Adaptive receiving system with identifier quantizer and estimator for fading channel

Réception adaptative avec quantification de la réponse du canal de transmission

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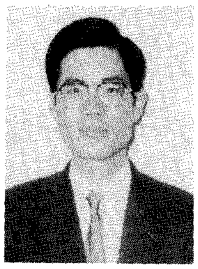

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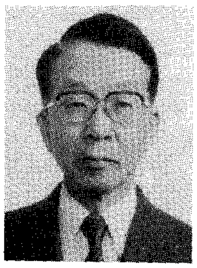

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SUMMARY

An adaptive receiving system which estimates a transmission signal by using Kalman filters has been proposed for a fading channel which is represented by a multipath continuum. This paper presents an improved system composed of an identifier, a quantizer, and an estimator. The identifier estimates the transfer function of the fading channel. The quantizer quantizes the parameters of the transfer function and stores the system parameters to use in the estimator.The estimator extracts the transmission signal by using parallel Kalman filters and a classification mechanism. A method for preventing degradation of system performance due to unexpected changes of the transmission characteristics of the fading channel is also proposed. The validity of the proposed system is demonstrated by comparison of the performance characteristics of the proposed system with those of the adaptive transversal equalizer for M-level quadrature amplitude modulation system M-level quandrature amplitude modulation system.

KEY WORDS

Adaptive receiving system, Fading channel, Parallel Kalman filters, Multipath continuum, Identifier, M-QAM system.

RÉSUMÉ

Considérant le cas d'un canal de transmission dont la réponse évolutive peut être assimilée à un continuum de trajets multiples, les auteurs proposent un récepteur pour estimer le signal transmis (affecté de fading). Il opère par estimation de la réponse du milieu (identifacteur), par quantification des paramètres de cette réponse et mémorisation (quantificateur) pour estimer le signal transmis en utilisant une réponse quantifiée proche de la réponse réelle (estimateur). Un mécanisme proposé par les auteurs permet de préserver les performances en cas de variation de la réponse. Dans la dernière partie, la comparaison des performances obtenues avec celles d'un égaliseur adaptatif transversal dans le cas d'une modulation d'amplitude à M niveaux (M-QAM) met en évidence la supériorité du système proposé.

MOTS CLÉS

Récepteur adaptatif, fading, filtre de Kalman, parallélisme, trajets multiples, estimation, quantification, modulation M-QAM.

1. Introduction

Reducing the degradation of communication quality due to fading is an important problem in radio communication. Diversity and equalization are the fundamental techniques for restoring the effects of selective fading [1, 2]. Recent advances in equalization techniques can significantly reduce the need for space diversity reception for digital radio systems. The use of adaptive equalizer is based on the modeling of the fading channel [3, 4]. Several statistical models of multipath fading have been proposed [5-7]. These models represent the multipath fading by a channel transfer function with time-varying parameters. Mutuura-Morikawa-Endo's model was applied to an adaptive receiving system based on Kalman filters [5]. Rummler's model [6] and Greenstein-Czekaj's model [7] were utilized to analyze multipath fading outages in terrestrial digital radio systems [8-10].

In adaptive equalization, there is a tradeoff between the convergence of adaptation algorithms, the computational cost for each adjustment and the implementation complexity. Since the period of selective fading is in the order of a few seconds to several tens of seconds, the convergence rate of adaptation algorithms for the variations of the channel transfer function is the most important factor in the fading problem. From the comparative performance results for several adaptive equalization algorithms by Kumar and Moore [11], the equalizer taps achieve about 1% mean-square error in about 200 iterations for a recursive least squares algorithm, which has the highest convergence rate in their simulations, and the estimation error of signal achieves steady state after about 500 iterations. Thus, the recursive least squares algorithm requires the training sequence of 200 samples to estimate the equalizer tap parameters before changing the channel transfer function. Since the random fluctuation of the fading channel does not permit an effective transmission of the training sequence, it is not advisable to use the training sequence. Furthermore the occurrence of radio outage cannot be avoided in the system design and the outage time depends on the ratio of the training sequence to the transmission signal and on the convergence rate of adaptation algorithm. In our adaptive receiving system [5], the training sequence is not necessary, and the estimation error of signal achieves steady state within 10 to 30 iterations, but sets of predetermined equalizer parameters are required.

In this paper, we extend the previously proposed system to a system with an identifier, a quantizer and an estimator. The identifier estimates the transfer function of the fading channel using a test signal or an estimated signal. The parameters of this transfer function are quantized and stored. The transmission signal is estimated with parallel Kalman filters. After initial characterization of the channel using a test signal, the identifier and the estimator run simultaneously, allowing the estimated transmission signal to be used to estimate the transfer function of the fading channel. If the new estimated parameters exist outside the predetermined fluctuation range, these parameters are quantized and stored in the system. The predetermined range is then extended without a test signal. This amounts to a major expansion of the system.

The validity of the proposed system is demonstrated by using M-level quadrature amplitude modulation (M-QAM) signals where the data of fading characteristics are obtained from short-wave radio propagation measurements performed by Kokusai Denshin Denwa (KDD) Co., Ltd. in March 1972 [12].

2. Model of transmission system

In long-distance propagation, as a result of reflection and scattering in the medium, the electromagnetic energy may travel from the transmitting point to the receiving point via many paths, each having a different propagation time. When the differences in propagation time are distributed uniformly within a certain range, the multiple paths are considered to form a multipath continuum [13]. In the communication between two fixed radio stations, although the multipath continuum fluctuates randomly depending on various factors such as the state of the ionosphere and atmosphere, it is expected that there exists a specific fluctuation range. Hence, by quantizing the transmission characteristics within this fluctuation range, the multipath continuum can be approximated by sets of impulse responses which can be treated as being constant over short intervals of time. The quantization of the transmission characteristics decreases in estimation accuracy of the transmission signal. The relationship between the quantized parameters and the performance of the adaptive receiving system for analog signal has been investigated [14].

