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Multichannel Lattice Adaptive Array Processor with Reduced Signal Cancellation

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ABSTRACT

An adaptive array processor which is implemented with a tapped delay line (TDL) filter suffers from a slow convergence rate due to a large eigenvalue spread ratio of input autocorrelation matrix. It is proposed that the lattice filter structure in which the backward prediction errors are uncorrelated between stages is employed in a composite adaptive array processor. It is shown that the covergence rate of the proposed array processor is faster than that of the conventional one due to the orthogonalization property of the lattice filter structure. Since the composite array processor shows no signal cancellation in an ideal sense, the proposed array processor yields a reduced signal cancellation as well as a faster convergence. The performance of the proposed array processor is compared with the TDL counterpart and a generalized sidelobe canceller.

要 約

탭지연여과기로 운용되는 적응어레이 처리기의 수렴속도는 입력상관매트릭스의 큰 아이겐수치의 분포도 때문에 느리게 된다. 단계간의 후방예측 오류가 상관되지 않는 래티스여과구조를 복합적응어레이 처리기에 사용하는 것이 제안되었다. 래티스여과구조의 직각성때문에 제안된 어레이 처리기의 수렴속도는 재래의 어레이 처리기의 수렴속도보다 빠르게 나타났다. 복합적응어레이 처리기에는 이상적으로 신호감쇄 현상이 없기때문에 제안된 어레이 처리기는 빠르게 수렴할 뿐 아니라 신호감쇄를 감소시킨다. 제안된 어레이 처리기의 성능을 탭지연여과기로 운용했을 경우와 일반 측면로브 감쇄기와 비교한다.

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Ⅰ. Introduction

An adaptive array processor consists of an array of antenna elements followed by an adaptive multichannel filter. The array of elements are steered electronically by delaying the element outputs by a proper amount to yield a maximum gain in a look direction (i.e., the direction of a desired signal) while the filter coefficients are updated recursively such that appropriate nulls are created at the nonlook directions (i.e., the directions of interference signals). Adaptive array processing techniques have been widely investigated in the literature [1-7]. The application area includes radar [5], sonar [6], and seismology [7], etc.

In the linearly constrained adaptive array processor proposed by Frost [1], the filter coefficients are updated by a constrained LMS (least mean square) algorithm. The processor responds to a signal coming from a look direction with a preset frequency response while discriminating against the interference signals coming from the nonlook directions. An alternative way of realizing the constrained array processor, referred to as 'generalized sidelobe canceller' was proposed by Griffiths and Jim [2]. In this scheme, the constrained array processor is implemented using an unconstrained LMS algorithm. The main advantage of this approach is that a variety of currently available adaptive multichannl techniques can be easily used.

It was shown [3] that the constrained array processor/generalized sidelobe canceller has the problem of signal cancellation due to interaction between the desired and interference signals during adaptive process. The master-slave type composite array processor introduced by Duvall successfully prevented the signal cancellation phenomenon by eliminating the desired signal in adaptive process using a subtractive preprocessing [3].

The adaptive array processors discussed above use a TDL filter structure and update the filter coefficients using the LMS algorithm. It is well known [8] that the LMS algorithm in the TDL filter results in a slow convergence which depends on the eigenvalue spread ratio of input autocorrelation matrix. One way of decreasing the eigenvalue spread ratio is to reduce the dimension of the autocorrelation matrix by orthogonalizing the input signals. In this respect, the lattice filter structure is appropriate for use due to its Gram Schmidt orthogonality property between stages.

In this paper, it is proposed to employ a lattice filter structure in the composite array processor to improve the convergence rate with reduced signal cancellation. To this end, the composite array processor is realized in an unconstrained way with a lattice filter strucure instead of the TDL one. Due to the orthogonalization property of the lattice structure, the proposed realization converges faster and yields a smaller estimation error than its TDL counterpart. Also, the signal cancellation is rarely observed in the proposed realization.

