A NEW SUBBAND STRUCTURE FOR AN ACOUSTIC BEAMFORMER WITH LEAKY ADAPTIVE FILTERS IN THE BLOCKING MATRICES

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ABSTRACT

In this paper, a new subband adaptive beamformer is proposed that is specifically applicable to microphone arrays that can allow up to $\pm 30^{\circ}$ target Direction of Arrival (DOA) range. The proposed subband beamformer contains a Blocking Matrix (BM) that uses leaky subband adaptive filters to resolve the problem of the signal blocking capability in the BM when the input signals to the microphone array are highly colored. Simulation results show that this subband proposed beamformer is able to give a better interference suppression than that the equivalent fullband beamformer.

1. INTRODUCTION

Adaptive microphone arrays have been widely used in teleconferencing, speech recognition, speech enhancement and hearing aids [1-5]. Adaptive beamformers such as the Griffiths-Jim beamformer or the Generalized Sidelobe Canceller (GSC) are probably the best known [6]. However, target signal cancellation can still occur in the presence of steering vector errors, and several signal processing techniques have been developed to mitigate this effect. Unfortunately, problems still exist such as loss the degrees of freedom of the target signal and the need for a large number of microphones. Hoshuyama et. al. [4] have proposed a robust adaptive beamformer which allows as large as $\pm 20^{\circ}$ target Direction of Arrival (DOA) range with just a small number of microphones (and this resolves most of the aforementioned difficulties). Recently, the authors of this paper proposed a new Leaky Adaptive Constrained Beamformer (LACB), that can achieve up to $\pm 30^{\circ}$ target DOA range [1]. It uses a three stage adaptive signal processing approach: the first stage is the LACB to extract the target signal from a wide specified range; the second stage is the Blocking Matrix (BM) with unconstrained adaptive filters to block the target signal and pass the interference; and finally, the Multiple Canceller (MC) with Norm-Constrained Adaptive Filters (NCAF) tries to cancel the interference while preventing the target signal cancellation.

However, the degree of interference suppression near/outside the target DOA region in the LACB of [1] may not be enough, and so in [2], we addressed this problem by introducing Leaky Adaptive Filters (LAFs) into the BM. The principle of the LAF is that it converges (in theory) to a sub-optimal (Wiener) solution. This ensures that a small target signal and a large noise power leak into the MC inputs, and this will cause the noise components to be more effectively cancelled by the NCAF in the MC. This results in at least 5 dB improvement over the beamformer in [1], even when the Signal-to-Noise Ratio (SNR) is as low as -5 dB. We will henceforth refer to this beamformer in [2] as the *fullband beamformer*. But one problem with [2] still remains, when we are dealing with signals (i.e. speech signals) that are highly colored. We may want to increase the interference suppression near/outside the allowable target DOA region, which is limited as a consequence of the frequency-dependent signal blocking capability of the BM - low frequency components of the colored signal are easily cancelled by the BM. So the MC is unable to cancel the low frequency components in the primary input effectively. This problem has been noted in [4,5].

So in this paper, our proposed solution is to restructure the fullband beamformer of [2], and perform the adaptive processing of the BM and MC structures in subbands. The purpose is to partition the underlying signals into subbands, and following decimation, the subband signals will be approximately spectrally flat, thus increasing the convergence speed. Having a white input will also resolve the problem of the signal blocking capability in the BM. In addition, subband beamforming also has the advantages of faster convergence and low sampling rate complexity when compared with a fullband beamformer [3]. We will see later (through simulation results) that our proposed method is able to give a better interference reduction performance than that the fullband beamformer in [2].