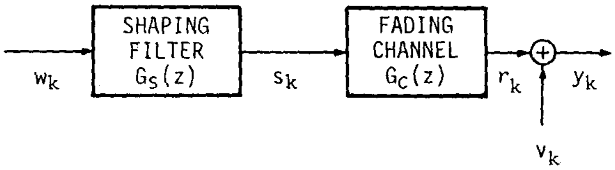

Fig. 1. – Model of transmission system.

We summarize the transmission model with the transmission signal and the fading channel proposed by the authors [5] below. Figure 1 shows the model of the transmission system. The transmission signal s_k is generated by a shaping filter with a transfer function $G_s(z)$

$$G_s(z)=\frac{\beta_1 z^{-1}+\ldots+\beta_n z^{-n}}{1+\alpha_1 z^{-1}+\ldots+\alpha_n z^{-n}}. \tag{1}$$

The transmission signal s_k is distorted during propagation by the fading channel whose transfer function $G_c(z)$ has an impulse response of the form

$$G_c(z)=h_1 z^{-1}+h_2 z^{-2}+\ldots+h_\sigma z^{-\sigma}. \tag{2}$$

The received signal y_k is the channel output r_k with an additive white noise v_k

$$y_k=r_k+v_k=h_1 s_{k-1}+\ldots+h_\sigma s_{k-\sigma}+v_k, \tag{3}$$

where v_k is an uncorrelated sequence with variance R,

$$E[v_i]=0, \qquad E[v_i v_j]=R\,\delta_{ij}.$$

The state equation and the observation equation for this transmission model can be represented by

$$x_k=\Phi x_{k-1}+\Gamma w_{k-1} \tag{4}$$

$$y_k=H x_k+v_k, \tag{5}$$

where the $n+\sigma\times n+\sigma$ state transition matrix Φ, the $n+\sigma\times 1$ input vector Γ and the $1\times n+\sigma$ observation vector H are defined by

$$\Phi=\left[\begin{array}{cccccc|ccc}
-\alpha_1 & 1 & 0 & \ldots & 0 & 0 & 0 & \ldots & \\
 & 0 & 1 & \ldots & 0 & 0 & & & \\
\vdots & & & \vdots & & 0 & & & \\
-\alpha_n & 0 & & & & 0 & & & \\
\hline
1 & 0 & 0 & & & 0 & 0 & 0 & \ldots \\
0 & & & & & & 1 & 0 & 0 \\
0 & & & & & & 0 & 1 & 0 \\
0 & & & & & & & & \vdots \\
0 & 0 & 0 & \ldots & 0 & 0 & 0 & \ldots & 0
\end{array}\right],$$

considered as not varying,

$$\Gamma=\begin{bmatrix}\beta_1\\ \vdots\\ \beta_n\\ \ldots\\ 0\\ \vdots\\ 0\end{bmatrix},$$

$$H=[0,\ldots,0 \,\vdots\, h_1,h_2,\ldots,h_\sigma].$$

In (4), the input sequence w_k is assumed to be an uncorrelated sequence with variance Q,

$$E[w_i]=0, \qquad E[w_i w_j]=Q\,\delta_{ij}.$$

The $n+\sigma$ state vector x_k includes the transmission signal in its elements. Namely the elements $x_{i,k}\,(i=1,\ldots,n+\sigma)$ of state vector x_k correspond to the transmission signal as follows:

$$\begin{cases} x_{1,k}=s_k \\ x_{n+j,k}=s_{k-j}, \qquad j=1,2,\ldots,\sigma. \end{cases} \tag{6}$$

The key step in characterizing $G_c(z)$ during multipath fading periods is to find a suitable observation vector H which contains a set of adjustable impulse response $(h_1,h_2,\ldots,h_\sigma)$. The delay number σ of the impulse response depends on the maximum value of delay difference and the carrier frequency. The maximum value of delay difference for microwave radio propagation predicted by Ruthroff [15] is

$$\tau_{max}=3.7\left(\frac{D}{20}\right)^3 ns, \tag{7}$$

where D is the path length in miles. The multipath delay difference τ and the variation of magnitude h result in the phase distortion and the amplitude distorsion, respectively. If the delay difference τ and the magnitud for each path are fixed, the number N of the observation vector H is equal to the number of paths. Since the delay difference τ and the magnitude h are time-varying, however, the adaptive receiver requires the sets of observation vector whose number is larger than the number of paths.

Fading occurs as a result of random changes of the impulse response contained in the observation matrix H with the probability density $p(H)$.

3. Adaptive receiving system

The block diagram in Figure 2 shows the proposed adaptive receiving system for the fading channel. The estimator and the identifier require real time computations, while the quantizer does not. The estimator and the identifier run either independently or simultaneously. First, the identifier estimates the transfer function of the fading channel using a test signal. This initial estimate is made with the estimation idle. The quantizer collects the estimated parameters and determines a fluctuation range of these parameters during the test. The quantizer then quantizes these parameters and stores the system parameters to use in the estimator. The estimator is composed of parallel

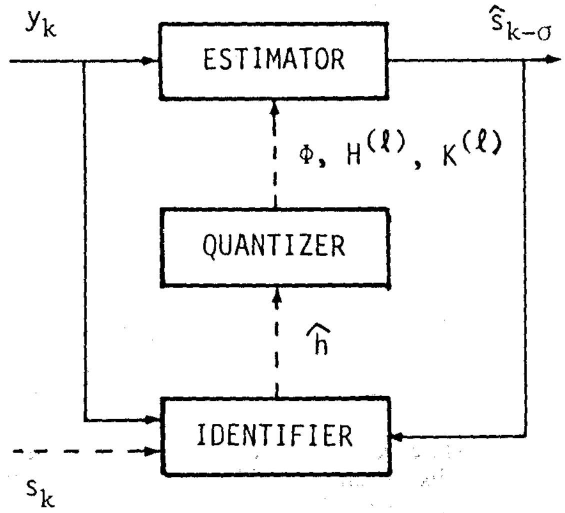

Fig. 2. – **Block diagram of adaptive receiving system.**

Kalman filters and a classification mechanism which recognizes the pattern of the parameters of the impulse responses representing the multipath continuum. When the estimator and the identifier run simultaneously, the estimated transmission signal is used to estimate the transfer function of the fading channel. If the estimated parameters exist outside the predetermined fluctuation range, these new parameters are quantized and stored. The predetermined range is then extended without a test signal.

