

Performance Evaluation of Adaptive Equalizer in 3-Way Fading Channel considered Impulsive Noise and AWGN

임펄스성 잡음 및 가우시안 잡음이 고려된 3-경로 페이딩 채널에서 적응 등화기의 성능 평가

Hong-Sik Keum*, Yong-Ro Kim*, Dong-Yoo Lee*, Heung-Gyoon Ryu*

김 홍 식*, 김 용 로*, 이 동 유*, 유 흥 균*

ABSTRACT

In this paper, we evaluate and compare the performances of the tapped-delay line equalizer, the decision feedback equalizer, and the lattice-structured equalizer, for the recovery of the digital signal corrupted by the impulsive noise and the white Gaussian noise under the fading channel environment. Adaptive least mean square algorithm and least square algorithm are used in each equalizers.

From the results of error performance analyses, we obtain that in order to produce 10^{-3} BER (bit error rate), lattice-structured equalizer have the SNR margin of 3.0 dB than LMS TDLE and of 3.9 dB than RLS TDLE, and SNR margin of 0.5 dB than LMS DFE and of -0.5 dB than RLS DFE in case that faded, and existed impulsive and Gaussian noise.

요 약

본 논문에서는 페이딩 채널에서 백색 가우시안 잡음과 임펄스성 잡음이 부가된 디지털 신호를 복원하기 위하여 적응 LMS 알고리즘과 RLS 알고리즘을 사용하여 TDL 등화기, 결정 레한 등화기, 그리고 격자 등화기의 성능을 평가하고 비교하였다.

오차 성능 분석 결과, 페이딩이 존재하고 임펄스성과 가우시안 잡음이 존재하는 채널에서 10^{-3} BER을 얻기 위해서, 격자 등화기는 LMS TDL 등화기보다 3.0dB, RLS TDL 등화기보다 3.9dB의 신호대 잡음비(SNR) 여유를, 그리고 LMS DFE 등화기보다 0.5dB, RLS DFE 등화기보다 -0.5dB의 SNR 여유를 갖음을 확인하였다.

1. Introduction

Linear equalizer and non-linear equalizer have been researched to recover the digital signal distorted by intersymbol interference channel noise and multi-path fading [1,2].

Practically, the tapped-delay line(TDL) linear equalizer with the few calculation quantity and

the simplicity, the decision feedback non-linear equalizer(DFE) with the good property for interference, and the lattice-structured equalizer(LE) with low relation for the eigenvalue distribution ratio of the correlation matrix of the channels have been generally used. The used adaptive algorithms are the LMS algorithm with the few calculation quantity and RLS algorithm with the fast convergence speed and good interbit orthogonalizing property [3,4].

In this paper, we have evaluated the perform-

*Dept. of Electronic Eng.
CHUNGBUK NAT'L UNIVERSITY

*충북대학교 전자공학과
접수일자: 1992. 12. 4.

ance of the TDL equalizer, the decision feedback equalizer, and the lattice-structured equalizer using the LMS and RLS algorithm, for the digital signal corrupted by the impulsive noise and the white gaussian noise under the fading channel environment.

II. Adaptive equalizer and channel

1. Adaptive equalizer

1) Tapped-delay line equalizer

TDL(transversal or nonrecursive) equalizer has simple structure among the many equalizer structures.

The present and past value $r(t-nT)$ of the received signal is linearly weighted by the equalizer coefficient(tap weight) and summed up to produce the output.

Equalizer coefficient $C_n, n=0,1,\dots,N-1$, is selected to make all the samples of the impulse response of the equalizer to zero at the T moment.

Tap weight updating equation of the equalizer using the LMS algorithm is represented as eq. (1).

$$C_n(k+1) = C_n(k) - \Delta \cdot e_k \cdot (t_0 + kT - nT) \\ n = 0, 1, \dots, N-1 \quad (1)$$

where, $C_n(k)$: n -th tap coefficient at time k ,

e_k : error signal,

t_0 : sampling time,

Δ : adaptive coefficient, or magnitude of step.

