論文 97-22-7-10

The Adaptive Reduced State Sequence Estimation Receiver for Multipath Fading Channels

Young-Jo Lee*, Sung-Lark Kwon**, Tae-Hyun Moon**, Chang-Eon Kang** *Regular Members*

이동통신 환경에서 적응 상태 축약 심볼열 추정 수신기

正會員 이 영 조*, 권 성 락**, 문 태 현**, 강 창 언**

※이 연구는 95년도 한국과학재단 연구비 지원에 의한 결과임(과제번호 : 951-0913-029-1).

ABSTRACT

In mobile communication systems, the Reduced State Sequence Estimation(RSSE) receiver must be able to track changes in the channel. This is carried out by the adaptive channel estimator. However, when the tentative decisions are used in the channel estimator, incorrect decisions can cause error propagation.

This paper presents a new channel estimator using the path history in the Viterbi decoder for preventing error propagation. The selection of the path history for the channel estimator depends on the path metric as in the decoding of the Viterbi decoder in RSSE. And a discussion on the channel estimator with different adaptation algorithms such as Least Mean Square(LMS) algorithm and Recursive Least Square(RLS) algorithm is provided. Results from computer simulations show that the RSSE receivers using the proposed channel estimator have better performance than the other conventional RSSE receiver, and that the channel estimator with RLS algorithm is adequate for multipath fading channel.

요 약

상태축약심볼열추정(RSSE: Reduced State Sequence Estimation) 수신기는 비터비 복호기와 채널 추정기로 구성된다. 이동통신과 같이 채널이 변하는 환경에서는 적응 채널추정기(adaptive channel estimator)로 채널의 변화를 계속적으로 추정해야 한다. 일반적으로 사용되는 채널 추정기는 임시결정된 비터비 복호기의 출력을 사용하여 채널을 추정 하는데, 비터비 복호기에서 잘못된 결정을 내릴 경우 이로 인해 오류전파(error propagation)가 발생할

*LG 정보통신
**연세대학교 전자공학과
論文番號:96167-0605
接受日字:1996年 6月 5日

수 있다.

본 논문에서는 좀더 정확한 채널 추정과 오류전파를 막기 위해 경로 메모리를 사용하는 새로운 채널추정기를 사용한다. 이 채널 추정기는 비터비 복호기의 여러 경로중에서 가장 작은 경로를 선택하여 그 경로상의 신호를 이용하여 채널 추정을 행한다. 그리고 채널 추정기의 적응 알고리듬으로서 LMS(Least Mean Square) 알고리듬과 Recursive Least Square(RLS) 알고리듬을 사용하여 비교한다. 실험 결과를 통해 제안된 채널 추정기를 사용하는 RSSE 수신기가 기존의 채널 추정기를 사용하는 RSSE 수신기에 비해 더 나은 성능을 나타내는 것을 볼 수 있으며, 페이딩이 존재하는 이동통신 환경에서는 LMS 알고리듬이 적합하지 않음을 알 수 있다.

I. Introduction

Communication systems such as mobile and indoor wireless communication typically are modelled as frequency selective fading channels. Frequency selectivity implies that the Intersymbol Interference(ISI) is present. To combat the ISI resulting from a multipath fading channel, the mobile communication systems require adaptive equalization[1]. It is well known that the Maximum Likelihood Sequence Estimation(MLSE) receiver is optimum for signals corrupted by ISI and white Gaussian noise.

Trellis coded modulation(TCM) provides a powerful means of combating the effects of thermal noise at the price of increased receiver complexity. Decoding is performed by utilizing a Viterbi algorithm that searches for the path with maximum likelihood in the trellis diagram of the code.

When TCM decoding and equalization is performed joinlty by the Viterbi decoder, the Viterbi algorithm operates on the trellis diagram of a finite state machine(FSM) that models the cascade of encoder and transmission channel in the decoder[2]. However, because the number of states increases exponentially with the length of the channel impulse response(CIR), high complexity trellis decoders would often be required. Due to the large number of states in MLSE, a Viterbi processor with reduced complexity is required.