A. Estimation of the channel characteristics

The channel characteristics are represented by a moving-average type expression consisting of the impulse response of the fading channel. The initial estimate made with the test signal s_i is dealt with in the following manners. From (3), we have

$$y_k = S_{k-1}^T \Psi + v_k, \tag{8}$$

where

$$S_{k-1}^T = [s_{k-\sigma}, \ldots, s_{k-1}],$$
$$\Psi = [h_\sigma, \ldots, h_1].$$

The estimate of the impulse response can be obtained by the following recursive formula [16]:

(9)
$$\hat{\Psi}_{k+1} = \hat{\Psi}_k + M_k S_k (S_k^T M_k S_k + 1)^{-1} (y_{k+1} - S_k^T \hat{\Psi}_k)$$

(10)
$$M_k = M_{k-1} - M_{k-1} S_{k-1} (S_{k-1}^T M_{k-1} S_k + 1)^{-1} S_{k-1}^T M_{k-1}.$$

When the test signal s_i is absent, s_i is replaced by the estimated transmission signal $\hat{s}_i$ which is the output of the estimator. This estimate is consistent but biased due to the estimation error of the transmission signal.

B. Quantization of the transmission characteristics

To construct the adaptive receiving system, it is necessary to quantize the transmission characteristics of the fading channel. The block diagram in Figure 3 shows the quantization procedure. The transmission characteristics are quantized by examining characteristics of the fading channel. This is done by determining which impulse response values appear most frequently among the many impulse responses estimated beforehand. As the order of each impulse response increases, the operation becomes enormous. To avoid this problem, once an impulse response is estimated, it is approximated by an autoregressive (AR) type transfer function with reduced order. The translation from the impulse response to the AR parameters is given by

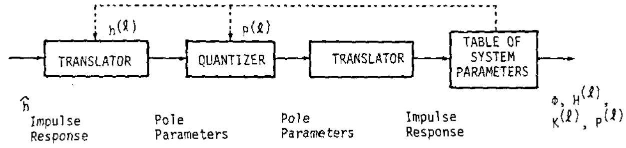

Fig. 3. – **Block diagram of quantizer.**

$$\begin{bmatrix} \gamma_m \\ \gamma_{m-1} \\ \vdots \\ \gamma_1 \end{bmatrix} = - \begin{bmatrix} h_1 & \cdots & h_m \\ h_2 & \cdots & h_{m+1} \\ \vdots & & \vdots \\ h_m & & h_{2m-1} \end{bmatrix}^{-1} \begin{bmatrix} h_{m+1} \\ h_{+2} \\ \vdots \\ h_{2m} \end{bmatrix}, \tag{11}$$

where γ_i's denote AR parameters of transmission characteristics and $2m \leqq \sigma$. Given an impulse response of the system, h_i, $i=1, 2, \ldots$, the estimation of γ_i's with minimum order is the minimum realization problem [17]. The order m of the minimum realization is the order of the associated Hankel matrix in (11). The AR parameters are also transformed into pole parameters which are the roots of the polynomial. The quantization is then carried out on the basis of the pole-distribution. In this method, since the pole is represented by its argument and radius which correspond to frequency and bandwidth, the structure of the fading channel can be understood easily. Moreover, since a lower order model than the impulse response is used, the quantization can be implemented at a lower bit rate (which corresponds to the number of Kalman filters) than that of the impulse response. This quantization method proposed by one of authors was used in the parameter quantization of speech coding [18].

N sets of impulse responses are calculated again from the quantized parameters. These are contained in the observation matrix H as follows:

$$\begin{aligned} H^{(1)} &= [0, \ldots, 0 \vdots h_1^{(1)}, h_2^{(1)}, \ldots, h_\sigma^{(1)}] \\ H^{(2)} &= [0, \ldots, 0 \vdots h_1^{(2)}, h_2^{(2)}, \ldots, h_\sigma^{(2)}] \\ &\vdots \\ H^{(N)} &= [0, \ldots, 0 \vdots h_1^{(N)}, h_2^{(N)}, \ldots, h_\sigma^{(N)}]. \end{aligned} \tag{12}$$

Once the system parameters, Φ, Γ, $H^{(l)}$, Q and R are determined, the Kalman gain can be computed readily beforehand. The system parameters Φ, $H^{(l)}$, $K^{(l)}$, $l=1, \ldots, N$, for the parallel Kalman filters are stored in the system.

When the estimated impulse responses are obtained by using the estimated transmission signal $\hat{s}_k$, the quantizer increases the number of system parameters using the following two step procedures.

First, the translator compares the estimated impulse response $\hat{h}_i$, $i=1,\ldots,\sigma$ and the prestored impulse responses $h_i^{(l)}$, $i=1,\ldots,\sigma$, $l=1,\ldots,N$.

$$\text{If } \min(\mathrm{DH}) = \min \left(\frac{\sum_{i=1}^{\sigma} |\hat{h}_i - h_i^{(l)}|}{\sum_{i=1}^{\sigma} |\hat{h}_i|} \right) > \varepsilon_1, \quad l=1,\ldots,N,$$

then the estimated impulse response is transformed into pole parameters $P_i=(x_i \pm jy_i)$, $i=1,\ldots,p=m/2$. Second, the quantizer compares the pole parameters P_i, $i=1,\ldots,p$ and the prestored pole parameters $P_i(l)$, $i=1,\ldots,p$, $l=1,\ldots,N$.

$$\text{If } \min(\mathrm{DP}) = \min \frac{\sum_{i=1}^{p} \left(\sqrt{(x_i - x_i^{(l)})^2 + (y_i - y_i^{(l)})^2}\right.}{\sum_{i=1}^{p} \sqrt{x_i^2 + y_i^2}} > \varepsilon_2,$$

$l=1,\ldots,N$, then the pole parameters $P_i^{(N+1)}$, $i=1,\ldots,p$ are quantized and transformed back into impulse response $h_i^{(N+1)}$, $i=1,\ldots,\sigma$. As a result, the quantizer increases the prestored system parameters and extends the range in the neighborhood of the prestored fluctuation range.

