Adaptive Subband Arrays for Multipath Fading Mitigation

Yimin Zhang[1], Kehu Yang[2], and Moeness G. Amin[3*]

[1] ATR Adaptive Communications Research Laboratories, Kyoto 619-0288, Japan
[2] Institute of Electronics Engineering, Xidian University, Xi'an, Shaanxi 710071, China
[3] Department of ECE, Villanova University, PA 19085, USA

E-mail : zhang@acr.atr.co.jp

1. Introduction

In high-speed digital mobile communications, inter-symbol interference (ISI) due to multipath fading and co-channel interference (CCI) become two significant problems that decrease the communication quality and limit the communication capacity. Adaptive array alone or together with temporal equalizations or diversity techniques are promising methods to suppress both of the ISI and CCI so that the communication performance can be greatly improved, and, it also leads to the increase of communication capacity.

The signal transmitted by the mobile station is usually reflected and scattered by surroundings, and yields multipath fading [1]. If the time delay between the different paths is large, the multipath fading will be frequency-selective. When an adaptive array is used at a base station in land mobile communication environment, equalizations are often required not only for the desired signal but also for the interference signals. Otherwise a large array aperture with more degree-of-freedom (DOF) is required [2]. However, equalization of the interference signals is usually very difficult since it is often the case that no information about the interference signals is provided.

In this paper, we propose an adaptive subband array with DFT filter banks for multipath fading mitigation in land mobile communications. The purpose is to improve convergence of the desired signal equalization as well as to effectively suppress the interferers. A DFT filter bank transforms the received time-domain data into frequency-domain, and as such yield a multiple narrow-band signal processing problem [3]. The transformed signal has higher signal correlation in each frequency subband due to the reduction in bandwidth. Note that the increase of signal correlation by subband technique is blind in the sense that does not require any information of the arrival signals. With increased coherence, the effect of multipath fading associated with both the desired and the interference signals, can be greatly reduced.

2. Problem Description

Consider an N-element array and K signal arrivals. The 1st signal is the desired signal and the other K-1 signals are interferers. Each signal may arrive from multiple paths.

We assume: (1) the symbol sequences $b_k(n)$ (k =1,2,…,K) are i.i.d. (independent and identically distributed) for a given user and amongst users, i.e., $E\{b_k(n)b_p^*(m)\} = \sigma_b^2\delta_{kp}\delta_{nm}$, where σ_b^2 is the average energy per symbol; (2) the noise is stationary Gaussian and i.i.d. for all the antennas, and is uncorrelated with each user's signal. Denote the kth signal as

$$a(t) = \sum b_k(n)c(t - nT_S) \tag{1}$$

where $c(t)$ is the shaping function, T_s is the symbol period. We consider two multipath rays from the same source with a time delay τ : $a(t)$ and $a(t-\tau)$. To make the analysis simple, we assume the critical sampling case where the sampling rate is once per symbol and the signal spectrum is flat over the signal bandwidth. From assumption (1) we know the signal correlation coefficient between the two rays is 0 when $\tau \geq T_S$.

* Dr. Amin is supported in part by ONR, grant # N00014-98-1-0176.

Consider the signal parameters in Table 1. The desired signal has two rays and the time delay between them is one symbol. There is one interference signal which has three rays, and the time delays are one symbol and two symbols. Even when the desired signal is equalized, conventional adaptive array requires 4 DOF to effectively suppress the interference signal paths and to form beams at the DOA of the desired signal paths. However, by using the proposed method, we will see a 3-element array, which has 2 DOF, is sufficient to do it.

3. Filter Bank-based Adaptive Subband Array

Fig.1 shows a simple DFT filter bank. The combination of time delay (z^{-1}) and decimation operator ($\downarrow M$) functions as a serial to parallel converter (S/P) that split the signal into blocks of length *M*. The *M*-point DFT transform the *M* time-domain blocks of data into *M* frequency-domain blocks of data. Each subband has a bandwidth of 1/*M* of the original one.

Here we consider how the DFT filter bank changes the signal correlation between 2 multipath rays. Again we consider the two signals $a(t)$ and $a(t-\tau)$, and denote their sampled sequence at the *j*th subband as $a^{(j)}(n)$ and $a^{(j)}(n-\tau')$, where τ' is an integer representing τ / T_S (we mention again that the sampling rate F_S is assumed to be once per symbol). The signal correlation coefficient is ρ =0 for $\tau' \geq 1$ or $\tau \geq T_S$. When the signals are split into M subbands, the new *'symbol period'* at the each subband becomes M samples, and the signal correlation coefficient at each subband is

$$|\rho| = 1 - \frac{|\tau'|}{M} \qquad \text{for} \quad |\tau'| \leq M \tag{2}$$

For example, $|\rho|$ becomes to 0.9 for M = 10 and $\tau' = 1$ (i.e., $\tau = T_S$). As $|\rho|$ grows close to 1, the two rays tend to have the property as a single ray so that the multipath fading effect can be mitigated [4].

