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Subband Signal Processing Adaptive Array with a Data Transmission Scheme Inserting a Cyclic Prefix

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SUMMARY

The advantages of subband signal processing adaptive array (SBAA) are the ability to perform parallel signal processing in each subband division and the substantial reduction in the total computational load. However, the price is not preventing a drop in performance, for example, the incorporation of delayed waves, and a countermeasure must be found. Therefore, in this paper, we propose introducing a scheme that inserts a guard interval (cyclic prefix), which is commonly adopted in Orthogonal Frequency Division Multiplexing (OFDM), into ordinary data transmission, and extracts the best possible performance of SBAA. We also evaluate the performance by a computer simulation. © 2002 Wiley Periodicals, Inc. Electron Comm Jpn Pt 1, 86(3): 1–8, 2003; Published online in Wiley InterScience (www.interscience.wiley.com). DOI 10.1002/ecja.10055

Key words: adaptive array; subband signal processing; cyclic prefix; interference suppression; waveform equalization.

1. Introduction

An adaptive array configuration that efficiently extracts the desired signal that has a delay spread due to multipath propagation and suppresses interference is the space-time domain signal processing adaptive array. Research aimed at the application to wideband mobile communication is actively pursued [1, 2]. The basic configuration for time domain signal processing is the Tapped Delay Line (TDL). The Tapped Delay Line Adaptive Array (TDLAA) is an adaptive array that fuses the TDL and spatial signal processing. Studies have addressed the Subband Signal processing Adaptive Array (SBAA) structure that obtains the target performance by transforming a portion of the time domain processing into the frequency domain by a Discrete Fourier Transform (DFT) [or Fast Fourier Transform (FFT)], dividing the frequency spectrum into subbands and processing the spatial signals in each band [3–7]. The advantages of the SBAA are its ability to perform parallel signal processing in each divided subband and the substantial reduction in the overall computational load. However, the price is not preventing a drop in performance, for example, the ability to incorporate the delayed waves, and solving this problem is thought to be critical [6, 7]. Therefore, in this paper, we propose introducing a scheme that inserts a guard interval (cyclic prefix), which is adopted in the Orthogonal Frequency Division Multiplexing (OFDM) [8], into normal data transmission to improve the multipath signal combining functions. We also evaluate the performance by a computer simulation.

2. The Advantages and Problems of SBAA

Figure 1 shows the subband signal processing adaptive array (SBAA) adopted in this paper. A typical TDL adaptive array (TDLAA) transforms a portion of the time domain signal processing into the frequency domain and optimizes the weights in each subband. This version stores

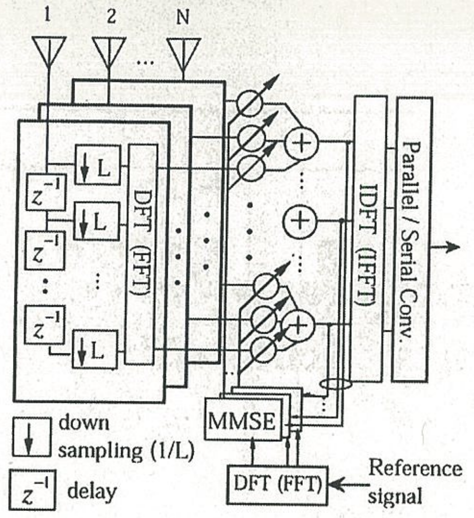

Fig. 1. Configuration of subband signal processing adaptive array (SBAA).

and batch processes multiple symbols (M). The down arrows in the figure represent subsampling (decimation) in time, once for every L samples. When one symbol has p divisions, the actual data read in becomes $L = Mp$ samples.