Ⅱ. Unconstrained Composite Array Processor

The composite array processor is a master-slave type array processor which prevents the signal cancellation phenomenon inherent in the constrained array processor. It is proposed that the composite array processor which employes the constrained LMS algorithm be implemented in an unconstrained way as shown in Fig. 1. It consists of a master and slave processors each of which is identical to the generalized sidelobe canceller structure. In the figure, D_i, $1 \le i \le N$ represents the time delay sample corresponding to the ith element. It is assumed that the antenna elements are equally spaced with half the wavelength. If a desired and interference signals, s(k) and n(k), are coming from θ_s and θ_n from the array axis, the input signal to the ith element can be represented by

$$x_i(k)=s(k+\tau_{si})+n(k+\tau_{ni}),\ 1\le i\le N, \quad (1)$$

where

$$\tau_{si}=(i-1)\tau_d\cos\theta_s, \qquad (2)$$
$$\tau_{ni}=(i-1)\tau_d\cos\theta_n, \qquad (3)$$

τ_d is a propagation delay in terms of sample between neighbering elements, and k is a discrete time index. Now, the outputs of antenna elements are delayed by τ_{si} samples to steer the array of elements to the look direction such that the desired signals are in phase in the delayed element outputs which are given by

$$x_i(k)=s(k)+n(k+\tau_{ni}-\tau_{si}),\ 1\le i\le N. \qquad (4)$$

In the master processor, the desired signal components are eliminated by a subtractive preprocessing between neighbering element outputs. The filter coefficients in the master processor are updated using a general adaptive multichannel noise cancellation approach [9] in which the $y_c(k)$ and $u_i(k)$, $1\le i\le N-2$ are used as the primary and reference inputs respectively. $y_c(k)$ is the output of the spectral constraint filter via a delay-and-sum array which is formed by the sample delay elements and weights c_i, $1\le i\le N-1$. The spectral constraint filter passes the input signal within the bandwidth of the desired signal with a gain of one. The outputs of the matrix processor $u_i(k)$, $1\le i\le N-2$ are processed through a multichannel TDL filter whose coefficients are updated by the unconstrained LMS algorithm [4] which is given by

$$A_l(k+1)=A_l(k)+\Delta(k)y(k)U(k-l),\ 0\le l\le L-1, \qquad (5)$$

where

$$A_l(k)=[a_{1,l}(k)\ a_{2,l}(k)\cdots a_{N-2,l}(k)]^T, \qquad (6)$$
$$U(k-l)=[u_1(k-l)\ u_2(k-l)\cdots u_{N-2}(k-l)]^T, \qquad (7)$$

$a_{i,l}(k)$ is the ith row coefficient at the lth tap in the multichannel TDL filter, $u_i(k-l)$ is the input to the lth tap of the ith TDL filter and consists of subtractive interference signals, $y(k)$ is the output of the master processor, $\Delta(k)$ is an (N-2)×(N-2) diagonal matrix, in which the ith diagonal element represents a time-varying convergence parameter given by

$$\mu_i = \frac{1-\beta}{\beta v_i(k-1) + (1-\beta)u_i^2(k)}, \quad 0 < \beta < 1. \qquad (8)$$

In (8), a simple one-pole lowpass filter is used to compute $v_i(k)$, which is an estimate of the variance of $u_i(k)$. Also, the smoothing parameter β controls the time constant of the one-pole lowpass filter. Since only the interference signals are involved in adaptive process as shown in (5), the filter coefficients in the master processor is updated with no interaction between the signal and interference. The updated coefficients are copied to the slave processor and processed with the interference signals with no subtractive preprocessing to produce the final array output. The purpose of the matrix processor is to eliminate the desired signal using $(N-2)\times(N-1)$ transformation matrix where each row is independent of others and the row elements sum to be zero. One way of satisfying these conditions is to use Walsh-ordered Walsh-hadamard matrices [10]. If the desired signal is not eliminated perfectly, the estimation performance will be degraded due to the leakage of the desired signal into the multichannel TDL filter. It is to be noted that the interference signals out of the subtractive preprocessor in the master processor have the same phase relationship as those in the delayed element output signals in the slave processor. Thus the copied coefficients in the slave processor produce appropriate nulls at the locations of the incoming interference signals. It was shown that the signal cancellation phenomenon was remarkably reduced in the composite array processor [3].

Even though the unconstrained LMS algorithm has been successfully applied to update the multichannel TDL filter coefficients, it suffers from a slow convergence rate due to the eigenvalue spread ratio of input autocorrelation matrix. To overcome this shortcoming, the multichannel lattice filter structure has been developed [11]. It is well known that the convergence rate of the adaptive multichannel lattice filter which uses the LMS algorithm is partially independent of the eigenvalue spread ratio of the autocorrelation matrix due to its Gram Schmidt orthogonalization property. To realize a fast convergence adaptive array processor with reduced signal cancellation, we apply the multichannel lattice structure to the composite array processor.

