ROBUST MICROPHONE ARRAYS USING SUBBAND ADAPTIVE FILTERS

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ABSTRACT

A new adaptive beamformer which combines the idea of subband processing and the generalized sidelobe canceller structure is presented. The proposed subband beamformer has a blocking matrix that uses coefficient-constrained subband adaptive filters to limit target cancellation within an allowable range of direction of arrival. Simulations compare the fullband adaptive beamformer and the subband adaptive beamformer show that the subband beamformer has better performance than the fullband beamformer when the input signals to the microphone array are coloured. In reverberant environments, also, the proposed subband beamformer performs better than its fullband counterpart.

1. INTRODUCTION

Adaptive arrays have been widely studied for teleconferencing, speech recognition, speech enhancement, and hearing aids. Using a small number of sensors, they are able to achieve high interference reduction by introducing nulls in the direction of arrival (DOA) of the interferences. However, adaptive arrays that are based on Griffiths and Jim sidelobe canceller (SC) [1] or generalized SC (GSC) structure, are sensitive to target signal leakage and cancellation in the presence of steering-vector errors and reverberations. Works which resolve the above problems to a limited extent are reported in [3, 4, 5, 6]. Most of these methods face difficulties such as the need of increasing the number of microphones to achieve good interference reduction performance, loss of degrees of freedom for interference reduction, and mistracking. Hoshuyama *et. at.* [2] have recently proposed a robust adaptive beamformer which avoids most of these difficulties by introducing an adaptive blocking matrix in the GSC structure. However, for wideband signals, its performance deteriorates in the presence of coloured lowpass interference signals.

Adaptive arrays are commonly classified under two groups; narrowband and wideband arrays. In the narrowband arrays, each interfering signal is approximated by a tone. Cancellation of each tone in array is possible through two degrees of freedom, i.e., two adjustable coefficients. Thus, a set of N narrowband interfering signals may be rejected with only $2N$ adjustable coefficients [7]. Dealing with wideband signals in adaptive arrays, such as speech signals in microphone arrays, on the other hand, is much more complicated. A wideband signal may be thought as a summation of a large number of tones. With this argument, cancellation of a wideband interference requires a large number of adjustable coefficients. Furthermore, adaptation of these coefficients also may result in some practical problem when the underlying signals are coloured. This particularly will become a challenging problem when one has to deal with speech signals, since such signals are highly coloured and non-stationary.

An elegant solution, which to a great extent simplifies the problem of wideband adaptive arrays, is to perform all the processings in subbands. The partitioning of the signals in subbands will effectively convert a wideband signal to a number of narrower band signals, thus a more effective processing will become possible. There are limited reports which address this solution in the recent literature [8, 9].

In this paper, a new wideband adaptive arrays structure which combines the idea of subband processing and the robust GSC structure of Hoshuyama *et. at.* [2] is proposed. We work out and present the details of a number of measures need to be taken for effective implementation of the proposed subband method. Computer simulations show that the proposed structure works better than its fullband counterpart.

2. THE PROPOSED SUBBAND ADAPTIVE BEAMFORMER

The structure of the proposed subband adaptive beamformer is shown in Fig. 1. The beamformer uses a GSC structure. Each microphone input signal is first separated into M subband signals as it passes through an analysis filter bank. The subband signals, which are decimated by a factor of L, are then fed into the fixed beamformer and the adaptive blocking matrix. The fixed beamformer is implemented simply by averaging the signals from similar subbands to obtain a set of M subband desired signals

$$D_m(k) = \frac{1}{R}\sum_{r=0}^{R-1} X_{r,m}(k) \tag{1}$$

where R is the number of microphones, $D_m(k)$ is the desired signal in mth subband, $X_{r,m}(k)$ is the mth subband signal of the rth microphone signal.

The blocking matrix is similarly divided into M parts, each serving its respective subband. Each subband has R coefficient-constrained adaptive filters and uses the corresponding desired subband signal as the common reference input. The coefficients of adaptive filters in each subband are bounded by a set of constraints that is designed with respect to the pre-specified allowable target direction range and the respective frequency bands. A modified NLMS algorithm that includes the constraints updates the coefficients of the adaptive filters in each subband. The algorithm is described by the following equations:

$$h'_{r,m,p} = h_{r,m,p}(k) + \frac{\alpha}{\mathbf{D}_m^T(k)\mathbf{D}_m(k)} Y_{r,m}(k) D_m(k-p) \quad (2)$$

where

$$\mathbf{D}_m(k) = [D_m(k), D_m(k-1), \ldots, D_m(k-P+1)]^T, \quad (3)$$

$$h_{r,m,p}(k+1) = \begin{cases} \phi_{r,m,p} & \text{for } h'_{r,m,p} > \phi_{r,m,p} \\ \psi_{r,m,p} & \text{for } h'_{r,m,p} > \psi_{r,m,p} \\ h'_{r,m,p} & \text{otherwise} \end{cases} \quad (4)$$

for $r = 0, 1, ..., R-1$, $p = 0, 1, ..., P-1$, and $m = 0, 1, ..., M-1$, and ϕ's and ψ's are a set of constraint.