2. PROPOSED SUBBAND BEAMFORMER

The structure of the proposed subband beamformer is shown in Fig. 1. As mentioned before, we are only performing the subband adaptive processing of the BM and MC structures and not the LACB. We designed the low group delay, cosine modulated filterbank, via the technique recently proposed in [7] and all the M subbands have the same bandwidth. The purpose

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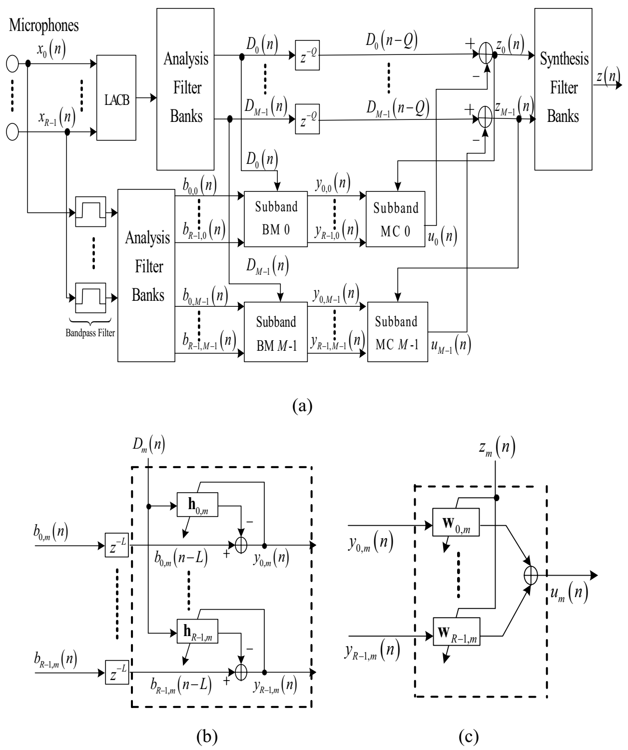

Fig. 1. (a) Structure of the proposed subband beamformer - the structure of the LACB is the same as in [1]; (b) structure of the *m*th subband BM; and (c) structure of the *m*th subband MC.

of the analysis filter bank is to partition the signal spectra into M subband signals. Computations are then performed in parallel at a lower rate. This is then followed by synthesis filter bank for reconstruction (see Fig. 1). In this case, the subband signals are decimated by a factor of $L=M$. As described in [2], the purpose of the LAF in the BM is to prevent the cancellation of the interference that arrives from outside the target DOA region, and so the target signal is guaranteed to be minimized in the limited target DOA region. As shown in Fig. 1, the output signal of the LACB is first divided into M subbands, and these serve as the input signals of the LAFs in each subband BM. The outputs of the LAFs in each subband BM are subtracted from the output delayed signal, $b_{r,m}(n-L)$, where the purpose of the imposed delay L is for causality. Using the leaky Normalized Least-Mean-Square (NLMS) algorithm in [2,8], the signal processing in the LAFs in each subband BM can now be described as

$$y_{r,m}(n) = b_{r,m}(n-L) - \mathbf{h}^{\mathrm{T}}_{r,m}(n)\mathbf{D}_m(n) \tag{1}$$

$$\mathbf{h}_{r,m}(n+1) = \mathbf{h}_{r,m}(n) - \beta\mathbf{h}_{r,m}(n) + \mu\frac{y_{r,m}(n)}{\left\|\mathbf{D}_m(n)\right\|^2}\mathbf{D}_m(n) \tag{2}$$

where $\beta = \mu\gamma, 0 \le \beta \le 1, \gamma \ge 0$ is the leakage factor; $0 \le \mu \le 2$ is the step-size parameter; and

$$\mathbf{h}_{r,m}(n) = \left[h_{r,m,0}(n), h_{r,m,1}(n), \ldots, h_{r,m,P_1-1}(n)\right]^{\mathrm{T}} \tag{3}$$

$$\mathbf{D}_m(n) = \left[d_{r,m}(n), d_{r,m}(n-1), \ldots, d_{r,m}(n-P_1+1)\right]^{\mathrm{T}} \tag{4}$$

