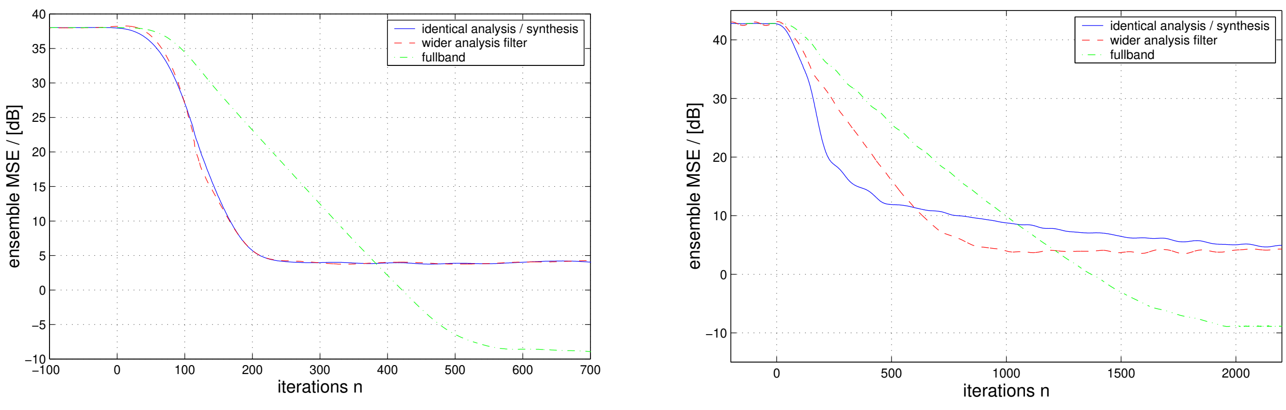


Figure 6. fullband and subband GSC simulations with narrowband signals centre with respect to (left) subband centre frequencies and (right) band edges.

no slow modes of adaptation are excited. The results in Fig. 6(left) indicates that based on identical modelling capabilities and convergence factors for the NLMS algorithm, the subband solution adapts faster than its fullband counterpart. Secondly, the frame based system with wider analysis filters does not give any benefit, as no signal components are incurred in the transition bands of the analysis filters.

A second simulation employs narrowband signals which are centred at the band-edges of the subbands with convergence results displayed in Fig. 6(right). Here again the subband architectures show a faster convergence than the fullband system. However, the subband GSC based on paraunitary filter banks exhibits slower modes of adaptation starting 200 and 500 iterations after adaptation is started ($n = 0$). The subband system based on wider analysis filters has, although initially slower, a uniform convergence, which reaches the steady-state MSE considerable faster than the other two architectures.

It is also noticeable from Fig. 6, that the steady-state MSE in case of the subband implementation is considerably worse than for the fullband. This is due to the aliasing caused in the subband domain, which here is limiting the performance, while the fullband implementation's MSE is limited by the additive Gaussian noise.

5 Conclusions

This paper has addressed convergence issues of a subband adaptive beamformer (SABF) based on LMS-type algorithms for the generalised sidelobe canceller. The structure has been briefly reviewed, with particular attention to the possibility of using different types of filter banks, namely paraunitary and frame filter banks, whereby the latter offer faster convergence for convergence modes in the transient filter bank regions. While this increase in convergence speed has been noted for different subband adaptive filter applications, the use with an SABF is limited, as the prewhitening due to the filter banks has no effect on the spatial dimension of the beamformer. Nevertheless, simulation results indicate that in general the subband based LMS-type GSC can outperform a fullband implementation with respect to convergence rate. Wider analysis filters of frame filter banks can lead to a more uniform and generally faster convergence, while the adaptation of the standard SABF with paraunitary filter banks may perform better in the initial phase of adaptation.

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