The output of the blocking matrix is then used as reference input to the adaptive noise canceller (ANC) filter. The ANC filter in each of the subband is updated by a modified NLMS algorithm with a different norm constraint. The algorithm is described as

$$\mathbf{w}'_{r,m} = \mathbf{w}_{r,m}(k) + \frac{\beta}{\sum_{j=0}^{R-1} \mathbf{Y}_{j,m}^T(k)\mathbf{Y}_{j,m}(k)} Z(k)\mathbf{Y}_{r,m}(k) \quad (5)$$

$$\Omega_m = \sum_{r=0}^{R-1} \mathbf{w}'^T_{r,m}\mathbf{w}'_{r,m} \quad (6)$$

$$\mathbf{w}_{r,m}(k+1) = \begin{cases} \sqrt{\frac{C_m}{\Omega_m}}\mathbf{w}'_{r,m} & \text{for } \Omega_m > C_m \\ \mathbf{w}'_{r,m} & \text{otherwise.} \end{cases} \quad (7)$$

where β is the step size parameter,

$$\mathbf{w}_{r,m}(k) = [w_{r,m,0}(k), w_{r,m,1}(k), \ldots, w_{r,m,Q-1}(k)]^T, \quad (8)$$

$$\mathbf{Y}_{r,m}(k) = [Y_{r,m}(k), Y_{r,m}(k-1), \ldots, Y_{r,m}(k-Q+1)]^T, \quad (9)$$

and C_m's are a set of norm-constraining constants.

As suggested in [2], the adaptations of the ANC and blocking matrix filters are only performed when signal-to-interference ratio (SIR) is low and high, respectively. The method of identifying the periods of high SIR and periods of low SIR is not trivial and is not discussed here, because of space limitation.

3. COMPUTER SIMULATIONS

We consider a 4-channel microphone array for our simulations. The inter-microphone distance is 4 cm. The sampling frequency is 8 kHz. The fixed beamformer is a simple summer. The maximum allowable target direction error is set 20^o.

The input signal to each microphone is generated by convolving the target interference signals with their respective source-to-microphone impulse responses. For the purpose of simulations, such impulse responses are generated by using the "lowpass impulse" method of Peterson [10]. The duration of each impulse response is 128 samples.

For the fullband realization, the following parameters are used: $\Delta_1 = 5$, length of each adaptive filter in blocking matrix is 16, $\alpha = 0.1$, $\Delta_2 = 10$, length of each adaptive filter in ANC is 64, $\beta = 0.2$.

For the subband realization, on the other hand, we choose: $M = 8$, $L = 4$, $\Delta_1 = 5$, length of each subband adaptive filter in blocking matrix, $P = 16$, $\alpha = 0.1$, $\Delta_2 = 10$, length of each subband adaptive filter in adaptive noise canceller, $Q = 16$, $\beta = 0.2$.

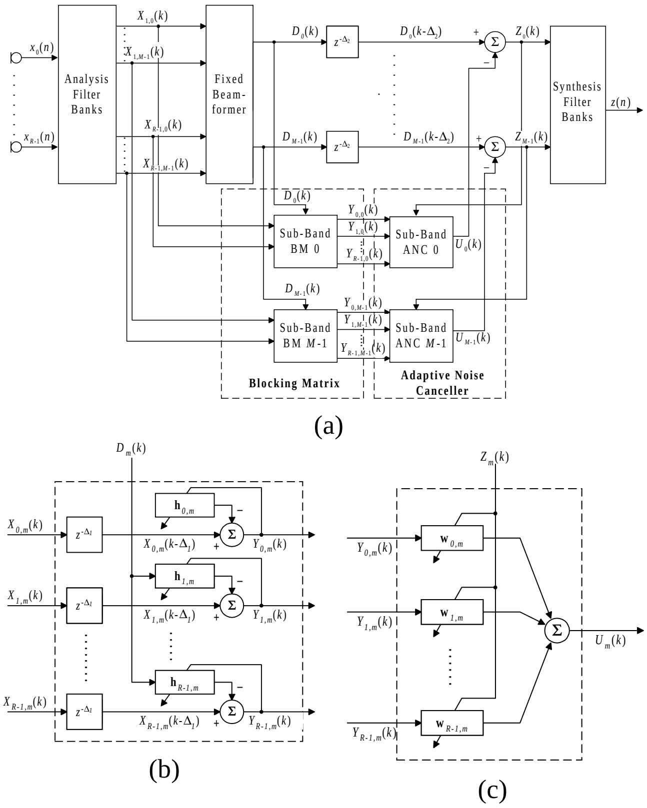


Fig. 1. (a) Structure of the proposed subband beamformer, (b) structure of the mth subband blocking matrix, (c) structure of the mth subband adaptive noise canceller.