C. Estimation of the transmission signal

Given the system parameters Φ, $H^{(l)}$, $K^{(l)}$, each Kalman filter returns an estimated state vector of the form

$$(13) \qquad \hat{x}_k^{(l)} = \Phi \hat{x}_{k-1}^{(l)} + K^{(l)} (y_k - H^{(l)} \Phi \hat{x}_{k-1}^{(l)}).$$

The estimation of the state vector x_k results in the simulataneous estimation of $\sigma+1$ transmission signals (*i.e.*: $b\hat{s}_{k-i}$, $i=0,1,\ldots,\sigma$). Our pilot study [5] revealed that the oldest $\hat{s}_{k-\sigma}$ of the estimates is the most accurate. Thus the σ-step delayed signal $\hat{s}_{k-\sigma}$ is used as the estimated value.

The adaptive estimator consists of a bank of N Kalman filters where each filter is matched to each possible member of $H^{(l)}$ with outputs $\hat{s}_k$ weighted according to the *a posterior* probabilities $p(H^{(l)}|Y_k)$, $l=0,\ldots,N$ where $Y_k=\{y_1,y_2,\ldots,y_k\}$ as

$$(14) \qquad \hat{s}_k = E[\hat{s}_k^{(l)}] = \sum_{l=0}^{N} \hat{s}_k^{(l)} p(H^{(l)}|Y_k) = \sum_{l=0}^{N} \hat{s}_k^{(l)} p_k^{(l)}.$$

In (14), $l=0$ denotes nonfading or fading where a suitable quantized transfer function of the fading channel is not stored in the system. Thus the estimated transmission signal is not estimated using the transfer function of the fading channel. When $p_k^{(0)}=1$, $\hat{s}_k=y_k$. This occurs when all Kalman filter outputs increase the distortion due to nonfading or due to an expected change in the fading channel. Thereby the estimated transmission signal is not degraded for any fading below that of the case in which no processing occurs.

To adapt the system to the random changes of channel characteristics due to fading, we apply the partition theorem [19, 20]. The transmission signal is estimated adaptively with the detected fading state using the following method.

An innovation sequence $v_k^{(l)}$, $l=1,\ldots,N$ [21] generated from each Kalman filter is used adaptively to detect the state of channel. This sequence is a part of equation (13), and is given by

$$(15) \qquad v_k^{(l)} = y_k - H^{(l)} \Phi \hat{x}_{k-1}^{(l)}.$$

Using this sequence, at every instance of input, the *a posteriori* probability $p_k^{(l)}$ for each Kalman filter is given by

$$(16) \qquad p_k^{(l)} = \frac{p_{k-1}^{(l)} L(Y_k|H^{(l)})}{\sum_{j=0}^{N} p_{k-1}^{(j)} L(Y_k|H^{(j)})},$$

where $L(Y_k|H^{(l)})$ denotes the likelihood of a previously received signal sequence Y_k. It is related to its conditional likelihood ratio $L(y_k|Y_{k-1},H^{(l)})$ as follows:

$$(17) \qquad L(Y_k|H^{(l)}) = L(Y_{k-1}|H^{(l)}) L(y_k|Y_{k-1},H^{(l)})$$

$$(18) \qquad L(y_k|Y_{k-1},H^{(l)}) = \frac{c_k^{-1/2} \exp(-(1/2) c_k^{-1} v_k^2)}{b_k^{-1/2} \exp(-(1/2) b_k^{-1} y_k^2)},$$

where b_k and c_k are the variances of the received signal y_k and the innovation sequence v_k, respectively, and are given by

$$(19) \qquad b_k = \frac{k-1}{k} b_k + \frac{1}{k} y_k^2$$

$$(20) \qquad c_k = \frac{k-1}{k} c_{k-1} + \frac{1}{k} v_k^2.$$

It must be noted that in the case of $l=0$ in (15), the received signal is regarded as the transmission signal, $v_k=y_k$ and the likelihood ratio in (18) is always 1.

The estimator constructed by (13)-(20), consists of two parts: The filtering part composed of the parallel Kalman filters, and the adaptation part composed of the classification mechanism. The block diagram of the estimator is shown in Figure 4.

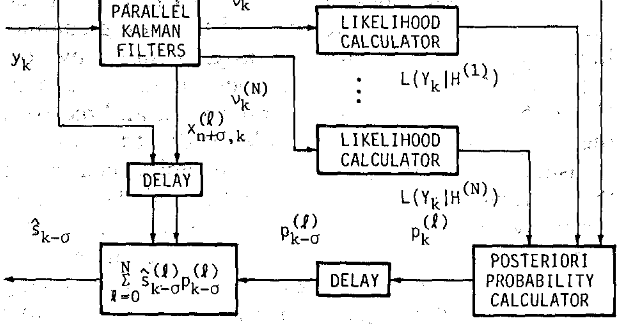

Fig. 4. — Block diagram of estimator.

4. stimulation

The performance of the receiving system should be evaluated from the viewpoint of follow-up performance for the variation of the transmission characteristics, both in space and in time. The former is represented by the fluctuation range of the fading channel. The later is represented by the fading period. In order to assess the validity of the proposed system, we compare the performance characteristics of the proposed system and the adaptive transversal equalizer for M-QAM system.