Ⅲ. Composite Array Processor with Lattice Structure

To realize a composite array processor with a lattice structure, we substitute the L-1 stages of a multichannel lattice structure as shown in Fig. 2 into the TDL counterpart in the master and slave processors in the composite array procesor. The resulting composite array processor with lattice structure is shown in Fig. 3. In Fig. 2, $\boldsymbol{F}_i(k)$ and $\boldsymbol{B}_i(k)$, $1 \le i \le L-1$ denote the $(N-2)\times 1$ forward and backward prediction error vectors in the ith stage, and $\boldsymbol{W}_i^f(k)$ and $\boldsymbol{W}_i^b(k)$, $1 \le i \le L-1$ are $(N-2)\times(N-2)$ forward and backward coefficient matrices. They are recursively updated using the unconstrained LMS algorithm to minimize the mean squared norms of $\boldsymbol{F}_i(k)$ and $\boldsymbol{B}i(k)$, $1 \le i \le L-1$, respectively. Also, $\boldsymbol{G}_i(k)$, $0 \le i \le L-1$ are $(N-2)\times 1$ coefficient vectors which are also updated iteratively using the LMS approach to minimize the mean squared value of the ith stage error signal $e_i(k)$, which is the difference between the $(i-1)$th stage error signal $e_{i-1}(k)$ and the weighted backward prediction error vector $\boldsymbol{B}i(k)$. Here, $e_{L-1}(k)$ corresponds to the $y(k)$ in Fig. 1. The relevant recurrence relationships are as follows.

$$\boldsymbol{F}_0 = \boldsymbol{B}_0(k) = \boldsymbol{U}(k), \tag{9}$$

$$\boldsymbol{F}_i = \boldsymbol{F}_{i-1}(k) - \boldsymbol{W}_i^f(k)\boldsymbol{B}_{i-1}(k-1), \tag{10}$$

$$\boldsymbol{B}_i = \boldsymbol{B}_{i-1}(k-1) - \boldsymbol{W}_i^b(k)\boldsymbol{F}_{i-1}(k),\ 1 \le i \le L-1, \tag{11}$$

$$e_0(k) = y_c(k) - G_0^T(k)B_0(k), \tag{12}$$

$$e_i(k) = e_{i-1}(k) - G_i^T(k)B_i(k),\ 1 \le i \le L-1, \tag{13}$$

where

$$\boldsymbol{B}_i(k) = [b_{i,1}(k)\ b_{i,2}(k) \cdots bi_{,N-2}(k)]^T, \tag{14}$$

$$\boldsymbol{F}_i(k) = [f_{i,1}(k)\ f_{i,2}(k) \cdots f_{i,N-2}(k)]^T, \tag{15}$$

$$\boldsymbol{G}_i(k) = [g_{i,1}(k)\ g_{i,2}(k) \cdots g_{i,N-2}(k)]^T, \tag{16}$$

and i is the stage index. The mth row and nth column component of $W_i^f(k)$ is the ith stage forward coefficient which predicts the mth forward prediction error using the nth one-sample delayed backward prediction error of the $(i-1)$th stage; The mth row and nth column component of $W_i^b(k)$ is the ith stage backward coefficient which predicts the mth backward prediction error using the nth forward prediction error of the $(i-1)$th stage. It has been shown [12] that the backward prediction errors are mutually orthogonal, i.e.,

$$E[\ B_i(k)\ \ B_i^T(k)] = \begin{cases} 0 & \text{for } i \ne j \\ \Sigma_i & \text{for } i = j, \end{cases} \quad 1 \le i, j \le L-1 \tag{17}$$

where $\boldsymbol{0}$ denotes an $(N-2)\times(N-2)$ null matrix. From (17), it is shown that each stage error is uncorrelated with other stage errors. The LMS algorithm updating the coefficient vectors/matrices is as follows.