In the RSSE, the complexity reduction may be obtained not only through the channel truncation but also by a partial consideration of the ISI in the constructon of the trellis diagram. Each branch metric is calculated on the basis of the particular survivor sequence. This RSSE is the technique that makes possible the compromise between the conflicting requirements of performance and complexity with the interesting result that the optimal performance can be often closely approached with significant complexity saving[3][4][5][6].

The receiver of the RSSE consists of a channel estimator and a Viterbi decoder. The channel estimator adaptively estimates the channel parameters and makes their values available to the Viterbi decoder. In order to obtain a correct model of the FSM at the receiver, instantaneous channel characteristics must be known at all times. But, because these characteristics are unknown and time-varying in the mobile environment, the RSSE must be made to learn and track the channel parameters adaptively. To track the channel parameters, decision-directed channel estimators in conjunction with the RSSE have been proposed[1][7]. A common approach in the decision-directed channel estimator is to obtain the reference sequence from tentative low delay decisions at the Viterbi decoder output in a decision directed mode[9][10][11][12]. But Since the low delay tentative decisions are not reliable[9][10], the incorrect decisions may occur, and incorrectly detected symbol would be fed to the transversal filter of the channel estimator. While incorrectly detected symbol is in the transversal filter, the adaptation of the channel estimator would be incorrect, causing error propagation.

For a more accurate channel estimate without decoding delay, a number of algorithms have been proposed which uses many channel estimators adapted independently on an associated data sequence[7][8]. However, the improvement is obtained at the cost of a substantial increase in complexity. This is because the number of the channel estimates equals the number of states in this approach.

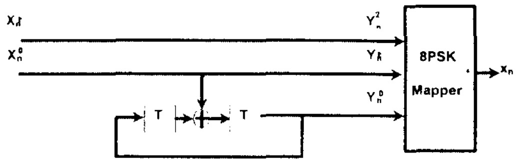
In the decision feedback equalizer, an incorrect data fed into the feedback filter would result in as incorrect adaptation. So for a more accurate input data to the feedback filter, the detected data symbols from the output of the Viterbi decoder may be used. However, in this case, there is a performance degradation because of the large decoding delay that is inherent in the Viterbi decoder. On the contrary when the tentative decisions of the Viterbi decoder are used, the decision signals are not reliable. Steele, et al, proposed the soft decision feedback equalizer, in which the feedback filter takes the symbols from path history of the Viterbi decoder[11].

In this paper, we use the symbols in the path history for channel estimator as in the soft decision feedback equalizer mentioned above. The data in the transversal filter of the channel estimator are not shifted, but copied from the path history in the Viterbi decoder. The selection of the path history in the Viterbi decoder depends on the path metric. In other words, the data sequence is copied from the path history selected for decoding of the Viterbi decoder in the RSSE.

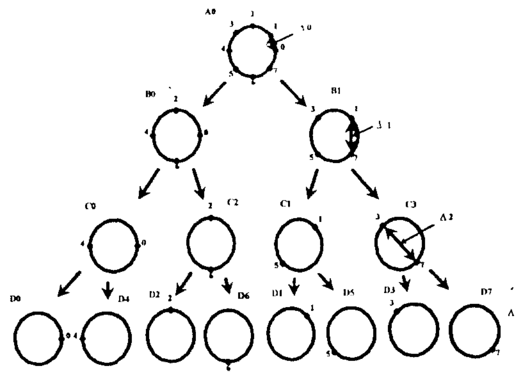
Ⅱ. System Model and Review of RSSE

In this section, we briefly review the RSSE for coded signal tranmission over ISI channel. In the Viterbi decoder, decoding and equalization are performed jointly. A sequence of indepentent identically distributed information symbol $\{u_k\}$ is mapped by a TCM encoder into a code sequence $\{x_k\}$. Here we consider trellis-coded 8PSK constellation signals. The trellis encoder, and set partitioning of the considered 4-state TCM are shown in figure 1(a) and figure 1(b).