Tap weight updating equation of the equalizer using the RLS algorithm is represented as eq. (2) [1.6].

$$\mathbf{H}(k) = \mathbf{H}(k-1) + \mathbf{R}(k) \cdot e(k) \quad (2)$$

where, $\mathbf{H}(k)$: coefficient vector at time k ,

$\mathbf{R}(k)$: weight vector,

$$= \frac{\beta^{-1} \cdot \mathbf{P}(k-1) \cdot \mathbf{X}(k)}{1 + \beta^{-1} \cdot \mathbf{X}^T(k) \cdot \mathbf{P}(k-1) \cdot \mathbf{X}(k)}$$

β : exponential weighting factor (≈ 1)

$$\mathbf{P}(k) = \beta^{-1} \mathbf{P}(k-1) - \beta^{-1} \mathbf{R}(k) \mathbf{X}^T(k) \\ \mathbf{P}(k-1)$$

: $N \times N$ vector,

$\mathbf{X}(k)$: input vector

$e(k)$: error signal

$$= d(k) - \mathbf{X}^T(k) \mathbf{H}(k-1),$$

$d(k)$: desired signal,

2) Decision feedback equalizer

Non-linear decision feedback equalizer which is useful for channels with severe amplitude distortion, uses decision feedback to cancel the interference from symbols which have already been detected.

The structure of decision feedback equalizer is shown Fig. 1. Equalized signal is the sum of outputs of the forward and feedback parts of the equalizer. The forward filter is like the TDL equalizer. Decisions made on the equalized signal are fed back via a second TDL filter.

The basic idea is that if the already detected value of the symbols are known (past decisions are assumed to be correct), then the ISI contributed by these symbols can be exactly rejected, by subtracting past symbol values with appropriate weighting from the equalizer output. The weights are samples of the tail of the system impulse response including the channel and the forward part of the equalizer.

The forward and feedback coefficients may be adjusted simultaneously to minimize the MSE. The update equation for the forward coefficients is the same as for the linear equalizer.

The feedback coefficients are adjusted according to the eq. (3).

$$b_m(k+1) = b_m(k) + \Delta \cdot e_k \cdot \hat{x}_{k-m}, \quad m=1, \dots, M \quad (3)$$

where, $b_m(k)$: m -th feedback coefficient at time k ,

\hat{x}_k : k -th symbol decision.

The optimum LMS settings of b_m are those that reduce the ISI to zero, within the span of

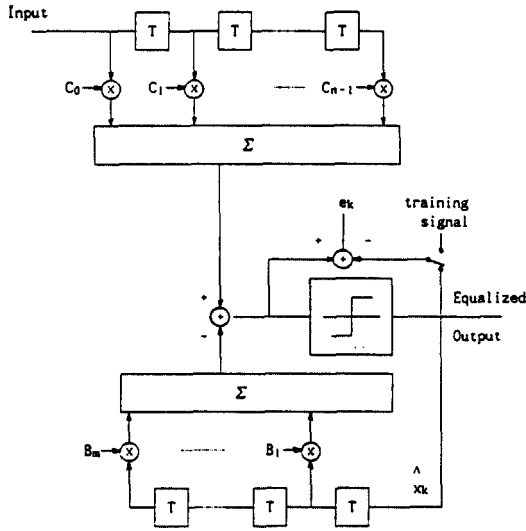
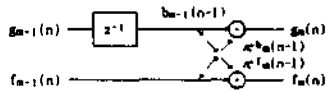


Fig. 1 The block diagram of the decision feedback equalizer (DFE)

the feedback part, in a manner similar to a Zero Forcing equalizer. Since the output of the feedback section of the DFE is a weighted sum of noise-free past decisions, the feedback coefficients play no part in determining the noise power at the equalizer output.

3) Lattice-structured equalizer

It is generally known that the lattice-structur-



< structure of the each stage >

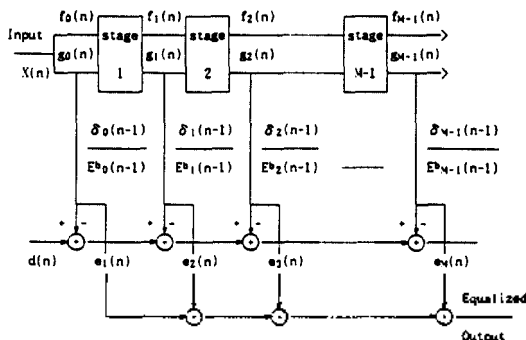


Fig. 2 The block diagram of the lattice-structured equalizer

ed equalizer few relate to the eigenvalue distribution ratio of the correlation matrix of the channels for convergence speed.

The structure of lattice-structured equalizer consists of a set of $n-1$ stages with internal coefficients which are commonly called the reflection or PARCOR coefficients.

The structure of lattice equalizer is shown Fig. 2.