$$G_i(k+1) = G_i(k) + \Lambda_i^g(k)\ e_i(k)\ B_i(k),\quad 0 \le i \le L-1 \tag{18}$$

$$W_i^{fT}(k+1) = W_i^{fT}(k) + \Lambda_i^f(k)\ B_{i-1}(k-1)\ \ F_i^T(k) \tag{19}$$

$$W_i^{bT}(k+1) = W_i^{bT}(k) + \Lambda_i^b(k)\ F_{i-1}(k)\ B_i^T(k),\quad 1 \le i \le L-1 \tag{20}$$

where $\boldsymbol{\Lambda}_i^g$, $\boldsymbol{\Lambda}_i^f$, $\boldsymbol{\Lambda}_i^b$ are $(N-2)\times(N-2)$ diagonal matrices at the ith stage in which the jth diagonal element is given by

$$\mu_{i,j}^{g}(k)=\frac{1-\beta}{\beta\, v_{i,j}^{g}(k-1)+(1-\beta)\, b_{i,j}^{2}(k)} \qquad (21)$$

$$\mu_{i,j}^{f}(k)=\frac{1-\beta}{\beta\, v_{i,j}^{f}(k-1)+(1-\beta)\, b_{i-1,j}^{2}(k-1)} \qquad (22)$$

$$\mu_{i,j}^{b}(k)=\frac{1-\beta}{\beta\, v_{i,j}^{b}(k-1)+(1-\beta)\, f_{i-1,j}^{2}(k)} \qquad (23)$$

and $v_{i,j}^{g}(k)$, $v_{i,j}^{f}(k)$, and $v_{i,j}^{b}(k)$ are the estimates for the variances of $b_{i,j}(k)$, $b_{i-1,j}(k-1)$, and $f_{i-1,j}(k)$, respectively. Since the successive orthogonalization makes each stage independent of others, the convergence rate does not depend on the eigenvalue spread of the L(N-2)×L(N-2) autocorrelation matrix, which is given by

$$\Sigma_{V} = E[\ V(k)\ V^{T}(k)] \qquad (24)$$

where

$$V(k)=[U(k)\,U(k-1)\cdots U(k-L+1)\ ]^{T} \qquad (25)$$

$$U(k-j)=[u_{1}(k-j)\,u_{2}(k-j)\cdots u_{N-2}(k-j)\ ]^{T} \qquad (26)$$

but only on the eigenvalue spread of the $(N\text{-}2)\times(N\text{-}2)$ autocorrelation matrix of $F_i(k)$ or $B_i(k)$, $0\le i\le L\text{-}1$, i.e., Σ_i in (17). This property provides a faster convergence rate which can not be achieved with the corresponding multichannel TDL structure, since the latter's convergence rate depends on the eigenvalue spread of Σ^{V} in (24).

It can be shown that the multichannel TDL structure coefficients are equivalent to the coefficients of the multichannel lattice structure via the following recurrence relationships in which the time index k is omitted for covenience.

$$P_1^{1} = W_1^{f} \qquad (27)$$

$$P_j^{i} = P_j^{i-1} \quad W_i^{f}\ R_{i-j}^{i-1} \qquad (28)$$

$$P_i^{i} = W_i^{f} \qquad (29)$$

$$R_1^{1} = W_1^{b} \qquad (30)$$

$$R_j^{i} = R_j^{i-1} - W_i^{b}\ P_{i-j}^{i-1} \qquad (31)$$

$$R_i^{i} = W_i^{b},\ 2\le i\le L-1 \text{ and } 1\le j\le i-1 \qquad (32)$$

$$A_1^{1} = G_0 \qquad (33)$$

$$A_m^{l\,T} = A_{l-m}^{l-1\,T}\ G_{l-1}^{T}\ R_{l-m}^{l-1} \qquad (34)$$

and

$$A_l^{l} = G_{l-1},\ 2\le l\le L \text{ and } 1\le m\le l-1 \qquad (35)$$

where P_j^i and R_j^i represent the jth forward and reverse prediction coefficient matrices; i past and future input vector signals are used to predict the current input vector; and A_m^L is the corresponding TDL coefficient vector following the signal vector $U(k\text{-}m\text{+}1)$. The above recurrence equations was used to find the beam patterns of the lattice realization of the composite array processor. It can be shown that the total number of coefficients for the $(L\text{-}1)$-stage multichannel lattice filter is $(N\text{-}2)L+2(N\text{-}2)^2(L\text{-}1)$ while that for the corresponding TDL filter is $(N\text{-}2)L$. The final output signal for the lattice realization, $\tilde{e}_{L-1}(k)$ is equivalent to $\tilde{y}(k)$ in Fig. 1.