We consider an equivalent discrete time additive



(a)



(b)

Figure 1. (a) 8-PSK 4-state TCM encoder
(b) Set partitioning for 8-PSK signal set

noise linear channel model[1] which consists of a cascade of the transmit filter, the modulator, the channel, and the demodulator. In the following, we shall refer to the equivalent channel simply as the channel. We take one sample per signaling interval T. The channel may be modelled as a finite impulse response (FIR) filter(also called a transversal filter). The coded signal is transmitted over a complex linear channel. Then the received signal y_n, y(t=nT), can be written as

$$y_n = \sum_{k=0}^{L} x_k h_{n-k} + n_n \qquad (1)$$

$$h_n = h_{In} + jh_{Qn}$$

$$n_n = n_{In} + jn_{Qn}$$

$$x_k = x_{Ik} + jx_{Qk}$$

where $\{h_n\}$ is the complex channel impulse response (CIR) of the overall channel, and L is the length of the CIR. The noise terms n_{In} and n_{Qn} are independent and Gaussian with zero mean.

In order to review the RSSE receivers, we introduce a viterbi decoder operating on a combined code and an ISI trellis with reduced states[3][4][5][6][13]. The state $S_n^{L'}$ is defined as

$$S_n^{L'} = (\sigma_n; u_{n-L'}(m_{L'}), u_{n-L'+1}(m_{L'-1}), \cdots, u_{n-1}(m_1)) \quad (2)$$

in which σ_n is the encoder state at epoch n, L' is the reduced channel memory length($L' \le L$, with L being the true channel memory length). And the L'-tuple $(u_{n-L'}(m_{L'}), u_{n-L'+1}(m_{L'-1}), \cdots, u_{n-1}(m_1))$ represents the reduced channel state. For each $u_{n-i}(m_i)$, m_i characterizes the depth of set partitioning. In the case of S state TCM encoder and a signal contellation W points, the combined trellis has $S2^{\sum_i^{L} m_i}$ states.

The branch metrics $M_n[S_n^{L'}]$ at epoch n is defined as

$$M_n[S_n^{L'}] = |r_n - \sum_{j=L'+1}^{L} f_j \hat{x}_{n-j}(S_{n-1}^{L'}) - \sum_{j=0}^{L'} f_j x_{n-j}|^2 \quad (3)$$

where $\hat{x}_{n-j}(S_{n-1}^{L'})$ denotes the ISI terms due to symbols $\{x_{n-j}\}$, $L'+1 \le i \le L$. Symbols which are not represented by the truncated state $S_{n-1}^{L'}$ are estimated using decisions taken from the path history associated with predecessor state $S_{n-1}^{L'}$. And f_j is the estimated CIR. $M_n[S_n^{L'}]$ functionally depends on the received signal r_n and the estimated channel impulse reponse f_j.

Under the assumption that $\{h_n\}$ is accurately known, the optimum receiver is composed of a filter matched to the pulse $\{h_n\}$ followed by a symbol rate sampler and a Viterbi decoder. This Viterbi decoder searches for the path with minimum metric in the trellis diagram of a FSM that models the cascade of encoder and transmission channel. But the CIR is not known in the receiver, the CIR f_j should be estimated. The channel estimation is accomplished using the channel estimator.

Ⅲ. The Channel Estimator using Path History

3.1 Channel Estimation

In TDMA mobile communications, every subframe contains a training sequence[14][15]. When data are transmmited over a distorting channel, whose channel impulse reponse is not known, the CIR must be esimated using the channel estimator. This is normally achieved by the means of a known sequence of data symbols. This training sequence must be transmitted in every subframe for each user. So the transmitted signals is often split into blocks, each containing a training sequence for a channel estimator and a data burst. And the channel estimator uses the received signals and the data sequence used as a reference to adjust the channel estimator.

Fig 2. shows a channel estimator and RSSE based equalizer which may be used to mitigate the ISI and improve the detection of the transmitted symbols. For channels which fade rapidly compared to the transmitted symbol rate or to the length of the data burst, in order to improve the symbol error rate, variations in the channel may be tracked during data transmission using the symbol estimates from the output of the RSSE. The channel estimator is updated once per symbol period, on reception of the transmitted symbol estimate, and the new channel estimate is fed into the RSSE where it is used together with the received signal to calculate the path metrics in the following

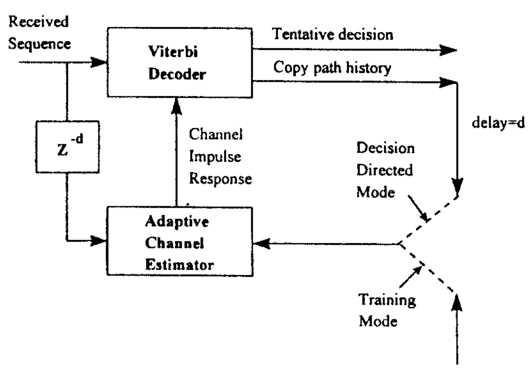


Figure 2. The structure MLSE receiver
(a) n = k (b) n = (k + 1)

iteration of the Viterbi algorithm.