The RLS lattice algorithm is represented as next equations.

$$\begin{aligned} & \text{- Lattice predictor : begin with } n=1 \\ & \quad \& m=0, 1, \dots, M-2 \end{aligned}$$

$$k_{m+1}(n-1) = w_{k_{m+1}}(n-2) + \alpha_m(n-2) f_m(n-1) g_m(n-2)$$

$$\kappa_{m+1}^f(n-1) = -\frac{k_{m+1}(n-1)}{E_{m+1}^b(n-2)}$$

$$\kappa_{m+1}^b(n-1) = -\frac{k_{m+1}(n-1)}{E_m^f(n-1)}$$

$$f_{m+1}(n) = f_m(n) + \kappa_{m+1}^f(n-1) g_m(n-1) \quad (4)$$

$$g_{m+1}(n) = g_m(n-1) + \kappa_{m+1}^b(n-1) f_m(n) \quad (5)$$

$$E_{m+1}^f(n-1) = E_m^f(n-1) - \frac{|k_{m+1}(n-1)|^2}{E_{m+1}^b(n-2)}$$

$$E_{m+1}^b(n-1) = E_{m+1}^b(n-2) - \frac{|k_{m+1}(n-1)|^2}{E_m^f(n-1)}$$

$$\alpha_{m+1}(n-1) = \alpha_m(n-1) - \frac{\alpha_m^2(n-1) |g_m(n-1)|^2}{E_{m+1}^b(n-1)}$$

$$\begin{aligned} & \text{- Lattice filter : begin with } n=1 \\ & \quad \& m=0, 1, \dots, M-1 \end{aligned}$$

$$\delta_m(n-1) = w_{\delta_m}(n-2) + \alpha_m(n-1) g_m(n-1) e_m(n-1)$$

$$\xi_m(n-1) = -\frac{\delta_m(n-1)}{E_{m+1}^b(n-1)} \quad (6)$$

$$e_{m+1}(n) = e_m(n) + \xi_m(n-1) g_m(n)$$

where, κ_m^f, κ_m^b : forward and backward reflection coefficients at stage m ,

E_m^f, E_m^b : forward and backward prediction errors at stage m ,

ξ_m : m -th tap coefficient.

2. Mobile communication fading channel

The mobile communication channel is modelled as the 3-way fading channel in Fig. 3.

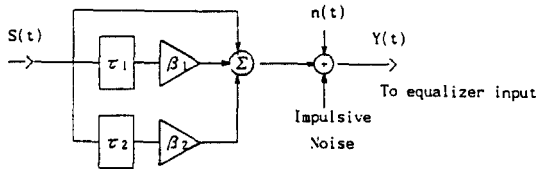


Fig. 3 The model of 3-way fading channels

The input signal $Y(t)$ of the equalizer is represented as eq. (7).

$$Y(t) = S(t)g(t-nT) + \beta_1 S(t)g(t-nT-\tau_1) + \beta_2 S(t)g(t-nT-\tau_2) + n(t) + in(t-nT) \quad (7)$$

where, $S(t)$: source data of '1/0',

$$g(t) = \begin{cases} 1, & |t| \leq T/2, \\ 0, & |t| > T/2 \end{cases}$$

β_1, β_2 : attenuation ratio of the each path,

τ_1, τ_2 : delay of the each path,

$n(t)$: white Gaussian noise,

$in(t)$: impulsive noise.

III. Performance analyses of the equalizer

For the performance analyses, the digital signal which passes through the 3-way fading channel and then corrupted by the Gaussian and impulsive noise is sampled 8 times per symbol period, and used as input of the equalizer.

Table 1 represents the initial values used for performance analysis of Fig. 4.

Table 1. Initial values of LMS equalizer

| | Tap order (forward/backward) | Tap coefficient (forward/backward) | Mag. of step (pried of train./normal) |
|------|------------------------------|------------------------------------|---------------------------------------|
| LE | 4/0 | 0.2/0 | 0.0002/0.0001 |
| DFE | 4/3 | 0.3/0.2 | 0.0002/0.0001 |
| TDLE | 4/0 | 0.15/0 | 0.005/0.001 |

Table 2 represents the initial values used for performance analysis of Fig. 6.

Table 2. Initial values of RLS equalizer

| | Tap order (Forward/Backward) | Tap coefficient (Forward/Backward) | P vector (Forward/Backward) |
|------|------------------------------|------------------------------------|-----------------------------|
| LE | 4/0 | 0.0/0 | 0/0 |
| DFE | 4/3 | 0.3/0.2 | 10/10 |
| TDLE | 4/0 | 0.15/0 | 8/0 |

Fig. 4 shows the performance comparison of tapped-delay line equalizer, decision feedback equalizer, and lattice-structured equalizer using LMS algorithm in case that impulsive noise doesn't exist.