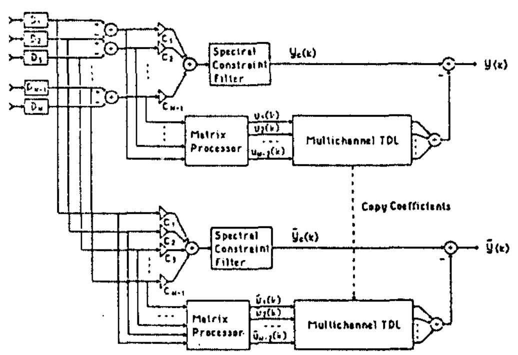

Fig. 1. Unconstrained composite array processor.

Ⅵ. Simulation Results

To observe the performances of the TDL and lattice realizations of the composite array processor, a desired and a interference signals, which are uncorrelated each other, are generated by processing two independent Gaussian random sequences through a 8th order Butterworth bandpass filter with a sampling frequency of 400Hz. The lower and upper cut-off frequencies for the desired signal are 40Hz and 60Hz and those for the interference are 48Hz and 52Hz, respectively. The power spectra for the desied and interference signals are shown in Fig. 4. A 4-element linear array with 5 taps per element was simulated. Thus the corresponding multichannel lattice filter has 4 stages. The element spacing is assumed to be half the wave length corresponding to 50Hz. The desired signal is incident at the direction perpendicular to the array axis and an interference signal arrives at 41.4° from the array axis. $c_i=0.25$, $1\leq i\leq 4$ are used for the weights of the delay-and-sum array. A 3×4 Walsh-ordered Walsh-Hadamard matrix is used for the matrix processor which is given by

$$\begin{vmatrix} 1 & 1 & -1 & -1 \\ 1 & -1 & -1 & 1 \\ 1 & -1 & 1 & -1 \end{vmatrix} \qquad (36)$$

Since no interference signal is coming from the look direction, an all-pass filter was employed for the spectral constraint filter (i.e., the coefficients are [1 0 0 0 0]) and 0.99 was used for the smoothing parameter. The beam patterns and frequency responses for the lattice realization of the composite array processor were plotted by using the equivalent TDL filter coefficients calculated by the relationships in (27)-(35). The error signals which are the difference between the desired signal and the array output are shown in Fig. 5 for the TDL and lattice realization for $3100\leq k\leq 3600$. Beam patterns at 50Hz and frequency responses at 41.4° for $k=3151$ are displayed in Figs. 6 and 7. It is observed that the lattice realization converges faster and yields deeper nulls around the interference direction and frequency than the TDL counterpart. To see the signal cancellation phenomenon, the power spectra of the desired signal and array output for the TDL and lattice realizations are displayed in Fig. 8. It is shown that signal cancellation is rarely observed in the power spectra for both realizations. For comparision, the correponding plots for the generalized sidelobe canceller are displayed. The error signals for the TDL and lattice realizations of the generalized sidelobe canceller are displayed in Fig. 9. The beam patterns at 50Hz and frequency responses at

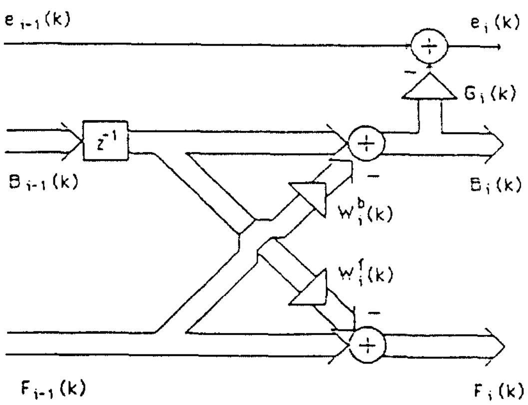

Fig. 2. The ith stage of an adaptive multichannel lattice filter.

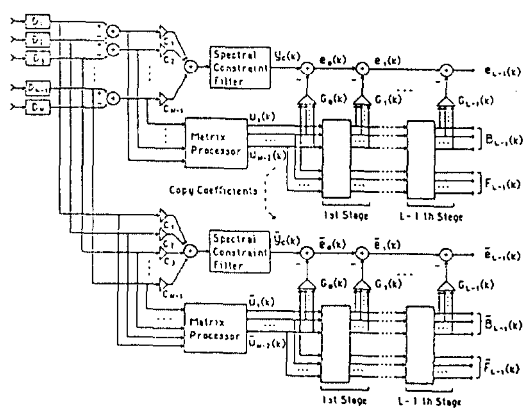

Fig. 3. Lattice realization of composite array processor.