Decision-directed channel estimators in conjunction with MLSE have been proposed. And in order to shorten the symbol estimation delay while maintaining the reliability of the estimate, a number of algorithms have been proposed which use a number of channel estimators adapted independently on an associated data sequence[7][9][10]. However, the improvement is obtained at the cost of a substantial increase in complexity; because the number of the channel estimates equals the number of states in this approach.

In decision directed mode, the channel estimator employs tentative low delay decisions at the Viterbi decoder output. Tentative decisions of the Viterbi decoder are used to form an error signal which is then employed to make appropriate adjustments of the current channel estimate[11][12]. Therfore channel estimator uses the delayed symbol estimates to approximate an old CIR $f_j(k-d)$ which is fed back to the RSSE for the purpose of path metric calculation.

The efficiency of the procedure is strongly related not only to the reliability of the tentative decisions but also to the decision delay. There is a compromise in the choice of the symbol estimation delay d. A short delay permits faster channel tracking while a long delay insures a better reliability of the symbol estimates resulting in a better long term convergence. In a fast fading channel, one has no choice but to use a short delay to keep up with channel variations. A poor reliability of symbol estimates and a substantial decrease in the overall performance of the procedure are thus unavoidable.

The structure of a transversal filter is used in the channel estimator. Adaptive algorithms to make the channel estimator adaptive are typically used to update the taps of a transversal filter. We will consider the two most widely used adaptive algorithms, namely LMS algorithm and RLS algorithm[16].

The LMS algorithm is the simplest and the most widely used adaptation algorithm. The coefficients of channel estimator that employs LMS algorithm are updated according to the following adaptation equation[17].

$$F(n+1) = F(n) + \mu e(n)\tilde{X}^*(n) \tag{4}$$

where $F(n) = [f_0(n), \ldots, f_L(n)]'$ is the tap weight vector ($'$ denotes transpose). The vector $\tilde{X}(n)$ has the components $(\tilde{x}_{n-d-L}, \ldots, \tilde{x}_{n-d-1}, \tilde{x}_{n-d})$ that are tentative decisions in the Viterbi decoder. ($*$ denotes complex conjugate). $\tilde{x}_{n-d}$ denotes tentative decision with decoding delay d. And μ is the step size.

The RLS algorithm is a deterministic version of the classical Kalman Filter algorithm. The RLS algorithm using tentative decisions is as follows[17]

$$K(n+1) = \frac{P(n)\tilde{X}^*(n)}{w + \tilde{X}'(n)P(n)\tilde{X}^*(n)}$$

$$P(n+1) = \frac{1}{w}\,[P(n) - K(n+1)\tilde{X}'(n)P(n)]$$

$$F(n+1) = F(n) + K(n+1)e(n) \tag{5}$$

Where K(n) is the Kalman gain Vector.

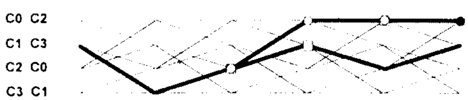
3.2 The channel estimator using the path history

The proposed channel estimator employs the sequence in the path history of the Viterbi decoder in contrary to the symbol by symbol decision of the Viterbi decoder used in the conventional channel estimator.

Because the tentative decisions are not always correct, error propagation can result. This is because such an incorrect tentative decision is shifted in the transversal filter of the channel estimator. Therefore if the tentative decision is incorrect, the adaptation of the channel estimator would be incorrect. In shorts, while incorrectly detected symbol is in the transversal filter, error propagation can occur, resulting in an incorrectly processed adaptation. As a result, some other candidate may be selected instead of the maximum likelihood candidate, becuase they have higher metrics than the maximum likelihood path.