Other types of SBAA are (i) the application of the DFT (or FFT) to each sample, the optimization for each subband, and obtaining the final output from one IDFT tap, and (ii) the application of the DFT (or FFT) to each sample, and optimization for the entire synthesized signal but not optimization in each subband [3]. Configuration (i) has no major difference in performance from the configuration in Fig. 1 in terms of the increase in the amount of signal processing [6]. Configuration (ii) has identical performance to TDLAA even for the convergence time [3, 6] and does not exhibit the advantages of SBAA in particular. For these reasons, we will examine the configuration in Fig. 1 and call this structure SBAA.

By comparing the features of TDLAA and SBAA, we can make the following adjustments.

(1) TDLAA optimizes the weights for the signal at the final stage. In contrast, SBAA optimizes each subband, that is, each part. If the number of taps is the same, the TDLAA has superior performance. (However, the degree of degradation depends on each assumed case [6].)

(2) SBAA operates in each subband, can be implemented as parallel processing, and substantially reduces the overall processing load. In other words, SBAA is extremely superior from the perspective of the computational load [6, 7].

The configuration based on subband signal processing is well known as a structure suited to processing wideband signals. Reference 4 first proposed a configuration that adopted decimation in the adaptive array. In Ref. 7, the number of computations required until convergence is estimated to be $(NMp)^3$ in TDLAA and the number of DFT/IDFT calculations + $(Mp)N^3$ in SBAA where N is the number of antennas. As Mp increases, the portion of DFT/IDFT calculations decreases relatively. If $(NMp)^3$ and $(MP)N^3$ are compared, the ratio of the two becomes $(Mp)^2$. For example, when $M = 8$ and $p = 4$, the SBAA configuration is estimated to have approximately 1/1000 the computational load of the TDLAA configuration (where the number of TDL taps has the same setting).

SBAA is not solely good, and the performance degrades as described in (1). Most important, the combining function of multipath waves having delay spread no longer functions as the signal-to-noise ratio (SNR) increases. In Ref. 7, when $N = 8$ (linear N element array with half-wavelength intervals), $M = 8$, and $p = 4$, we calculate the signal-to-interference plus noise ratio (SINR) after optimization based on the minimum mean square error (MMSE) criterion when the desired signal from $\theta = 0°$ direction synchronized to the reference signal and the desired signal delayed by L symbols from the 30° direction have arrived. When the input SNR of the elements is 0 dB, even a delayed wave of one symbol is removed (bold line in Fig. 7 of Ref. 7 and in Fig. 4 of this paper). In this case, if two signals are effectively incorporated, the output SINR is expected to increase by 3 dB. This can be interpreted as follows. The correlation coefficient of the delayed signal to the reference signal (correlation coefficient in block units absorbed as a batch: ρ) is reduced, the power of the delayed signal is divided into a correlated (coherent) component and an uncorrelated (incoherent) component at $|\rho|^2{:}(1 - |\rho|^2)$, and acts as a signal with an SINR $= |\rho|^2/(1 - |\rho|^2)$ [7].

From the performance perspective, these problems can be solved by improving the processing performance in the time domain, that is, by increasing the number of TDL taps. However, these problems arise from different perspectives, such as delays and convergence times in the signal processing. By signal processing only using the received signals, such degradation inherent to SBAA is believed to be inevitable. Therefore, by modifying the transmitted signal, we propose a new data transmission scheme that brings out the advantages of subband signal processing. By combining the new data transmission scheme and SBAA, the computational load is smaller and high-performance data transmission is possible.

3. The Proposed Transmission Scheme and the Antenna Configuration

The problems described earlier are present in the batch processing each time, and the correlation of the

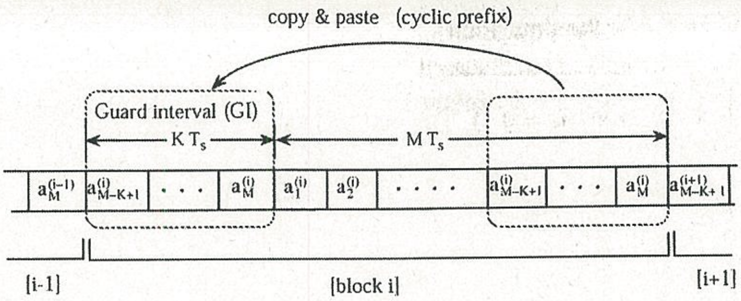

Fig. 2. Data transmission with an inserted cyclic prefix (CP).