The adaptation is carried out in the following manner: the blocking matrix adaptive filters are adapted for 50,000 iterations. The ANC filters are then adapted for the next 150 000 iterations. The normalized output power after convergence is obtained by normalizing the output power averaged over the last 20,000 iterations by the power of the assumed target direction.

3.1. Threshold Vectors

Fig. 2 shows the total squared-norm, Ω_m, as a function of DOA when the updating NLMS algorithm is not norm-constrained. The Ω_m's in the lower subbands are significantly higher than the Ω_m's in the higher subbands. In order to effectively restrain the excess growth of the tap coefficients of the adaptive filters, it is necessary to have a different threshold for each subband. The threshold values, C_m's, in the lower subbands should have larger values than the threshold values in the higher subbands.

Four subband threshold vectors are tested in an anechoic environment:

$$\mathbf{C}_1 = [60, 34, 19, 11, 6, 3, 2, 1]^T; \quad (10)$$

$$\mathbf{C}_2 = [128, 64, 32, 16, 8, 4, 2, 1]^T, \quad (11)$$

$$\mathbf{C}_3 = [600, 250, 100, 40, 15, 6, 3, 1]^T, \quad (12)$$

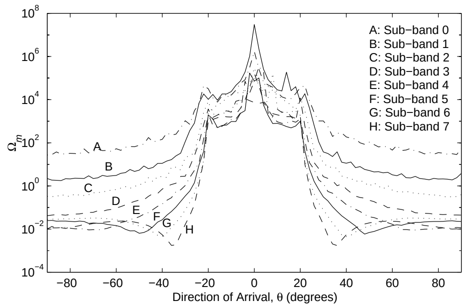


Fig. 2. Total squared-norm of $\mathbf{w}_{r,m}$, Ω_m, when the updating algorithm is not norm-constrained.

$$\mathbf{C}_4 = [1200, 430, 160, 60, 20, 8, 3, 1]^T. \tag{13}$$

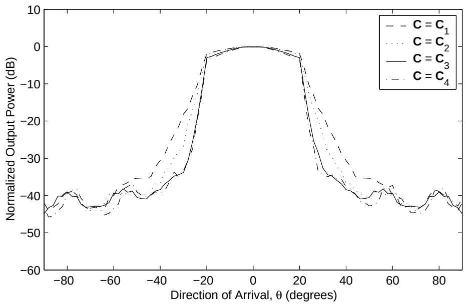


Fig. 3. Normalized output power as a function of DOA, for four threshold vectors.

The subband thresholds in the lower subbands are much higher than the thresholds in the upper subbands and they decrease exponentially. Fig. 3 shows the normalized output power as a function of DOA for four norm-constraint threshold vectors. From Fig. 3, we can see that the performance of the norm-constraint adaptive noise canceller is dependent on the threshold values. Threshold vectors with higher threshold values, $\mathbf{C}_3$ and $\mathbf{C}_4$, give better spatial selectivity performances, *i.e.,* the beamformer is able to cancel out most effectively the interferences arriving from directions that are near but outside the allowable target direction range, while keeping the desired target signals that are within the allowable range. The beamformer with the smallest threshold vector, $\mathbf{C}_1$, has a more gradual interference reduction performance. However, the disadvantage of a higher threshold vector is that the signals within the allowable target direction range are also attenuated. For example, for the signals at the maximum allowable target direction, $\pm 20^o$, the ANC's with $\mathbf{C}_1$, $\mathbf{C}_2$, $\mathbf{C}_3$ and $\mathbf{C}_4$ are attenuated by approximately 0.5 dB, 0.5 dB, 1.4 dB and 2.5 dB, respectively.

3.2. Comparison Between Fullband and Subband Beamformers

The performance of the proposed subband adaptive beamformer is compared with the performance of the fullband adaptive beamformer. The subband adaptive beamformer used the threshold vector, $\mathbf{C}_3$.

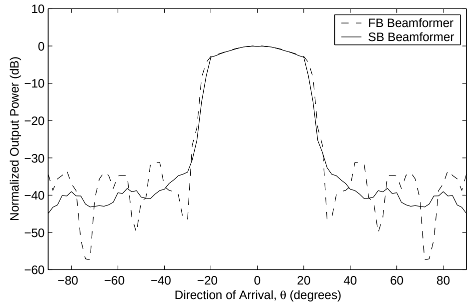


Fig. 4. Normalized output powers of the fullband beamformer and the subband beamformer for a band-limited white Gaussian input signal.