In this paper, the tentative decisions are used for the proposed channel esimator. But to prevent the error propagation, the data in the transversal filter are not shifted as in the conventional estimator, but copied from the path history in the Viterbi decoder. The selection of the path history in the Viterbi decoder depends on the path metric. In order words, the data are copied from the path history selected for decoding of the Viterbi decoder in the RSSE. So the channel estimator is adapted using $\hat{x}_{n-j}(S^{L'}_{n-1})$, $L'+1 \le j \le L$, in path history, and the data the data $\{x_{n-i}\}$, $0 \le i \le L'$, corresponding to the state with the minimum path metric.

(a) n=k



(b) n=(k+1)

	t=nT	t=(n+1)T
tentative decision	(C3 ,C3, C0)	(C3, C0, C0)
using path history	(C3 ,C3, C0)	(C3, C2, C0)

Figure 3. The comparison of the content of the transversal filter for each channel estimator

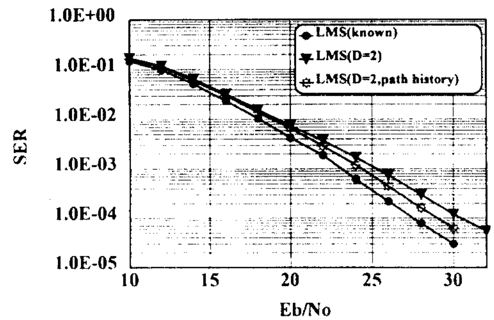


Figure 4. Performance of 16-state RSSE receivers with LMS algorithmfor $f_d = 30$Hz

In the conventional channel estimator, the channel estimator is updated using the symbol by symbol decisions of the viterbi decoder. The signals in the transversal filter are $(\tilde{x}_{n-d-L}, \ldots, \tilde{x}_{n-d-1}, \tilde{x}_{n-d})$. And the signals are shitfted on every iteration. On the contrary, in the proposed channel estimator, the signals in the transversal filter, $(\hat{x}_{n-d-L}, \ldots, \hat{x}_{n-d-1}, \hat{x}_{n-d})$, are copied from the path history with the minimum path metric. $\hat{x}_{n-d}$ is the symbol with delay d in the path history. A pictorial comparison is examplified in the fig. 3. In this case, L = 3, $d = 1$, and $L' = 0$. Fig. 3 shows the trellis and the subsets of the signals which are the contents of the transversal filter for each channel estimator. The symbols, which are used to update the channel weight, belongs to the subset. And the subsets are assigned on the branch as fig. 3. Fig. 3(a) shows trellis extention at n = k, and fig. 3(b) shows trellis extention at n = (k + 1). The bold lines are the paths with the smallest path metric, and the second smallest path metric. And rectangles denote the signals in transversal filter for conventional channel estimator, and circles denote the signals for the proposed channel estimator. As shown in fig. 3, signals in conventional channel estimator is shifted, contrary to the signals in the proposed channel estimator which are copied from the path history.

In the conventional channel estimator, incorrectly detected symbols of the tentative decision remain in existence while being shifted in the transversal filter. So one error is propagated the same number of times as the number of transversal filter taps. But when the data are obtained each time from the path history in the Viterbi decoder, the data in the transversal filter is not shifted. Therefore no propagation of error occurs. Therefore the data in the transversal filter is more accurate than the data detected by the tentative decision. In this case error propagation does not occur and adaptation becomes more accurate, resulting an improvement in performance.

Ⅳ. Simulation Results

Several computer simulations were performed to compare the performance of the RSSE using the proposed channel estimator to that of the conventional RSSE. Considered signal constellation is the trellis

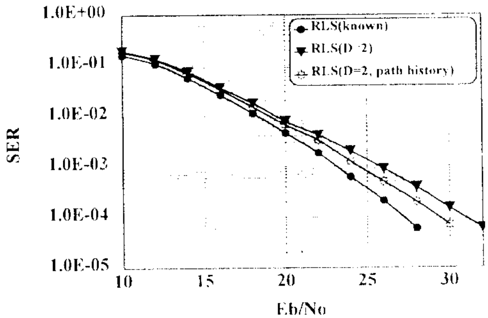


Figure 5. Performance of 16-state RSSE receivers with RLS algorithmfor $f_d = 30$Hz