The data of fading characteristics was acquired by KDD during a test of the propagation characteristics of short-wave transmission [12]. Both the transmitted and received reproduction of a composite sinusoidal signal with eight frequency components (320, 440, 580, 730, 880, 1,040, 1,230 and 1,420 Hz) were recorded and sampled at a frequency of 3 kHz. The carrier frequency was 8.217,1 MHz. Figure 5 shows

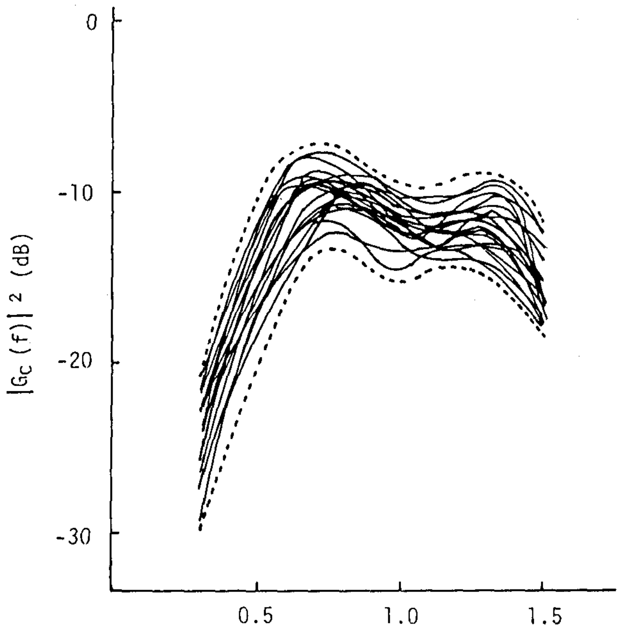

Fig. 5. – Relative frequency characteristic of the transmission channel at baseband.

the baseband frequency characteristics of the total transmission system obtained from the estimated impulse responses. As a result of multipath and time-varying characteristics of the transmission channel, the transmission characteristics of the fading channel fluctuate with time within the range bounded by dotted lines. Although the frequency characteristics of Figure 5 contains that of the transmitter and receiver, the fluctuation range coincides with that of the transmission characteristics of the fading channel if the characteristics of the transmitter and the receiver are fixed. By applying (11) for the estimated impulse response, the characteristics of the fading channel are approximated by a 4th order transfer function having two sets of poles. Figure 6 shows a typical example of pole locations which are determined by solving for roots of the polynomial whose coefficients are the

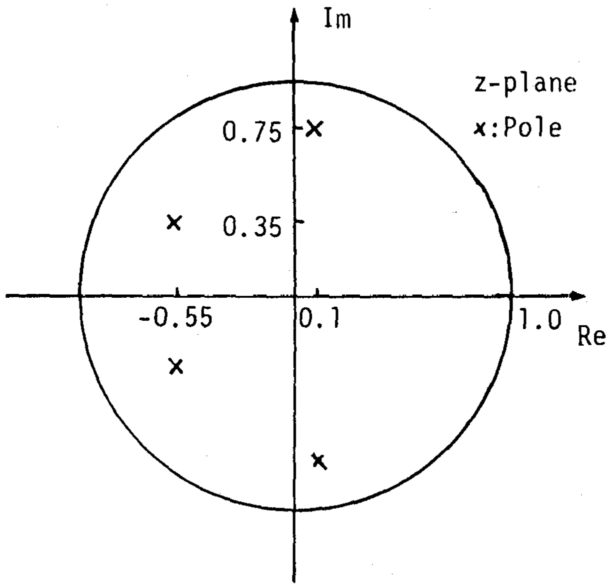

Fig. 6. – Pole locations of the transfer function of the transmission channel.

AR parameters γ_i's. They are located at $0.1 \pm j\,0.75$ and $-0.55 \pm j\,0.35$ on z-plane as shown in Figure 6. Further details of the experiment are described in the reference [12, 14].

The published adaptive equalizers in radio systems designed for operation at an intermediate frequency range (e. g., adaptive slope equalizer and adaptive notch equalizer) or baseband range (e. g., adaptive transversal equalizer) [4]. Since the data is recorded at baseband, the simulation is performed at baseband.

The parameters of the shaping filter in Figure 1 depend on the modulation scheme. Since the baseband transmission signal is M-QAM signal, the shaping filter is neglected in the system so that s_k is w_k. Thus the state transition matrix Φ, the input vector Γ and the observation vector H in (4) and (5) are given by

$$\Phi = \begin{bmatrix} 0 & & & 0 \\ 1 & 0 & & \\ 0 & 1 & & \\ \vdots & & \vdots & \\ 0 & \ldots & 0 & 1\,0 \end{bmatrix}, \qquad \Gamma = \begin{bmatrix} 1 \\ 0 \\ \vdots \\ 0 \end{bmatrix}$$

and

$$H = [h_1, h_2, \ldots, h_\sigma],$$

respectively. These structures of the state equation and the observation equation are equivalent to the adaptive equalizer proposed by the authors for multiroute digital transmission line [22].

The final output of M-level symbols is obtained from the estimated signal $\hat{s}_k$ by passing through a detector with thresholds which correspond to modulation levels. The system performance is evaluated by the bit error rate (BER) and the following signal distortion measures:

$$\text{(21)} \qquad D_s \triangleq 10 \log_{10} \frac{E[S_k^2]}{E[(\hat{s}_k - s_k)^2]}$$

and

(22) $$D_y \triangleq 10 \log_{10} \frac{E[s_k^2]}{E[(y_k - s_k)^2]}.$$

D_s denotes the distortion remaining in the estimated signal and D_y denotes the distortion of the received signal due to fading and additive noise. The fluctuation ranges of the transmission characteristic poles shown in Fig. 6 are assumed to fall within the ranges illustrated in the lower part of Figure 7. It is also assumed that fading is caused by the random fluctuation of these poles within this range. During simulation, we use three fluctuation ranges, (*a*), (*b*) and (*c*), and three fading periods, 500, 1,000 and 1,500 steps. When transmission characteristics are randomly changed after every 500 samples, for example, the fading period is (500) × (sampling period of basehand signal).