delayed signal to the reference signal degrades. The OFDM transmission scheme provides a temporal guard interval (GI) that can absorb the spread of the propagation delay in the transmitted signal transformed by the IDFT and performs an efficient DFT calculation on the receive side. In other words, errors caused by intersymbol interference are prevented [8]. In order to enhance this effect, we use the cyclic prefix (CP), which copies the last part of the data after the IDFT in the same symbol in the GI to the beginning. Therefore, this scheme is incorporated into ordinary data transmission. If this signal is received and processed by SBAA, high-performance data transmission can be expected overall. Figure 2 shows the format of the transmit signal where the data of M symbols is one block and the K symbols of the CP are inserted at the beginning. The transmission efficiency becomes $M/(M + K)$ in this stage. For example, even if $M = 16$ and $K = 4$, the degradation in the efficiency is 1 dB. If a larger gain than the degradation is obtained, the proposed method is effective. Similar to OFDM on the receive side, the M first symbols where all of the desired multipath signals having delays in the GI period become synchronized signals are extracted and processed. This is shown in Fig. 3. If this data is extracted, one period of the data block (Data Block i in the figure) is entered in

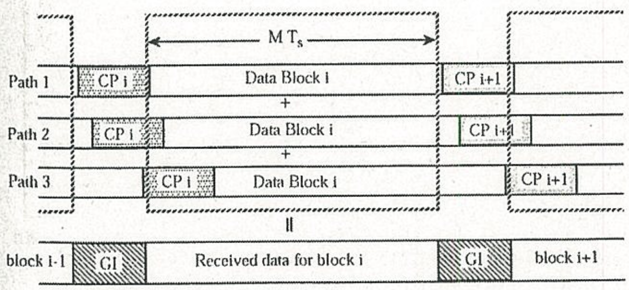

Fig. 3. Extraction of the block signal from the received data having delay spread.

all of the paths. Excluding the GI removal circuit, the SBAA configuration is the unmodified SBAA configuration shown in Fig. 1. The difference is only the decimation ratio $[L = Mp \rightarrow L = (M + K)p]$ that accompanies the data block for every $(M + K)p$.

If extracted in this manner, all of the path signals having delays within the GI period become equivalent on the frequency domain with respect to the reference signal. The degradation in the correlation for the delay signals, which become a problem in the ordinary transmission format that does not insert the CP, is overcome. Thus, good incorporation of the delayed waves can be expected.

4. The Simulation Method

Bandlimited QPSK modulated signals (baseband signals) are used for both the desired signal and the interference signal. Table 1 shows the parameters (signal waveforms and antenna specification) and their settings.

Since the performance evaluation is emphasized in this paper, we use the SMI (Sampled Matrix Inversion) algorithm that directly finds the Weiner solution in the weight determination. The performance after weight convergence is evaluated by the signal-to-interference plus noise ratio (SINR). The following equation calculates the SINR from the correlation coefficient ρ between the synthesis signal $y(i)$, for $i = 1, 2, \ldots$, and the reference signal $r(i)^{\dagger}$:

$$\mathrm{SINR} = \frac{|\rho|^2}{1 - |\rho|^2}, \qquad \rho = \frac{< r^* y >}{\sqrt{< |r|^2 >< |y|^2 >}} \qquad (1)$$

where < > is the mean over a sufficiently long time.

†The quantity found by Eq. (1) corresponds to the carrier-to-interference signal plus noise ratio (CINR). Since CINR = SINR for QPSK, this is also the SINR.