When the sampling rate F_S per symbol is greater than 1, $|\rho|$ will depend on the shaping function and usually eqn.(2) does not apply. Detailed analysis is not given for this case. However, providing $M > F_S$, it remains true that using DFT filter bank increases the signal correlation coefficient between two rays with time delay.

Fig. 2 shows the block diagram of the adaptive array with DFT filter bank. The LMS algorithm is considered here. The output of each array element is divided into *M* blocks and then transformed by DFT. The reference signal is also transformed into *M* blocks in the frequency-domain by the same process. Adaptive array processing is performed at different subbands and then combined into a single output after IDFT. To summarize the process, the input signal and the reference signal are transformed as

$$x_i(t) \to \{x_i^{(1)}(t), x_i^{(2)}(t), ..., x_i^{(M)}(t)\} \tag{3-1}$$

$$r(t) \to \{r^{(1)}(t), r^{(2)}(t), ..., r^{(M)}(t)\} \tag{3-2}$$

where $(\cdot)^{(j)}$ denotes the *j*th subband component, and $\to$ expresses the transform that splits the signal into subbands. The steady state weight vector at the *j*th subband is given by

$$\mathbf{w}^{(j)} = \mathbf{R}_{XX}^{(j)-1}\mathbf{R}_{Xr}^{(j)} \tag{4-1}$$

where

$$\mathbf{R}_{XX}^{(j)} = E[\mathbf{x}^{(j)*}(t)\mathbf{x}^{(j)T}(t)] \tag{4-2}$$

$$\mathbf{R}_{Xr}^{(j)} = E[\mathbf{x}^{(j)*}(t)r^{(j)T}(t)] \tag{4-3}$$

$(\cdot)^*$ and $(\cdot)^T$ denote complex conjugate and transpose, respectively. The output signal is

$$z^{(j)}(t) = \mathbf{w}^{(j)T}\mathbf{x}^{(j)}(t) \tag{5-1}$$

$$z(t) \leftarrow \{z^{(1)}(t), z^{(2)}(t), ..., z^{(M)}(t)\} \tag{5-2}$$

where $\leftarrow$ expresses the syntheses process which combines the subband signals into a single one.

4. Alternate Representation at Time-domain

Here we consider the alternate representation of the above process in time-domain. The output of the *i*th array element is weighted by $\mathbf{w}_i = \{w_i^{(1)}, w_i^{(2)}, ..., w_i^{(M)}\}$ at the different subbands. This is equivalent to using a FIR filter with *M*-tap weights $\tilde{\mathbf{w}}_i$ where $\tilde{\mathbf{w}}_i$ is given by the IDFT of $\mathbf{w}_i$ as

$$\tilde{\mathbf{w}}_i \leftarrow \mathbf{w}_i = \{w_i^{(1)}, w_i^{(2)}, ..., w_i^{(M)}\} \tag{6}$$

Therefore, the weighting operation can be equivalently realized by FIR filters in the time-domain, and the proposed method by using filter bank can be considered as a powerful to obtain the weights of an adaptive array with tapped-delay lines. The block diagram of the adaptive array with using tapped-delay lines is shown in Fig. 3.

5. Simulation Results

In this section, we examine the behavior and performance of the proposed method using computer simulations. A 3-element equi-spaced linear array is considered and the interelement spacing is half-wavelength. We use the signal parameters as given in Table 1. QPSK modulation is assumed for both the desired and the interference signals. Raised cosine FIR filter is used for data shaping and the roll-off rate is 0.5. The signals are sampled and processed at the symbol rate.

Fig. 4 shows the BER performance of the array vs. the number of subbands. It is evident from this figure that BER decreases rapidly as M increases ($M > 2$). Therefore, the BER performance can be improved by uniformly partitioning the input data bandwidth into three or more narrow bands. Fig. 5 depicts the effect of increased value of M on improved constellations.

6. Conclusions

We have proposed an adaptive subband array based on DFT filter banks. The new scheme is very powerful to reduce the effect of multipath fading at the presence of long time delay profile. It is shown that the BER performance can be greatly improved by using few number of subbands.

Acknowledgements

The authors thank Dr. Bokuji Komiyama, President, Dr. Yoshio Karasawa and Dr. Yoshihiko Mizuguchi, former and current Head of Department 3, all of ATR Adaptive Communications Research Laboratories, for their encouragement and helpful discussions. We also thank the helpful discussion by Dr. Hao Yuan and Dr. Abdesselam Klouche-Djedid.

References

[1] W. C. Lee, "Mobile Communication Engineering", McGraw-Hill, 1982.

[2] Y. Zhang, H. Yuan, and K. Hirasawa, "Adaptive array performance in the presence of spatially spreaded interference signals", IEEE AP-S Int'l Symp., Montreal, July 1997.

[3] N. J. Fliege, "Multirate Digital Signal Processing", John Wiley, 1994.

[4] J. E. Hudson, "Adaptive Array Principles", Peter Peregrinus, 1989.