$r = 0,1,\ldots,R-1$ refers to each microphone, $m = 0,1,\ldots,M-1$ refers to each subband and P_1 is the number of the taps of the LAFs in each subband BM. Note that in (2), the $\beta\mathbf{h}_{r,m}(n)$ term is to restrain the excess growth of the tap weight vector, so if we select a large value of β, the target signal power at the outputs of the BM will increase, and so target signal cancellation will occur in the MC. Alternately, if we select a small value of β (i.e. β is close to zero), then minimization of the target signal is guaranteed while the performance of the interference reduction will deteriorate in the BM. This is because, in a real-time environment, the minimization of the interference signal at the output of the BM is inevitable. This means that the LAFs in each subband BM are able to minimize the target signal while maintaining the noise power level as high as possible at the BM output, so that the correlated noise component is easily cancelled by the NCAFs in each subband MC. As described in [5], the purpose of the NCAFs in each subband MC is to prevent the excess growth of the tap-weight vector, and it is necessary to have a different threshold for the tap-weight vector for each subband. The tap-weight vector thresholds in the lower subbands must be higher than those thresholds in the upper subbands, and so we allow these thresholds to decrease exponentially. As mentioned before, for a fullband beamformer, the low frequency components are easily cancelled by the BM, and so the MC is subsequently unable to cancel the low frequency components effectively. However, by using the LAFs in each subband BM, the cancellation of the low frequency components in the BM will deteriorate. So by permitting larger thresholds in the lower subbands, the NCAFs in each subband MC are able to cancel the correlated components in the primary input more effectively. Finally, since the signal processing in each subband MC has also been developed in [5], then because of space limitation there will be no further discussion here, except to say that we have changed the notation of [5].

3. SIMULATIONS

Three simulations will now be used to demonstrate the performance of this proposed subband beamformer. It will be compared with that of the fullband beamformer in [2], and for comparison purposes, the experimental setup is exactly same as in [2]. A four-channel (R = 4) microphone array was used. The inter-microphone distance was 5cm and the sampling frequency was 8kHz. For the first simulation, a bandlimited (0-4kHz) white Gaussian signal was used as the target signal and the assumed DOA was 0°. As mentioned before, we are only implementing the BM and MC in subbands, and so the structure of the LACB remains the same as in [1], and so the selection of its parameters can be obtained in [1]. So for comparison we use the following parameters that were used in the fullband beamformer in [2]: we set the filter length of the LAF in the BM and the NCAF in the MC as 40 and 80 respectively, and their step-size parameters were 2 and 0.3 respectively. Moreover, the selection of the β parameter of each LAF in the BM was 1×10^{-6}, and the threshold (K) of each NCAF in the MC was 0.01. And for our proposed subband beamformer, we set M = 4, L = 4 (where M is the number of the subbands and L is the decimation factor). The filter length (P_1) of the LAFs in each subband BM, and the filter length (P_2) of the NCAFs in each subband MC were 10 and 20 respectively. As discussed before, we must select the subband threshold vector ($\mathbf{C}$) so that the thresholds for the lower subbands are much higher than the thresholds in the upper

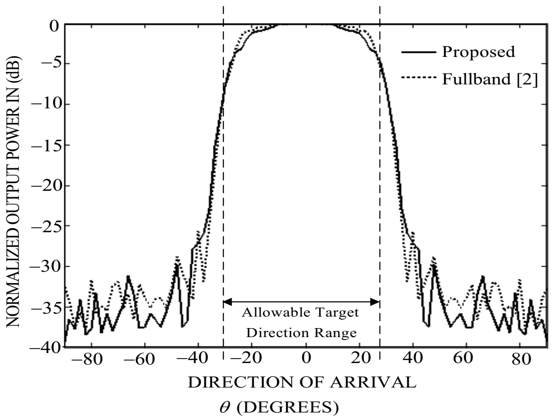

Fig.2. SIMULATION ONE: Normalized output powers of the fullband and subband beamformers for bandlimited white Gaussian input signals.