Table 1. Parameters and settings for the simulation

Modulation	QPSK
Filter	Roll-off filter ($\alpha = 0.5$)
Symbols/block	$M = 16$
Guard interval	$K = 4$
Samples/symbol	$p = 4$
Antenna array	$N = 8$ ($d = \lambda/2$)
Delay taps	$Mp = 64$
Adaptive algorithm	SMI

5. Operation and Performance

5.1. The capturing function of the desired signal having arrival angle spread and delay spread

Table 2 shows five cases (environments) for evaluating the capturing function for multiple incident signals having different incident angles and delays. The strengths of the paths are equal. Figure 4 shows the SINR characteristics for the five cases when $N = 8$, $M = 16$, and $K = 4$ (solid lines). The vertical axis is the relative SINR with reference to $SNR_{in} \times N$ (= 0 dB). For example, if the input SNR (SNR_{in}) of one signal is 10 dB for $N = 8$, the calculated output level is 19 (= 10 + 9) dB. This level is set to 0 dB. For comparison, the result for $M = 16$ is also shown for data with no CP inserted (dashed line). It is seen that when the SNR_{in} is low in the conventional method, absorption occurs to some extent for the low SNR_{in}, but the case where the SNR_{in} is higher than 0 dB becomes the mode that removes nearly everything as described in Section 2. In contrast, the proposed method has higher gain as expected for the desired signal delayed by up to three symbols (expected value of 6 dB for Case 4). As described earlier, the loss (= 1 dB) in the transmission efficiency due to CP insertion must be discounted, but gains exceeding this value are obtained. Consequently, the method adequately exhibits this effect in a strong multipath environment.

Table 2. Environments for evaluating the absorption function for multiple incident signals (same strength in each path)

Path definition

Path	Angle (deg)	Delay (symbol)
1	0	0
2	15	1
3	-15	2
4	30	3
5	-30	4

Case definition

Case	Path
1	1
2	1+2
3	1+2+3
4	1+2+3+4
5	1+2+3+4+5

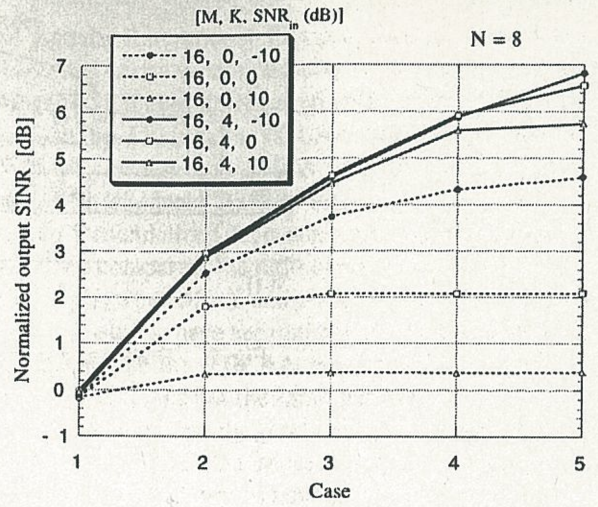

Fig. 4. Absorption performance of the desired signal with various delays (see Table 2 for cases).

Figure 5 compares the antenna patterns of the proposed method and the conventional method when SNR_{in} = 10 dB in Case 4 of Fig. 4. The patterns are calculated with the converged weights of the subbands ($f = 0$ of the baseband signal) corresponding to the center frequency. From the same figure, it is seen that there was multipath rejection when the CP was not inserted, but excellent incorporation of multipath signals is evident in the proposed method.