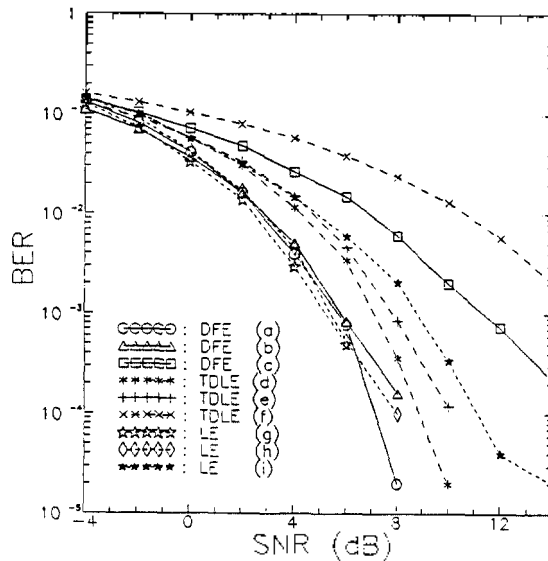


Fig. 4 Performances [1] of the equalizers using LMS algorithm

Fig. 5 is the graph of performance comparison of tapped-delay line equalizer, decision feedback equalizer, lattice-structured equalizer using LMS algorithm in case that impulsive noise does exist.

The used initial values are same as the table 1.

Fig. 6 is the graph of performance comparison of tapped-delay line equalizer, decision feedback

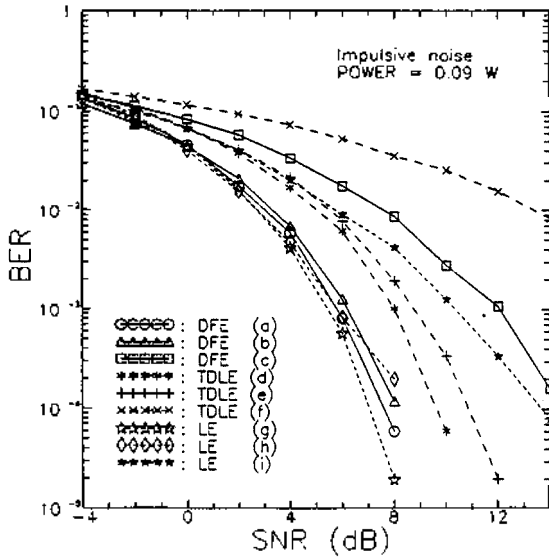


Fig. 5 Performances [2] of the equalizers using LMS algorithm

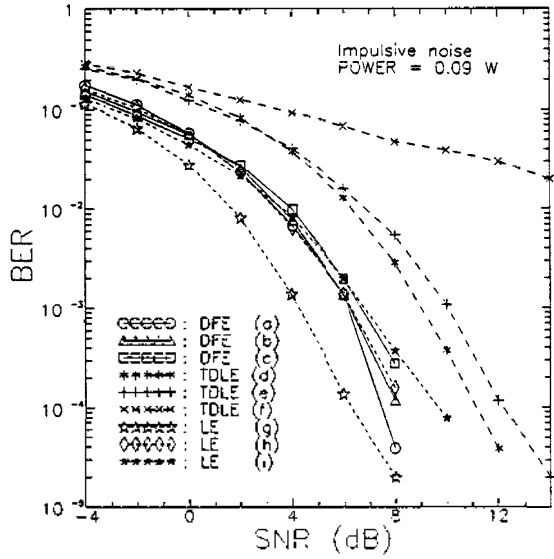


Fig. 7 Performances [2] of the equalizers using RLS algorithm.