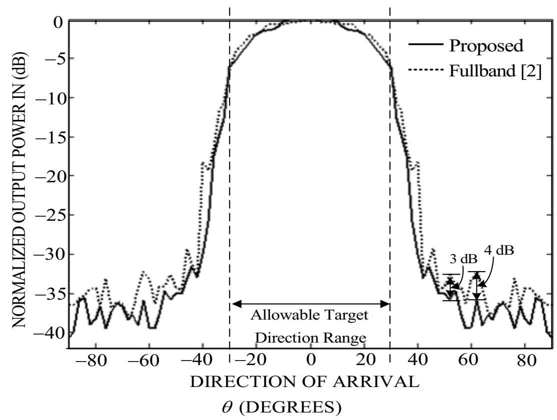

Fig. 3. SIMULATION TWO: Normalized output powers of the fullband and proposed subband beamformer for lowpass input signals.

subbands. This is because a large threshold will give better spatial selectivity, so the beamformer is able to cancel the interferences from outside the target DOA region. But the disadvantage of a higher threshold will cause the signals within the target DOA also to be attenuated. So we have to choose the threshold appropriately. Throughout all the simulations, we set the threshold vector as $\mathbf{C} = \left[0.1\ 0.0005\ 0.0003\ 0.0001\right]^{\mathrm{T}}$. All other parameters are exactly the same as in the fullband beamformer of [2]. They were selected so that the time span of the subband BM $\left(\text{i.e. } L \times P_1 = 40\right)$ and the subband MC $\left(\text{i.e. } L \times P_2 = 80\right)$ are the same as the time span of the fullband BM (i.e. 40) and MC (i.e. 80). So for the first simulation we get Fig. 2. It is clear to see that when the input signal to the microphone array is a bandlimited white Gaussian signal, the performance of the proposed subband beamformer is approximately the same (it only gives slightly better interference reduction outside the target DOA region than that the fullband structure in [2]). But now consider what happens when the input signal is highly correlated.

For the second simulation, the input signal is now generated by passing a white noise through the following low pass filter.

$$H(z) = \frac{1}{1 + 0.9z^{-1}} \tag{5}$$

Fig. 3 now compares the fullband beamformer [2] and the proposed subband beamformer when the input signal to the microphone arrays is highly colored. It has been shown that in this case the beamformer in [4-5] performs poorly, and the interference outside of the target DOA region is not reduced significantly. But from Fig. 3, the fullband beamformer [2] maintains the maximum target DOA range (i.e. $\theta = \pm 30^\circ$) while performing high interference suppression. But the interference suppression near/outside the allowable target DOA region can be further reduced due to the fact that with the frequency-dependent signal blocking capability of the BM then low frequency components of the colored signal are easily cancelled by the BM. So the MC is unable to cancel the low frequency components in the primary input more effectively. This problem is solved by our proposed subband approach. We can now see from Fig. 3 that the proposed subband beamformer is able to achieve slightly better performance near/outside the target DOA region ($30^\circ < \theta < 40^\circ$) while maintaining high interference reduction performance within ($40^\circ \le \theta \le 90^\circ$), which is improved by 1-4 dB over that the fullband beamformer [2] in most cases.

So finally, for the third simulation, we compared the fullband beamformer [2] and the proposed subband beamformer for different SNRs. A target signal source generated a bandlimited white Gaussian signal only for the first 50,000 iterations and it was positioned at 10° off the assumed target DOA. This is to simulate a speech-like burst signal. Another bandlimited white Gaussian signal acts as the DOA of the interference signal for only 90,000 iterations, and it was scanned from -90° to 90°. This was to simulate a continuous interference signal. Both the target and interference signals are "colored" by being passed through the low pass filter in (5). The SNR was defined as the power ratio of the target and interference signals. For the fullband beamformer [2], we set the adaptation of the LAFs in the BM for only 50,000 iterations. This was to simulate when the target signal stops, and the adaptation of the BM also stops, i.e. we use target tracking. And the NCAFs in the MC were then adapted for 90,000 iterations. But for our proposed (4-band) subband beamformer, the adaptation of the LAFs in each subband BM was now reduced to 12,500 iterations, and the adaptation of the NCAFs in each subband MC was for only 22,500 iterations, because with the use of the analysis filter banks, it partitions the input signal spectra into M subbands, and so it will operate at a lower rate. And all other parameters are exactly the same as in the simulations one and two. So Tables I and II show the comparison between the suppression levels of the noise power output in the fullband beamformer [2] and the proposed subband beamformer design structure after convergence as a function of DOA and SNR. As discussed before, for the fullband beamformer, the low frequency noise components are easily cancelled by the BM, so the MC is unable to cancel the interference effectively. However, for the proposed subband beamformer, the input spectrum to the BM is now approximately flat, so it resolves the problem of the blocking capability of the BM. If the BM is adapted during the high SNR periods, the BM is able to cancel the target signal (desirable signal) effectively and leave the interferences (undesirable signals) at the BM outputs. So the MC in this proposed subband