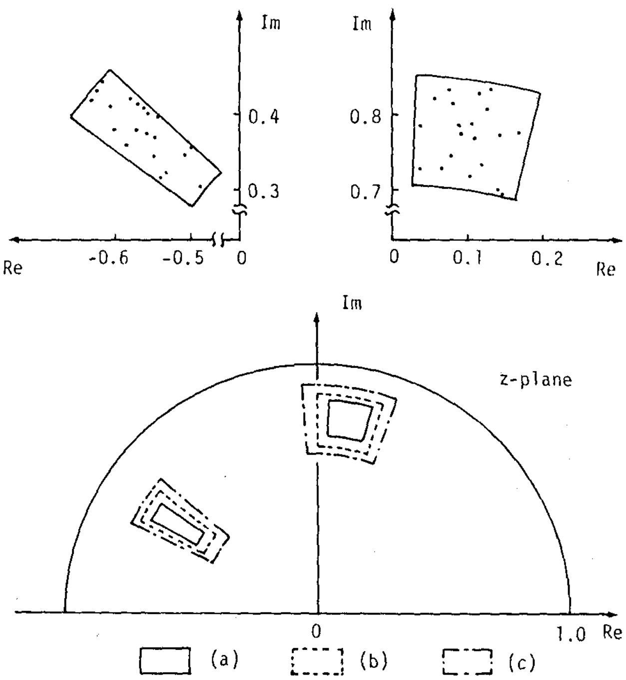

Fig. 7. – **Fluctuation range of the pole locations of the transfer function of the transmission channel.**

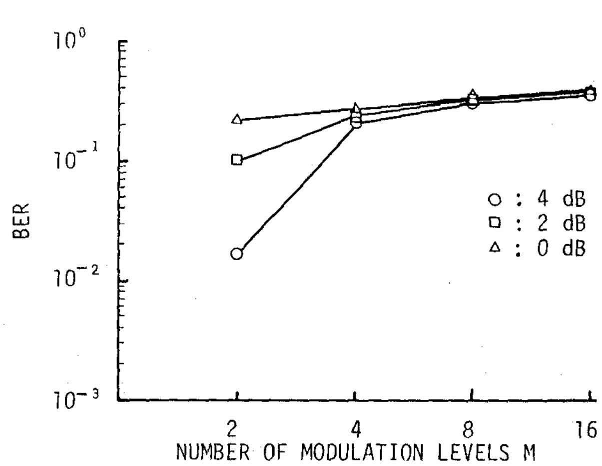

Fig. 8. – **BER as a function of the number of modulation levels M at various D_y.**

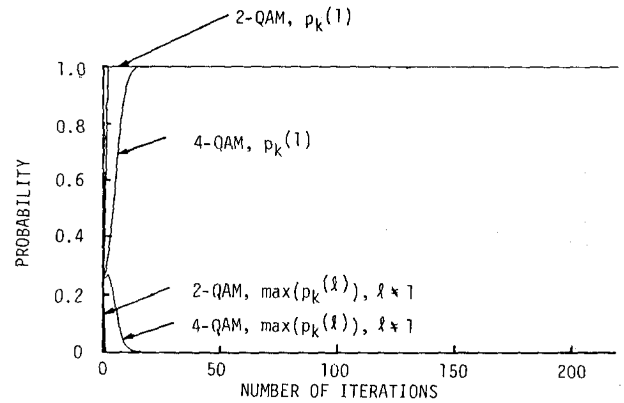

Fig. 9. – **Convergence of *a priori* probability in the proposed system for 2- and 4-QAM systems.**

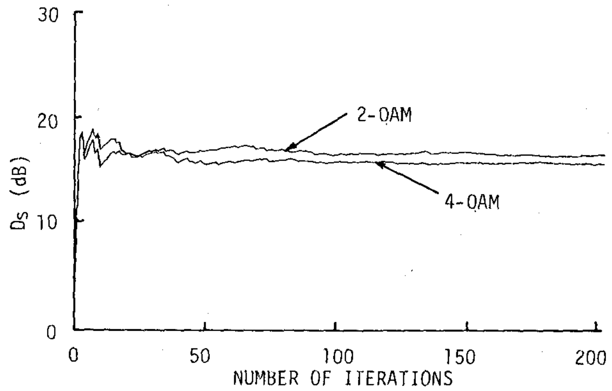

Fig. 10. – **Convergence of D_s in the proposed system for 2- and 4-QAM systems.**

Figure 8 shows the relation between BER as a function of the number of modulation levels M and the transmission distortion D_y of the fading channel. Three levels of D_y, 0, 2, 4 (dB), are obtained from the observed signal y_i with the received signal-to-noise ratio (SNR) = 30 (dB) for the fluctuation range (*a*). The results show a necessity of quantization for every modulation level of M-QAM signals.

We show the convergence characteristics of the proposed system and the adaptive transversal equalizer using the fading channel with the range (*a*). Figures 9 and 10 plot the *a posteriori* probability and D_s of the system with 4 level quantized system parameters for 2- and 4-QAM systems, respectively. The *a priori* probability of Kalman filter with the minimum DH of 0.15 in prestored impulse responses converges to 1 after about 20 iterations, and the signal distortion D_s achieves 16.4 (dB) for 2-QAM system and 15.5 (dB) for 4-QAM system within these iterations, respectively. There is no bit error during the initial transient period in 2-QAM system, while only one error occurs at the second sample in 4-QAM system.

The relation between the upper bound BER_0 of BER and D_s for the number of modulation levels M is given by

(23) $$D_s = \frac{2}{3}(M-1)\ln(2/BER_0)$$

in [10]. From (23), we can find the required D_s under the number of modulation levels M and the desired

BER_0. When BER_0 is 10^{-3} and M is 4, for example, the required D_s is 15.2 (dB). When the receiver knows the value and fluctuation period of the impulse response, the optimum Kalman filter has the maximum D_s of 27 (dB). The performance degradation of the system from the maximum D_s is caused by the quantization and the unknown fading period. The value of DH corresponds to the quantization error between the actual impulse response and the prestored impulse response. The convergence performance of *a priori* probability and D_s is not signigicantly dependent on the modulation level of QAM system since the Kalman filter estimates the signal s_i as an analog signal. The results reveal that the system can minimize the estimation error during the transient period.