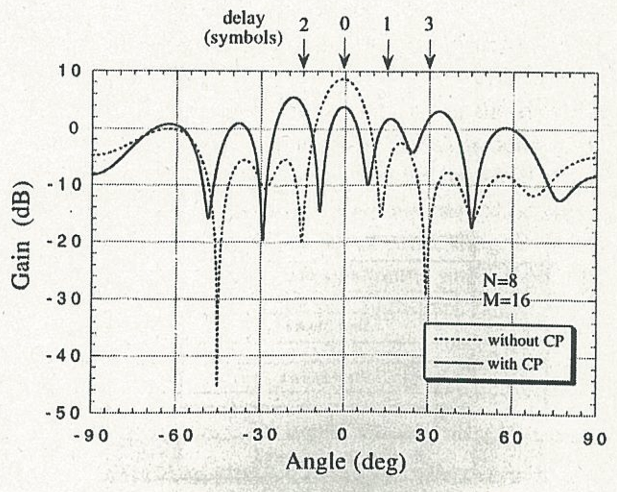

Fig. 5. Antenna patterns of Case 4 (SNR_{in} = 10 dB) in Fig. 4 (subband #1: $f = 0$).

5.2. Waveform equalization function

Since analysis and synthesis are not possible in spatial signal processing when the advanced signal and the delayed signal of the desired signal from the same direction are incident simultaneously, the equalizer operates only in the time domain (or the frequency domain). We show the result for only a single antenna ($N = 1$) in this section. The difference from operating as an array is only the difference of the output SINR due to the difference in the antenna gains. Figure 6 shows the equalization performance of the delay difference where the parameter is the power ratio of the advanced signal to the delayed signal (3, 6 dB). For comparison, the results for the proposed method ($M = 16$, $K = 4$) and for normal SBAA without CP ($M = 16$, $K = 0$) are presented. In the figure, the rapid rise in the portion in one symbol of the delay difference is caused by the coherent synthesis that makes the two signals have the same phase. We would like to compare parts with at least a 1-symbol delay difference. By inserting $K = 4$ CPs, a significant improvement is obtained until a 3-symbol delay difference.

Although an adaptive array is not used on the receive side and only subband signal processing is performed as explained earlier, we show that a transmission method robust in multipaths can be implemented. Although this method differs from OFDM, it shares the meaning of a transmission method that is robust in multipath fading. In the case of OFDM, rigorous linearity for the transmission system is necessary in order to keep wave form shape having large amplitude variations. The proposed method is ordinary constant envelope signal transmission and does not have these restrictions, and its configuration has many advantages. Therefore, the scope of this paper, in which the transmission scheme functionally operates as SBAA, is surpassed, and a general transmission method can be expected.

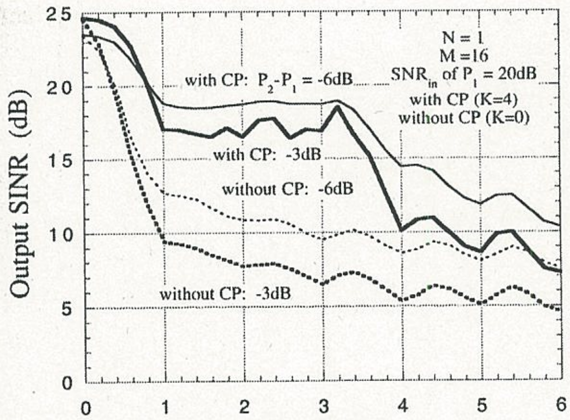

Fig. 6. Waveform equalizing function for two signals with delays from the same direction.

5.3. Interference suppression function

If there are R interference signal sources where each one has a delay difference of at least one symbol and S waves arrive, and the interference is suppressed only by space domain signal processing, in principle, the system requires at least $RS + 1$ antenna elements. If the effects of each delayed signal can be adequately equalized, the number of antenna elements can be reduced to $R + 1$. Below, we show that adequate performance is achieved for multipath inputs that surpassed the degrees of freedom of the antenna.