DOA of interference in degrees	SNR input in dB			
	-5	-5	0	0
	Full.	Prop.	Full.	Prop.
-80	35.38	35.79	37.92	37.95
-60	34.02	34.80	30.20	31.29
-40	30.93	33.21	18.67	22.63
-30	16.64	17.44	5.82	20.27
-20	6.90	8.78	5.14	8.39
-10	0.49	3.81	0.79	6.83
0	0.36	2.52	2.33	5.03
10	0.96	1.46	0.67	4.51
20	8.42	9.00	4.79	8.68
30	17.66	20.25	8.97	15.89
50	32.43	38.09	33.28	35.22
70	33.43	36.98	30.19	32.21
90	35.08	35.73	36.85	38.45

Table I. SIMULATION THREE: Comparison between the suppression level of the noise power output in the fullband beamformer (Full.) [2] and the proposed subband beamformer (Prop.) structure after convergence as a function of DOA with different SNR inputs.

DOA of interference in degrees	SNR input in dB			
	5	5	10	10
	Full.	Prop.	Full.	Prop.
-80	17.42	27.88	23.29	29.67
-60	14.39	23.05	18.20	26.42
-40	11.56	20.15	15.64	20.96
-30	7.46	12.43	12.18	16.28
-20	9.29	12.01	11.64	16.52
-10	4.33	10.66	9.85	12.02
0	4.74	10.92	8.37	12.39
10	4.16	10.89	7.52	12.42
20	8.87	14.89	10.58	14.98
30	10.17	16.28	14.39	20.37
50	18.63	21.85	17.22	22.92
70	25.33	28.82	16.72	25.11
90	25.27	28.57	18.96	27.23

Table II. SIMULATION THREE (Cont'd): Comparison between the suppression level of the noise power output in the fullband beamformer (Full.) [2] and the proposed subband beamformer (Prop.) structure after convergence as a function of DOA with different SNR inputs.

beamformer is able to cancel the interferences more effectively As shown in Table II, when the SNR input is high (e.g. 5 dB and 10 dB), the interference suppression inside/outside the target DOA region of this proposed subband beamformer is 3-8 dB better than that the fullband beamformer of [2]. However, if the BM is adapted during the low SNR periods, there will be a large target signal at the MC outputs, and this will cause large misadjustment to the filter coefficients in the MC. This misadjustment error will result in the degradation of the interference reduction performance. When the SNR input is low (e.g. 0 dB and -5 dB in Table I), the interference suppression of this proposed subband beamformer inside the target DOA region is less than when compared with that when the SNR input is high (e.g. 5 dB and 10 dB, see Table II). However, this proposed subband beamformer is still able to achieve high interference reduction outside the target DOA region, even when the SNR input as low as -5 dB. In addition, when the SNR input is low (e.g. 0 dB and -5 dB in Table I), the interference suppression of this proposed subband beamformer still remains higher than that the fullband beamformer in [2].

4. CONCLUSION

We have taken the fullband beamformer structure in [2], re-designed it, and implemented the adaptation process of the BM and MC in subbands. This has the result of solving the problem of the signal blocking capability in the BM when the input signals to the microphone arrays are highly colored. Simulation results show that the interference suppression of this proposed subband beamformer is improved by 1-4 dB when compared with the fullband beamformer of [2] (see Fig. 3), when the input signals to the microphone arrays are highly colored. Moreover, when the SNR input is high (see 5 dB, and 10 dB in Table II), the interference suppression inside/outside the target DOA region of this proposed subband beamformer is improved by 3-8 dB compared with the fullband beamformer. In addition, our new approach has the advantages of reduced complexity (i.e. reduced sampling rate) and faster convergence in the adaptive BM and MC when compared with the fullband beamformer in [2] (see simulation three).

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