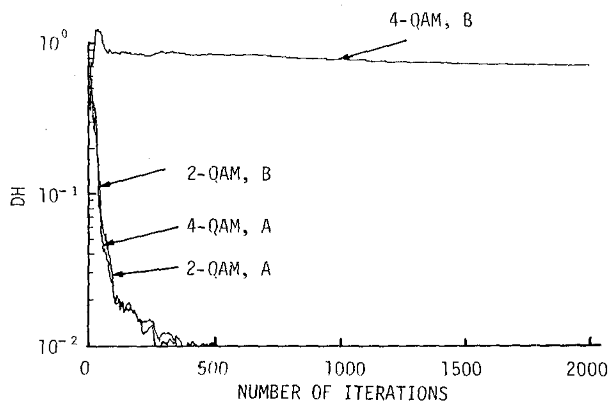

Fig. 11. — **Convergence of DH in the adaptive transversal equalizer for 2- and 4-QAM systems. The taps of transversal equalizer are adjusted by the test signal in the case A and the estimated signal in the case B, respectively.**

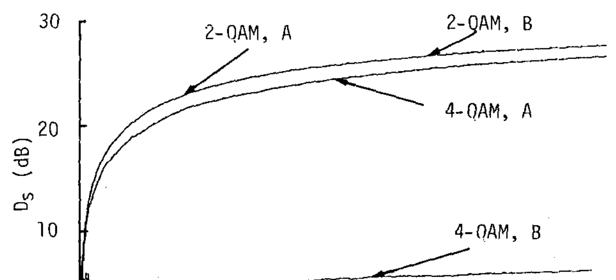

Fig. 12. — **Convergence of D_s in the adaptive transversal equalizer for 2- and 4-QAM systems. The taps of transversal equalizer are adjusted by the test signal in the case A and the estimated signal in the case B, respectively.**

Figures 11 and 12 plot the convergence of DH and D_s for the adaptive transversal equalizer, respectively. The adaptive transversal equalizer is examined in two cases. In the first case (A), the taps of transversal equalizer are estimated by the recursive least squares algorithms (9) and (10) using a test signal during the initial transient period. In the second case (B), a clipped version of estimated signal $\hat{s}_k$ by passing through a detector is used in the estimation algorithm. If there is no bit error, therefore, the clipped signal is equivalent to the test signal. This scheme in the algorithm is called as quantized-state scheme [11]. The number of taps is fixed at 20 since the estimated impulse response converges to a very small value (*i.e.*, $|h_i| < 0.005$, $i > 20$) at delays above 20. In the case A, the equalizer taps achieve DH of 0.01 within about 200 iterations for both 2- and 4-QAM systems. In the case B, the equalizer taps of 4-QAM system achieve DH of 0.69 so that BER is 0.15, while the 2-QAM system has the same convergence rate as that obtained for the case A since only three bit errors occur at the 2th, 3th and 4th samples. Once DH becomes above 0.3, the estimation algorithm can not adjust the taps using estimated signal since it increases parameter error.

A study of two figures reveals that the adaptive transversal equalizer requires the test signal of about 200 samples during the initial transient period and the adaptive transversal equalizer of multi-level QAM system without the test signal can not be applied to this fading channel. Therefore, the proposed system surpasses the adaptive transversal equalizer in the ability of equalization for these fading characteristics.

We show follow-up performance of the proposed system for the variations of the transmission characteristics in space. Adaptive reception is performed with six quantization levels of $1 \sim 6$, each of which is applied to the fluctuation ranges, (*a*), (*b*) and (*c*) with fading period of 1.500 steps. The quantization of the pole locations is accomplished by the algorithm described in Section 3.B using test signals for each fluctuation range. The transmission characteristics are randomly changed with respect to argument and radius of each pole within predetermined fluctuation range.

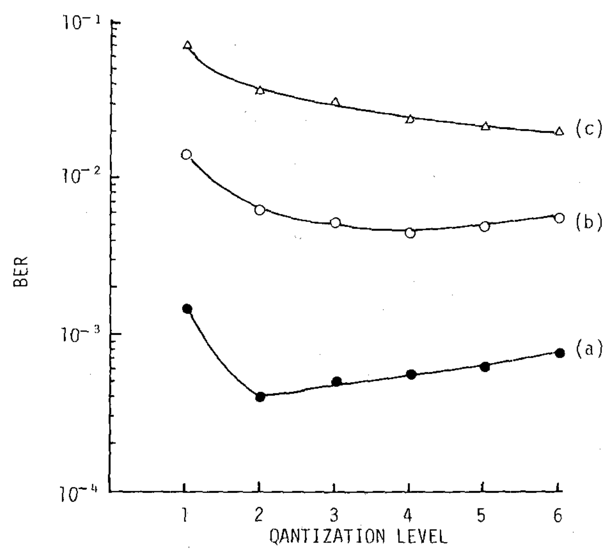

Fig. 13. — **BER as a function of the quantization level for various fluctuation ranges of the transmission characteristics.**

The results of adpative reception for 4-QAM signal are shown in Figure 13. In the range (*a*), the minimum BER is 0.000,4 at 2 level quantization. The increase of BER at the quantization levels above 2 is caused by the error of the classification mechanism. The larger the number of quantization levels in Kalman filter bank, the longer becomes the interval to achieve steady state error. The increase of BER by increasing one set of system parameters in Kalman filter bank is about 0,000.1. In the ranges (*b*) and (*c*), BER is large. This is caused by the fact that the small expansion of the range in poles results in the large expansion of the range in impulse responses. The results show that the prestored system parameters generated by the test signals are not suitable for the ranges (*b*) and (*c*). Thus we investigate the relation among BER, DH and DP for these ranges.