41.4° for k=3151 are displayed in Figs. 10 and 11. The power spectra are shown in Fig. 12. Comparing the results of the composite array processor with those of the generalized sidelobe canceller, the former shows a faster convergence and more powerful nulling of the interference signal in the space and frequency domains. Also, it is observed that signal cancellation phenomenon is significantly reduced in the composite array processor compared with that of the generalized sidelobe canceller.

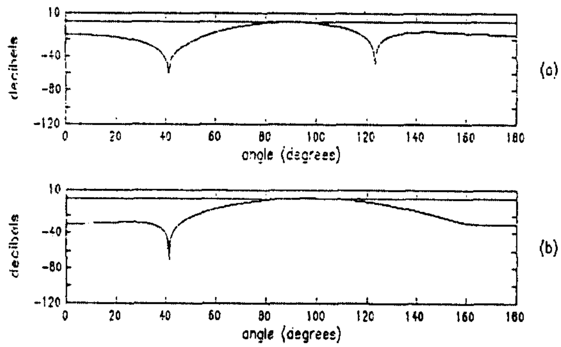

Fig. 6. Beam patterns: (a) TDL and (b) lattice realizations.

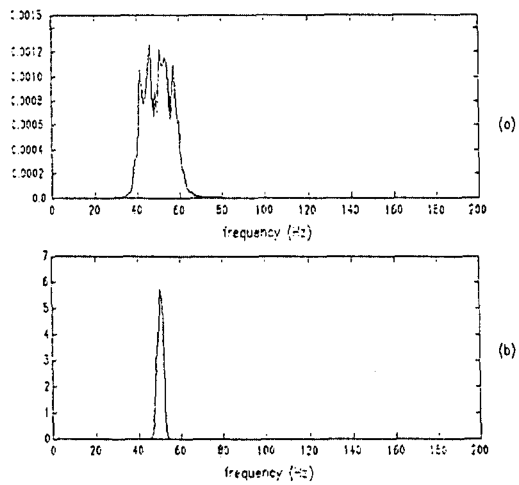

Fig. 4. Power spectra (a) desired and (b) interference signals.

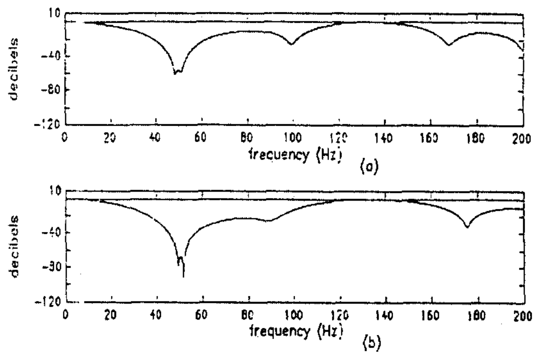

Fig. 7. Frequency reponses: (a) TDL and (b) lattice realizations.

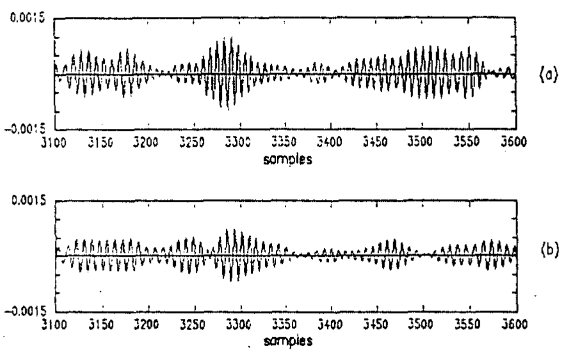

Fig. 5. Error signals: (a) TDL and (b) lattice realizations.

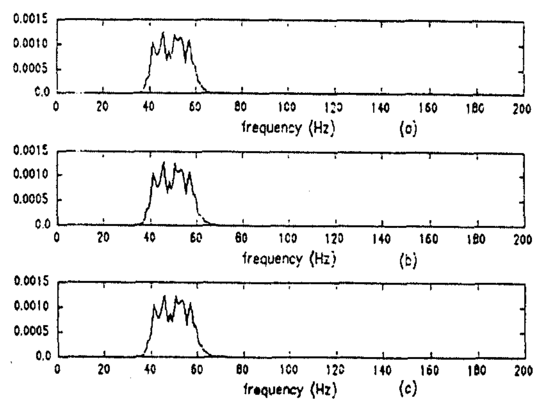

Fig. 8. Power spectra: (a) desired signal; (b) TDL, and (c) lattice realizations.