Table 1 Parameters of the arrival signals

	# of paths	Input SNR(dB)	Time delays (T_S)	DOA (deg)
Desired signal	2	10, 10	0, 1	0, 60
Interference signal	3	10, 10, 10	0, 1, 2	-30, 30, -60

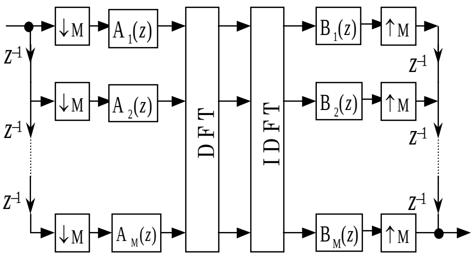


Fig. 1 DFT filter bank

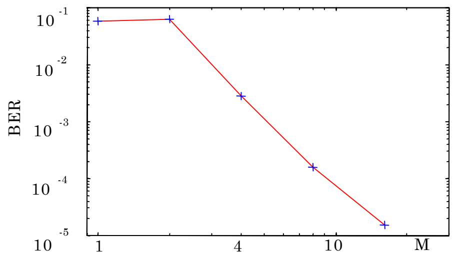


Fig. 4 BER vs. *M*

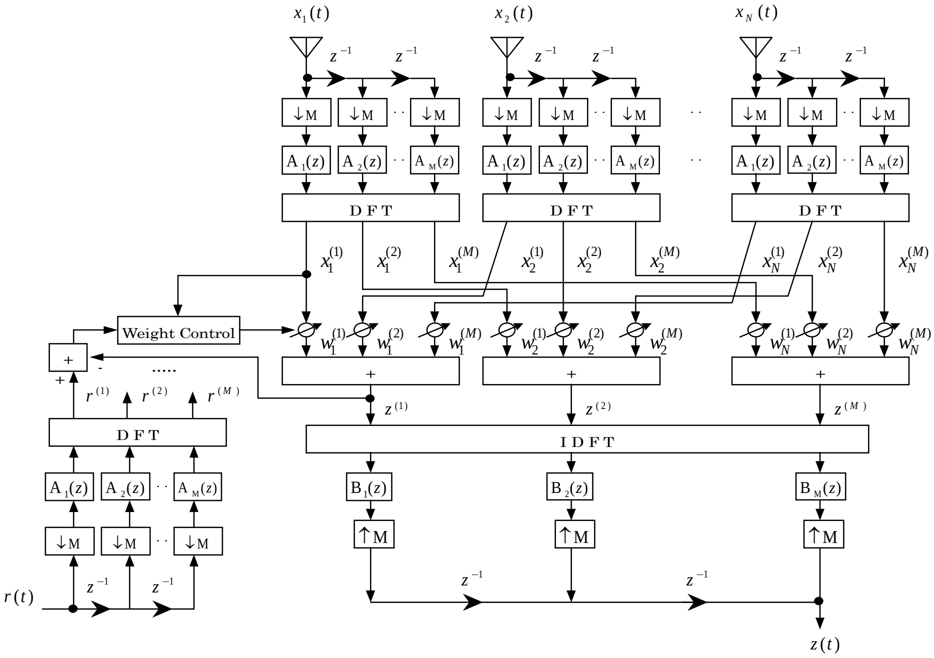


Fig. 2 Adaptive array with DFT filter bank

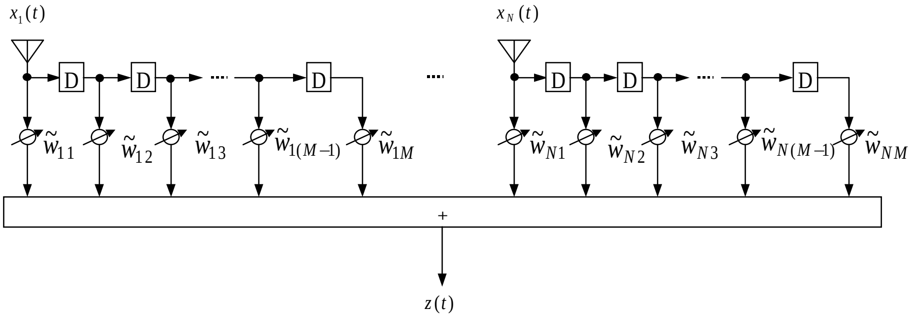
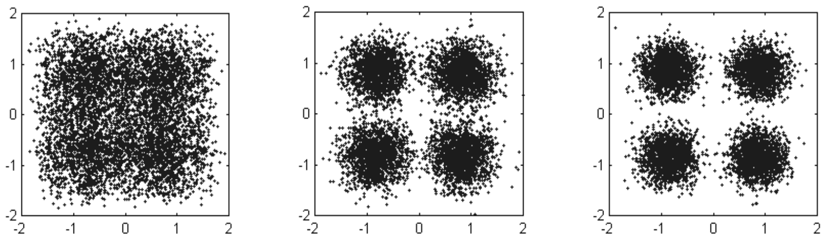


Fig. 3 Adaptive array with tapped delay lines



(a) no filter bank (b) M=4 (c) M=8

Fig. 5 Constellations