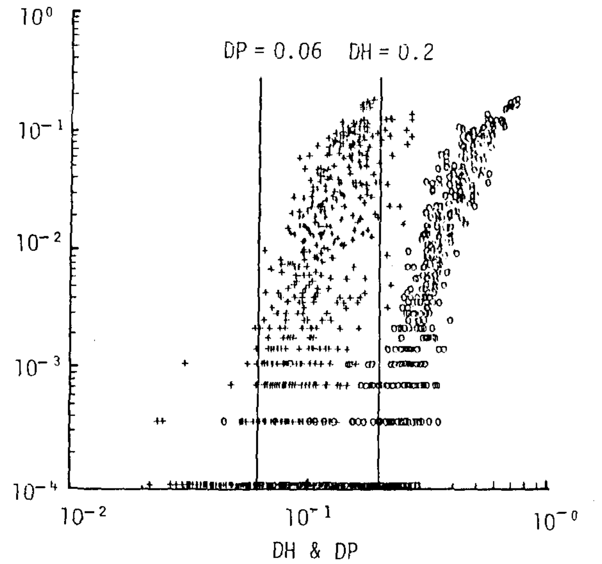

Fig. 14. — BER as a function of DH (•) and DP (+) for every steady state intervals.

Figure 14 plots various BER results for every steady state intervals with 1,500 samples obtained from each set of Kalman filters. In this figure, the sign 0 plots the relation between BER and DH, and the sign + plots the relation between BER and DP. The value of DP is relatively small comparing with the value of DH. When DH is smaller than 0.2 which corresponds to DP of about 0.06, BER is smaller than 10^{-3}. The ratio of impulse response whose DH is smaller than 0.2 is 47.1%, 17.2% and 5.4% for the ranges (*a*), (*b*) and (*c*), respectively. The classification mechanism selects the impulse response whose mean value of DH is 0.155, 0.229 and 0.327 for the ranges (*a*), (*b*) and (*c*), respectively. When the desired BER is 10^{-3}, at least, one set of parameters where DH < 0.2 is required at every intervals. That can be accomplished by the algorithm described in Section 3.B using the following thresholds. When the range is (*b*) and the prestored parameter level is 2, by using the thresholds of $\varepsilon_1 = 0.2$ and $\varepsilon_2 = 0.06$, the identifier and the quantizer extend the predetermined fluctuation range. The final result of BER for the range (*b*) is 0.92×10^{-3} and the quantization level is 6. The result suggests that this mechanism reduces the test working time for determining the predetermined fluctuation range of the fading channel.

The wider the fluctuation ranges of the transmission characteristics, the larger becomes the number of impulse responses that must be stored in the system in order to attain the specified performance. However, it is found from Figure 13 that BER is saturated with a specific quantization level in a given fluctuation range. This phenomenon is caused by the fact that the increase of quantization level results in the decrease of the minimum DH in the prestored impulse responses at each interval but it increases the error of the classification mechanism.

When the adaptive transversal equalizer is applied in the range (*a*), BER of 10^{-3} requires the test signal of about 100 samples at every initial transient periods. Therefore, 6.7% of the transmission signal is required as the test signal.

We show follow-up performance of the proposed system for the variations of the transmission characteristics in time. A simulation is executed using the fluctuation range as shown in Figure 7 (*a*) with three fading periods 1.500, 1.000 and 500 steps. The quantization level was the same as the previous simulation whose results are given in Figure 13. The results of adaptive receiving for these fading periods are shown in Figure 15. A study of this figure shows that the increase of the number of samples in steady state interval results in the decrease of BER. The difference of BER caused by the difference of the fading period suggests that the bit error occurs ar early portion of transient.

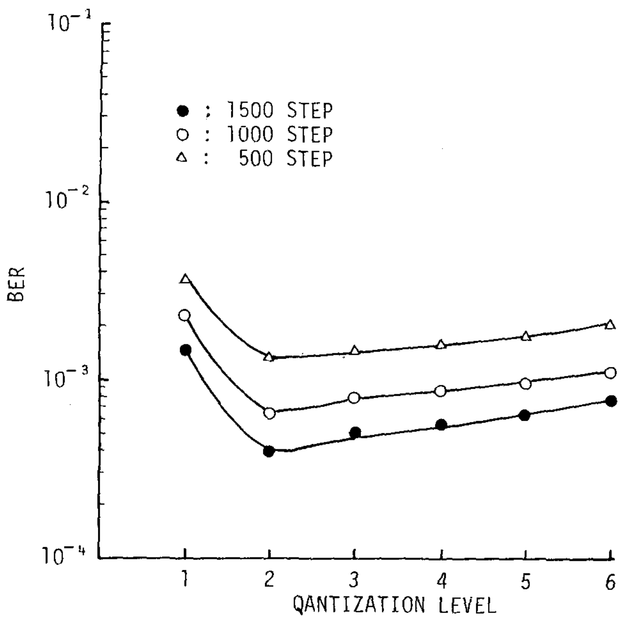

Fig. 15. — BER as a function of quantization level for various periods of fluctuation of the transmission characteristics.

As examplified, the proposed system can be used to equalize the fading characteristics with the variations in space and in time. We must note that the identifier in the system has a mechanism of the adaptive transversal equalizer. The identifer can adjust the impulse response in the Kalman filter bank using the estimated signal $\hat{s}_i$ at every samples, if it is required.

Although these simulations were executed using the fading characteristics in short-wave transmission range, the system is directly applicable to microwave radio system. This expectation is natural from the fact that the system can estimate the transmission characteristics at baseband which does not depend on carrier frequency. Notice that in the identification and quantization algorithm, the delay number σ of the impulse response for microwave radio propagation is smaller than that for short-wave radio propagation since the path legnth for microwave is shorter than that for short-wave in general.

5. Conclusions

We have proposed an adaptive receiving system for a fading channel. The system is composed of an identifier, a quantizer and an estimator. The identifier estimates the transfer function of the fading channel. The quantizer quantizes the estimated parameters of the transfer function and stores the system parameters to use in the estimator. The estimator is composed of parallel Kalman filters and a classification mechanism which recognizes the state of the fading channel, and estimates the transmission signal. To present degradation of reception, the system responds to the changes in the channel characteristics out of the predetermined fluctuation range. We showed that the proposed system surpasses the adaptive transversal equalizer in the ability of fading equalization using M-QAM system.

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