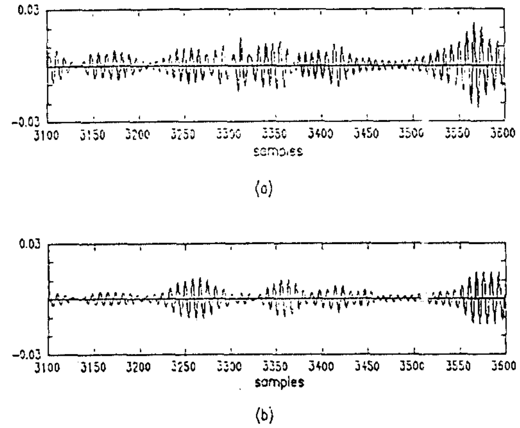

Fig. 9. Error signals: (a) TDL and (b) lattice realizations for generalized sidelobe canceller.

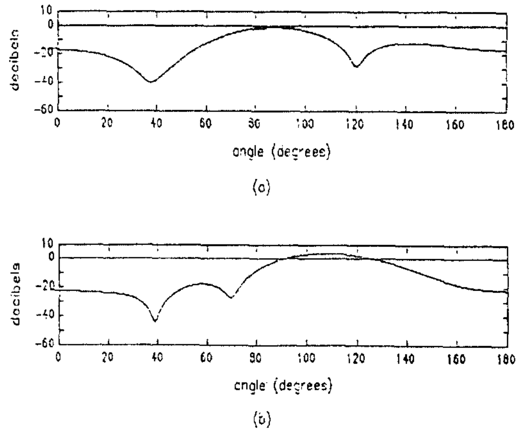

Fig. 10. Beam patterns: (a) TDL and (b) lattice realizations for generalized sidelobe canceller.

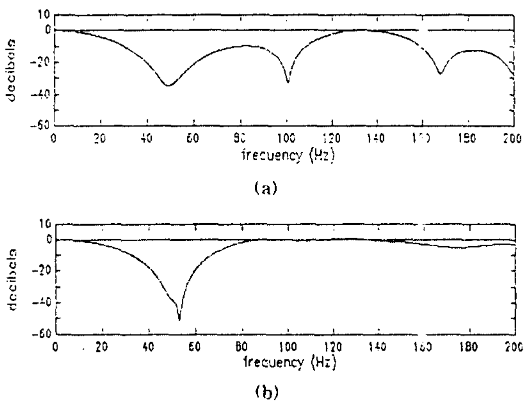

Fig. 11. Frequency responses: (a) TDL and (b) lattice realizations for generalized sidelobe canceller.

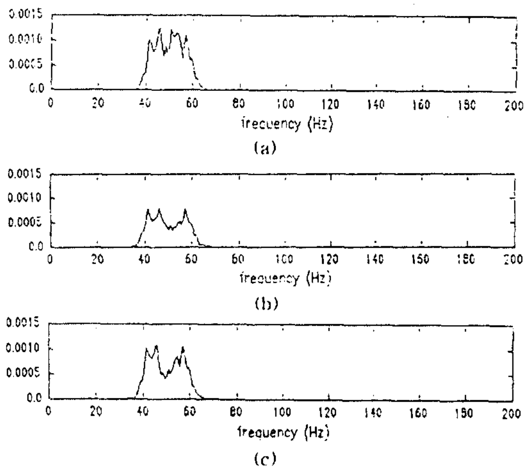

Fig. 12. Power spectra: (a) desired signal; (b) TDL, and (c) lattice realizations for generalized sidelobe canceller.

V. Conclusions

To realize a fast convergence adaptive array processor with reduced signal cancellation, it was proposed to use a multichannel lattice filterstructure in the unconstrained realization of compsoite array processor. The partial orthogonalization of the input signals by the lattice filter structure provides a fast convergence rate than the TDL structure. It was observed that the proposed realization performs better than the TDL counterpart in terms of convergenced rate and elimination of the interference signal in space and frequency domains. Also, almost no signal cancellation phenomenon was observed in the proposed array processor. The performance of the composite array processor were compared with that of generalized sidelobe canceller. It is recommended that the TDL as well as lattice realizations of the composite array processor be used in practical array systems in estimating/detecting a desired signal.

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