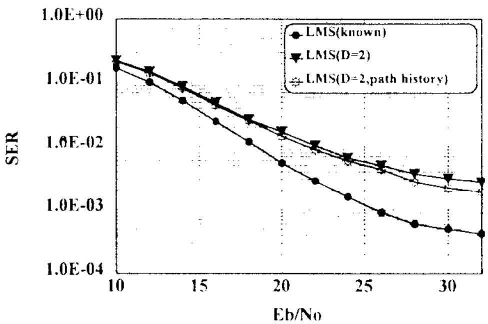


Figure 6. Performance of 16-state RSSE receivers with LMS algorithmfor $f_d = 80$Hz

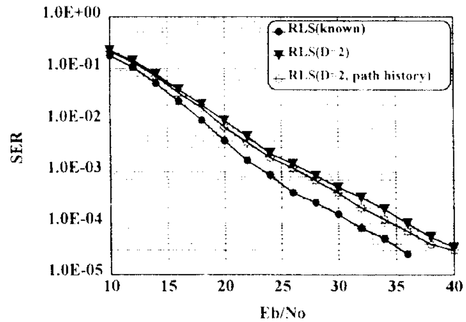


Figure 7. Performance of 16-state RSSE receivers with RLS algorithmfor $f_d = 80$Hz

coded 8-PSK, in which the trellis coding has 4 states. The state trellis of the Viterbi decoder is 16 state trellis, and this is obtained by truncating the channel memory, $L' = 1$, $m_1 = 2(4\times4)$. No synchronization error is assumed.

To reproduce a typical mobile communication environment, a TDMA data frame is assumed, where each user transmits a block of data symbols proceeded by a known preamble and followed by a known tail. For computer simulations, each data block is formed by 60 information symbols and 16 known symbols. The symbol rate is set to 200 Ksymbol/sec.

We assumed a 3-ray channel model with a channel memory L = 2, and that the random process modeling the elements of the discrete impulse response have the same standard deviation. Several values of doppler shift have been considered, in the range of 30~80Hz. All simulations are proceeded with $\mu = 0.02$ for LMS and w = 0.96 for RLS.

The performance has been optimized with respect to the parameter d, and the optimal decision delay is d=2.

Figure 4 and 5 show the simulated performance (symbol-error rates(SER) as a function of E_b/N_0) for fading rate $f_d = 30Hz$. The RSSEs of the channel estimator using the path history is found to perform about 1.0~2.0dB better than other conventional RSSEs. The error propagation from incorrect tentative decisions creats a loss of 1.0~2.0dB.

Figure 6 and 7 show the performance for $f_d = 80Hz$. The difference of the performance is approximately 1dB. The RSSE receiver using LMS algorithm shows a significant degradation and presents floor about 2×10^{-3} for high doppler frequency. This shows that the channel estimator using LMS algorithm is not suitable for the mobile communication.

V. Conclusion

RSSE should track the changes in the channel, and this is performed by the channel estimator. We have presented a new channel estimator structure using the path history in the Viterbi decoder, which may be used to improve the performance of the RSSE. To prevent the error propagation in the conventional channel estimator, the data in the transversal filter are not shifted, but copied from the path history in the Viterbi decoder. The selection of the path history depends on the path metric as in the decoding of the Viterbi decoder in RSSE. Since incorrect tentative decisions don't remain in the transversal filter, no propagation of error occurs. Therefore, a more accurate adaptation is provided.

Results from computer simulations show that the RSSE receivers using the proposed channel estimator have better performance than the other conventional RSSE receiver. The simulation results also show that a simple adaptive algorithm such as LMS algorithm is not suitable for mobile communication systems. One can obtain reasonable performance with a robust algorithm such as RLS algorithm.