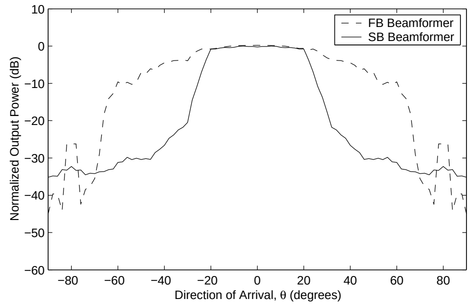


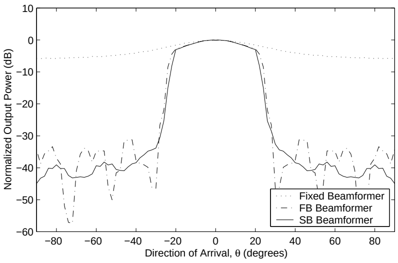
Fig. 5. Normalized output powers of the fullband beamformer and the subband beamformer for a lowpass input signal.

Fig. 4 shows the normalized output power as a function of DOA, for a band-limited flat spectrum (0.4 - 3.6 kHz) Gaussian signal. The results for both fullband and subband beamformers are similar. The subband beamformer has only performed slightly better at the DOA near the maximum allowable direction, 20^o.

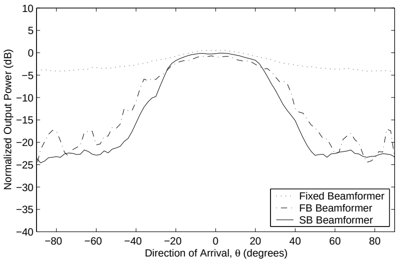
Fig. 5 compares the performances of the fullband beamformer and the subband beamformer for a lowpass input signal. In this case, the fullband adaptive beamformer performs poorly. The signals whose DOA's are outside the allowable target direction range are not reduced significantly. For example, only the signals arriving from directions beyond 44^o are able to achieve a reduction of more than 6 dB. This problem is caused by the frequency-dependent signal blocking capability of the blocking matrix. The low frequency components of the signal are highly correlated and are easily cancelled at the blocking matrix output. Thus the ANC could not effectively cancel the correlated low-frequency components in the primary input. This difficulty of the fullband beamformer has also been noted in [2]

3.3. Effects of Reverberations

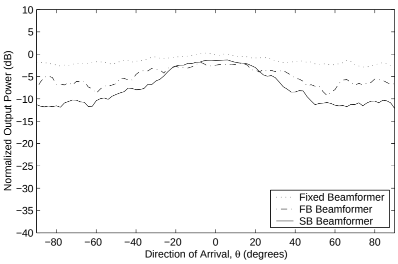
The "lowpass impulse" method [10] is used to model the reverberant environments. The simulated reverberant room is 5 x 6 x 3 m^3. The center of the microphone array is positioned at coordinates (2.4, 2.3, 1.6). The microphone array is tilted at 45^o to the right. The target/interference source is positioned at 1 m away



(a)



(b)



(c)

Fig. 6. Normalized output powers after convergence in a reverberant environment for (a)$\Gamma = 1.0$, (b)$\Gamma = 0.6$, (c) $\Gamma = 0.2$.

from the array center. The wall absorption coefficients, Γ, are varied to simulate different reverberation times, *i.e.*, different ratios of direct path noise and diffuse noise. The absorption coefficients are chosen to be equal for all six walls.

The effects of the reverberations on the performances of the proposed subband beamformer and the fullband beamformer are compared in Fig. 6 for absorption coefficients equal to 1.0 (anechoic), 0.6 and 0.2. The plot in Fig. 4 for the anechoic environment is reproduced in Fig. 6 for comparison. The input signal used is the band-limited signal. We see that as the absorption coefficient is decreased, the performances of both beamformers deteriorate. In the regions within the allowable range, the attenuation at the subband beamformer output is less than the attenuation at the fullband beamformer output. On the other hand, in the regions outside the allowable range, although the effectiveness of the beamformers at interference reduction is weakened, the subband beamformer still outperforms the fullband beamformer.

4. CONCLUSIONS

A subband adaptive beamformer applicable to microphone array has been presented. The subband adaptive beamformer has a blocking matrix that uses coefficient-constrained subband adaptive filters and an adaptive noise canceller that uses norm-constrained subband adaptive filters. The subband beamformer has been shown to have better performance than the fullband beamformer when the input signals to the microphone array are lowpass. Simulations on the effects of reverberations have shown that the proposed subband beamformer is able to suppress interference arriving from direction outside the allowable target direction range more than the fullband beamformer, while reducing the attenuation of the desired signal arriving from direction within the allowable range.

5. REFERENCES

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