Figure 7 shows the angle and delay of the incident signal for the evaluation. The desired signal (D) and the interference signals (I_1, I_2, I_3) are constructed by three signals having different values to represent the spread of the arrival angles and delay spread caused by the multipaths. The incident angle is represented by the number at the tip of the arrow. The delay is represented by the number in the center of the arrow (unit: symbols). The strengths of the incident signals are the same with SNR_{in} of 10, 0, and −10 dB. The cyclic prefix is inserted only in the desired signal. The four cases assumed are Case 1: D only (3 signals), Case 2: $D + I_1$, Case 3: $D + I_1 + I_2$, and Case 4: $D + I_1 + I_2 + I_3$.

Figure 8 shows the simulation results. It is seen that a significant improvement is obtained in the characteristics by inserting the CP, and this improvement is conspicuous for a large SNR_{in}. In Case 4, the inputs ($4 \times 3 = 12$) greatly exceed the degrees of freedom of the antenna ($N - 1 = 7$). Nevertheless, the degradation is extremely small. This occurs because the signal processing in the time domain (in the frequency domain for SBAA) equalizes the distortion accompanying the delay. The degrees of freedom required in the array become the number of signal sources + 1 (5 in Case 4), and the degrees of freedom are sufficient in interference suppression. This advantage is generally shared by space-time domain signal processing adaptive arrays.

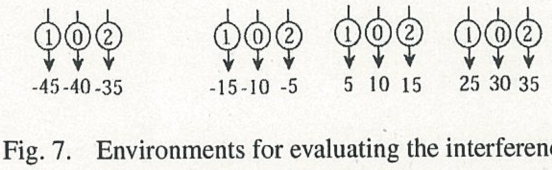

Fig. 7. Environments for evaluating the interference suppression characteristics.

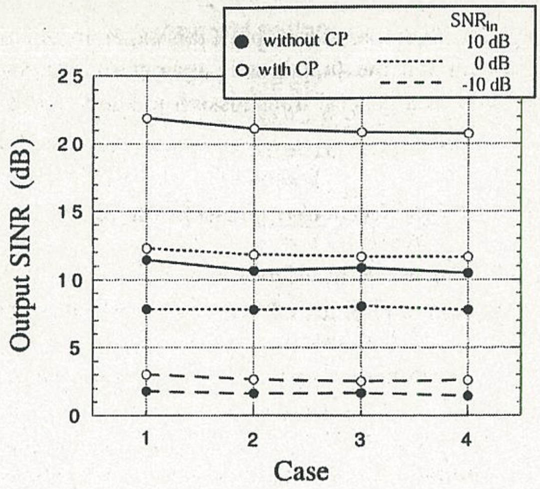

Fig. 8. Interference suppression characteristics.

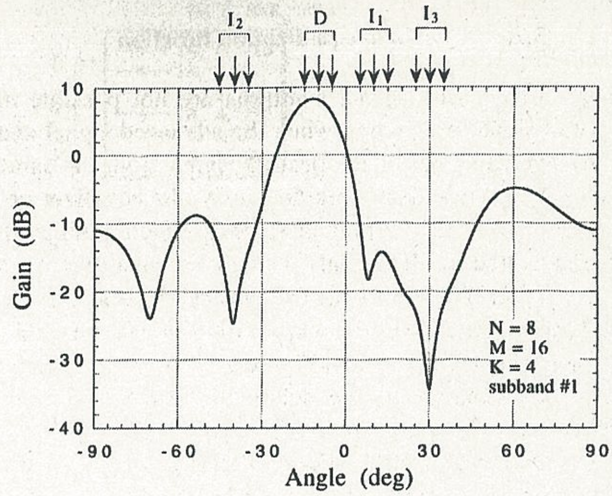

Fig. 9. Antenna pattern after weight convergence in Case 4 of Fig. 8.

In Fig. 8, the SINRs when CP is inserted in Case 1 (○) are 21.9, 12.3, and 3.0 dB for SNR_{in} = 10, 0, and –10 dB, respectively. SNR_{in} + 10 log$N$ (= 9 dB) is expected for the input of each signal, but the SINRs become 19, 9, and –1 dB. The values described above are 2.9, 3.3, and 4.0 dB higher. This means that even if the arrival angle difference is relatively small, frequency domain signal processing functions well. And the desired multipath signals are equalized and incorporated for a delay spread up to the GI length, and the multipath interference is equalized and removed.