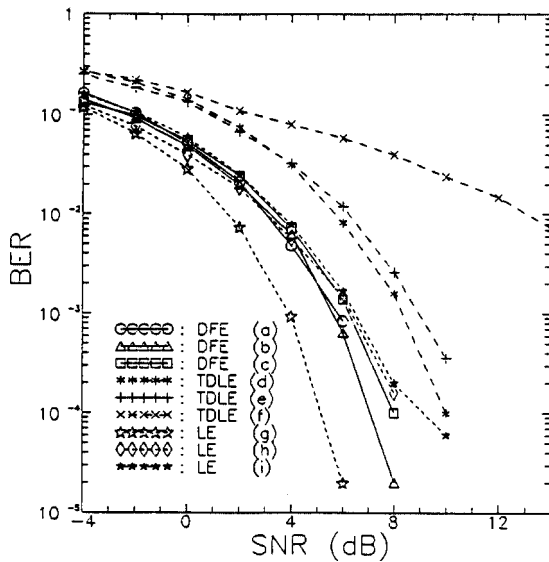


Fig. 6 Performances [1] of the equalizers using RLS algorithm

equalizer, lattice-structured equalizer using RLS algorithm in case that impulsive noise doesn't exist.

Fig. 7 is the graph of performance comparison of tapped-delay line equalizer, decision feedback equalizer, lattice-structured equalizer using RLS

algorithm in case that impulsive noise does exist.

The used initial values are same as the table 2.

In the Fig. 4, 5, 6, 7, the graph (a),(d),(g) represent in case that the fading doesn't exist, the graph (b),(e),(h) represent in the case of $T/4$ delayed at the path 1, the graph (c),(f),(i) represent in the case of $T/4$ delayed at the path 1 and $-T/4$ delayed at the path 2.

It is confirmed that lattice-structured equalizer has better performance than DFE equalizer and TDL equalizer, and that the equalizer perform-

Table 3. SNR comparison between lattice equal. and other equalizers

(In the case of 10^{-3} BER)

| Impulsive Noise | | Exist | | Non exist | |
|-----------------|------------|---------|--------|-----------|--------|
| | | DFE | TDLE | DFE | TDLE |
| L M S | Non fading | 0.45 dB | 2.6 dB | 0.6 dB | 1.9 dB |
| | τ_1 | 0.5 dB | 3.0 dB | 0.4 dB | 2.1 dB |
| R L S | Non fading | 1.9 dB | 4.8 dB | 1.8 dB | 4.4 dB |
| | τ_1 | -0.5 dB | 3.9 dB | -0.8 dB | 2.6 dB |

ance degrades in case that fading and impulsive noise also exist.

In the case of 10^{-3} BER, the comparison of SNR between lattice-structured equalizer and other equalizer is represented as table 3.

where, the used parameters are as follows.

$K=0.08$, (ρ =power ratio of indirect signal to direct signal)

Power of impulsive noise= $0.09[w]$

Power of signal= $1.0[w]$

$\tau_1=T/4$ delay, $\tau_2=-T/4$ delay

IV. Conclusion

In order to recover the distorted digital signal in the mobile communication fading channel, we have analyzed the performance of the tapped-delay line equalizer with the few calculation quantity and the simplicity, the decision feedback equalizer with the good property for interference, and lattice-structured equalizer that is known with low relation for the eigenvalue distribution ratio of the correlation matrix of channels using the LMS algorithm and the RLS algorithm.

We have obtained the analyses results that in order to produce 10^{-3} BER, lattice-structured equalizer have the SNR margin of 3.0 dB than LMS TDLE and of 3.9 dB than RLS TDLE, and the SNR margin of 0.5 dB than LMS DFE and of -0.5 dB than RLS DFE in case that don't faded, and existed impulsive and gaussian noise.

From the results of the performance analysis, we have obtained that lattice-structured equalizer has better performance than DFE equalizer and TDL equalizer, also that the equalizer performance deteriorates in case that fading and impulsive noise concurrently exist.

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▲Hong-Sik Keum



1988. 3~1992. 2 : Dept. of Electronics Eng. CHUNGBUK NAT'L UNIV.
(Bachelor's Degree)

He is in the course of M.S. in the department of electronics engineering from March, 1992.

His interesting area is statistical communication system and adaptive filter applications.

▲Yong-Ro Kim



1986. 3~1991. 2 : Dept. of Electronics Eng. CHUNGBUK NAT'L UNIV.
(Bachelor's Degree)

He is in the course of M.S. in the department of electronics engineering from March, 1992.

His current research interests are in communication theory, digital communication systems, and its applications.

▲Dong-Yoo Lee



1988. 3~1992. 2 : Dept. of Electronics Eng. DAEJEON NAT'L TECHNICAL UNIV.
(Bachelor's Degree)

He is in the course of M.S. in the department of electronics

engineering of chungbuk national university from March, 1992. His current research interests are in communication system and digital signal processing.

▲Heung-Gyoon Ryu : Vol.10, No.4, 1992.