References

1. J. G. Proakis, Digital communications, McGraw-Hill, 1989.
2. G. D. Forney, "Maximum likelihood sequence estimation of dfigital sequence in the presence of intersymbol interference," IEEE Trans. Inform. Theory, vol. IT-18, pp. 363-378, May. 1972.
3. V. M. Eyubo lu and S. U. Qureshi, "Reduced-state sequence estimation with set partitioning and decision feedback," IEEE Trans. Commun., vol. 36, pp. 13-20, Jan. 1988.
4. V. M. Eyubo lu and S. U. Qureshi, "Reduced-state sequence estimation for coded modulation on intersymbol interference channels," IEEE J. Select. Areas Commun., vol. SAC-7, pp. 989-995, Aug. 1989.
5. P. R. Chevillat and E. Eleftheriou, "Decoding of trellis encoding signals in the presence of intersymbol interference and noise," IEEE Trans. Commun., vol. 37, pp. 669-676, Jul. 1989.
6. W. H. Sheen and G. L. st ber, "MLSE equalization and decoding for multipath-fading channels," IEEE Trans. Commun., vol. 39, pp. 1455-1464, 1991.
7. N. Seshadri, "Joint Data and Channel Estimation using Blind Trellis Search Techniques," IEEE Trans. Commun., vol. 42, no. 2/3/4, pp. 1000-1011, 1994.
8. Jianjun Wu, A. H. Aghvami and J. E. Pearson, "A Reduced State Soft Decision Feedback Viterbi Equaliser For Mobile Radio Communications," VTC 94, Stockholm, Sweden. pp. 234-238, 1994.
9. Jianjun Wu, A. H. Aghvami, "A New Adaptive Equalizer with A Channel Estimator For Mobile Radio Communications," ICC 95, Seattle, USA, pp. 998-1002, 1995.
10. Riccardo Raheli, Andreas Polydoros, and Ching-Kae Tzou, "Per-Survivor Processing: A General Approach to MLSE in Uncertain Envionments," IEEE Trans. Commun., vol. 43, no. 2/3/4, pp. 354-364, 1995.
11. Joseph C. S. Cheung, Raymond Steele, "Soft-Decision Feedback Equalizer for Continuous Phase Modulated Signals in Wideband Mobile Radio Channels," IEEE Trans. Comm., vol. 42, no. 2/3/4, 1994.
12. W. U. Lee and F. S. Hill, "Maximum-Likelihood Sequence Estimator with Decision Feedback Equalization," IEEE Trans. Comm., vol. COM-25, pp. 971-979, Sept. 1977.
13. G. Marino, "Applications of Reduced State Sequence Estimation to Terrestrial Digitla Radio Links," VTC 94', pp. 347-352, Stockholm, Sweden.
14. J. G. Proakis, "Adaptive equalization for TDMA digital mobile radio," IEEE Trans. Veh. Technol., vol. 40, no. 2, pp. 333-341, May 1991.
15. R. D'avella, L. Moreno, and M. Sant'agostino, "Adaptive MLSE receiver for TDMA digital mobile radio," IEEE J. Select. Areas Comm., vol. 7, no. 1, pp. 122-129, Jan. 1989.
16. Weilin Liu, Herve Romanens, "Adaptive Channel Equalization for High-Speed Train," VTC 94', pp. 225-229, 1994.

17. S. Haykin, Adaptive Filter Theory, Prentice-Hall Englewood, N. Y., 1991.

李 永 朝(Young Jo Lee)정회원

1987년 3월 : 연세대학교 전자공학과 입학

1991년 2월 : 연세대학교 전자공학과 졸업(공학사)

1993년 2월 : 연세대학교 본대학원 전자공학과 졸업(공학석사)

1997년 2월 : 연세대학교 본대학원 전자공학과 졸업(공학박사)

1994년 3월~현재 : LG 정보통신 선임연구원

權 聖 樂(Sung Lark Kwon)

1988년 3월 : 연세대학교 전자공학과 입학

1992년 2월 : 연세대학교 전자공학과 졸업(공학사)

1994년 2월 : 연세대학교 본대학원 전자공학과 졸업(공학석사)

1994년 3월~현재 : 연세대학교 본대학원 전자공학과 박사과정

文 部 鉉(Tae Hyun Moon)정회원

1986년 2월 : 연세대학교 전자공학과 졸업(공학사)

1986년~1991년 : 대우전자 중앙연구소 연구원

1993년 8월 : 연세대학교 본대학원 전자공학과 졸업(공학석사)

1997년 8월 : 연세대학교 본대학원 전자공학과 졸업(공학박사)

康 昌 彦(Chang Eon Kang) 정회원

통신학회 논문지 제 20권 제 1호 참조