Figure 9 shows the antenna pattern (f = 0 subband) obtained when the CP is inserted in Case 4, SNR_{in} = 10 dB. A deep null cannot be produced for each interference wave because each of the three signals of each source becomes one coherent signal by the equalization function of the antenna and acts to cancel out entirely. Figure 10 plots the QPSK demodulated signals (signal points with the optimum sample timing of the reference signal, 3800 points) in the same case (Case 4, SNR_{in} = 10 dB, CP insertion). We can verify extremely good characteristics in spite of an extremely severe environment.

5.4. Convergence characteristics

The previous simulation results presented values having adequate convergence. Specifically, the simulation was conducted on all of the 100 blocks (= 1600 symbols). Here, we present the actual convergence characteristics. For example, we examine Cases 1 and 4 in Fig. 9 described in the previous section (SNR_{in} = 10 dB in both cases). An adaptive algorithm is SINR and determines the weights again each time the block data increases by one.

Figure 11 shows the results. It is seen that convergence reaches an error of 1 dB or less in nearly 50 blocks. Usually, the number of iterations necessary to reach convergence for SMI and RLS are said to be about several times the number of weights to be determined. This result is nearly the same.

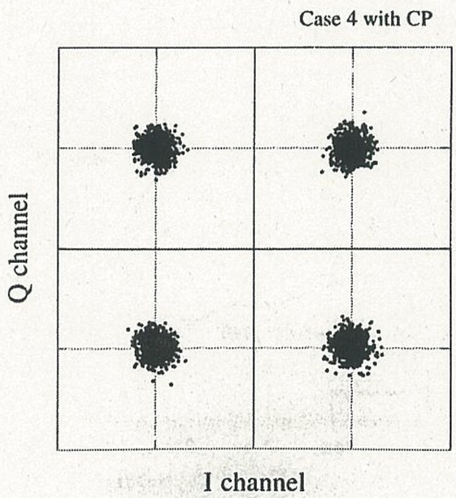

Fig. 10. Constellation of demodulated signal points (3800 points) in Fig. 9.

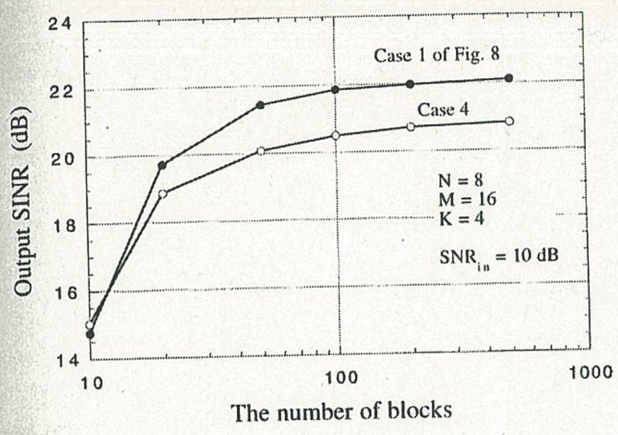

Fig. 11. Convergence characteristics.

6. Conclusion

The scheme of the cyclic prefix used in OFDM was introduced into an ordinary data transmission scheme. By doing this, the ability to incorporate delay signals, which is a weakness of the subband signal processing adaptive array (SBAA), demonstrated adequate ability to absorb within the delay range of the inserted guard interval. As in the future image of providing software radio, this kind of combination will probably demonstrate its strength in the era of transmission schemes capable of freely adapting to the environment. In this paper, we described both data transmission scheme with an inserted cyclic prefix and SBAA. Wider application can be expected by using the general relationship between the former and subband signal processing (= simplified data transmission of OFDM robust to multipaths) as well as SBAA.

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