DBpia

이동 위성 채널에서 적응 RSSD 수신기에 관한 연구

정회원 송 형 규*

An Adaptive Reduced-State Sequence Detection Receiver in the Mobile Satellite Channel

Hyoung-Kyu Song* *Regular Member*

요 약

본 논문에서는 이동 위성 채널에서 신호를 검파하는데 있어서 최적의 수신 방식인 MLSD 대신에 복잡도를 줄이면서 거의 같은 성능을 얻는 적응 RSSD 수신 방식을 제안한다. 제안된 RSSD를 시변 채널인 이동 위성 채널에 적용하기 위해서는 채널의 상태를 추정해야 하는 문제가 발생하며, 본 논문에서는 채널을 특성을 추정하는 방법으로 steepest descent 알고리듬을 이용하며, 보다 효율적인 채널 추정을 위해 symbol-aided 방법이 사용된다. symbol-aided 방법은 신뢰성있는 채널 추정을 위하여 송신단에서는 주기적으로 알고 있는 심볼을 삽입한 후에, 수신단에서는 알고 있는 심볼이 수신되면 비터비 프로세서는 임시 데이터 결정을 하여 채널 추정기로 보내는 방법이다. 이러한 임시 데이터 결정 방법을 사용한 채널 추정기는 긴 결정 지연 시간을 이용한 방법보다도 이동 위성 채널에서 빠르고 신뢰성 있게 채널을 추정한다. 본 논문에서 제안한 적응 RSSD 수신 방식은 적응 MLSD에 대한 준최적 수신 방식임에도 불구하고 복잡도를 크게 줄이면서 이동 위성 채널을 빠르고 신뢰성 있게 추적한다.

ABSTRACT

This paper proposes an adaptive RSSD receiver, which is a suboptimal alternative to the adaptive maximum likelihood detector and is able to track the time-varying ISI channels in the mobile satellite channel. The structure of the proposed adaptive RSSD is a modified RSSD utilizing a per-survivor processing as well as the symbol-aided method and a channel estimation using the tentative data sequences with a small decision delay. The complexity and performance of the proposed adaptive RSSD are controlled by the number of system states and ISI cancelers and the inserting period of the known symbols. The simulation results show the fact that the proposed adaptive RSSD obtains excellent tracking performance over time-varying ISI channels.

I. Introduction

It is widely known that maximum-likelihood sequence detection(MLSD) is the optimum detection technique for a digital signal corrupted by intersymbol interference(ISI) and additive white Gaussian noise(AWGN). The time variant characteristics of the channels require adaptive operation of an MLSD receiver. Various kinds of adaptive MLSDs have been developed. The adaptive MLSD generally consists of a channel estimator and an MLSD implemented by the Viterbi algorithm(VA). In the adaptive MLSD, a transmitted information sequence is estimated by the MLSD based on a channel impulse response (CIR), and the CIR is estimated by the channel estimator using an estimate of the transmitted sequence derived in the MLSD[3, 5, 8].

The use of the conventional MLSD receiver can bring problems. As the complexity of the VA

* 전자부품연구원 (songhk@nuri.keti.re.kr)
논문번호 : 98249-0618, 접수일자 : 1998년 6월 18일

www.dbpia.co.kr

grows exponentially with the number of states, it becomes difficult to implement the case of a large number of states. And as the decision delay is inherent in the VA, the adaptive MLSD cannot avoid a channel estimation delay on time-varying channels. The decision delay is a variable and its amount is unbounded. However, it is known that the decision delay is less than 5L, where L is the channel memory length. The decision delay causes a channel estimation delay, eventually resulting in a poorer tracking performance on time-varying channels[2, 4].

A great deal of research has been undertaken to reduce the complexity of MLSD while retaining most of its performance. One approach is to use a prefilter to shape the CIR into one having a shorter length and then apply MLSD with a smaller number of states. Another method is to simplify the VA by retaining a small number of likely path.

Recently two sequence estimators which provide a good performance/complexity trade-off have been proposed. Reduced-state sequence detection (RSSD) is useful for a system with a large signal constellation, and delayed decision-feedback sequence detection (DDFSD) is useful for a system with a long CIR. Both sequence detection techniques use the VA to search for the more likely path. In these schemes, a feedback mechanism must be introduced in the calculation of the branch metrics because of the reduction in the number of system states. The feedback introduces error propagation. However, the effect of the error propagation was shown to be much smaller than that with a decision feedback equalization(DFE). For channels with a finite CIR, DDFSD can be conveniently modeled as a special case of RSSD. Symbol-aided method has been studied in reference[10][11]. The common idea in these papers is that one out of every group of transmitted symbols is known at the receiver and can be used to wipe out the modulation from the corresponding signal sample. This provides periodic measurements of the fading process, from which channel estimates at symbol-rate are derived through filtering and interpolation.

This paper proposes an adaptive RSSD utilizing the symbol-aided technique for the purpose of reducing the complexity and tentative decision delay of the MLSD. Section Ⅱ describes the system and channel model. The conventional RSSD structure is described in section Ⅲ. In section Ⅳ, an adaptive RSSD algorithm is presented in the mobile satellite channels. Finally, in section Ⅴ, we offer some conclusions.

Ⅱ. Transmitter and channel

Since the pioneering work of Ungerboeck in 1982, trellis-coded modulation(TCM) has become an effective coding technique for bandlimited channels. Using the TCM approach with memoryless modulations, such as M-ary phase shift keying (MPSK) or quadrature amplitude modulation(QAM), significant coding gain can be achieved without bandwidth expansion. Approaches which combine encoders with modulators that have memory have also been studied. One of the modulations, continuous-phase frequency shift keying(CPFSK), is very useful for digital transmission systems due to its constant envelope and small RF bandwidth requirement. Combining coding with CPFSK to achieve coding gain is, therefore, of practical interest[1].

The signal produced at the output of an M-ary CPFSK modulator can be described by

$$s(t, \boldsymbol{a}) = \sqrt{\frac{2E}{T}} \cos(2\pi f_0 t + \theta(t, \boldsymbol{a}) + \theta_0), \quad t \geq 0 \qquad (1)$$

where E is the symbol energy, T is the symbol interval, f_0 the carrier frequency, and θ_0 the initial carrier phase. The transmitted information is contained in the phase

$$\theta(t, \boldsymbol{a}) = 2\pi h \sum_{i=0}^{\infty} a_i \, f(t - iT), \quad t \geq 0 \qquad (2)$$

The parameter h is called the modulation index. When this parameter is a rational number, the number of states in CPFSK is finite. Thus it is

www.dbpia.co.kr

customary to assume that h is the ratio of two integer numbers, and, in particular, that it has the form

$$h=\frac{K}{P} \tag{3}$$

with P, K relatively prime integers. Otherwise, the CPFSK system has an infinite complexity. The information sequence

$$a=(a_0, a_1, \ldots,),$$

$$\begin{cases} a_i \in \{\pm1, \pm3, \ldots, \pm M-1\}, & M \text{ even}, \ i \geq 0 \\ a_i \in \{0, \pm2, \ldots, \pm M-1\}, & M \text{ odd}, \ i \geq 0 \end{cases} \tag{4}$$

is an M-ary sequence.

The function $f(t)$ is the phase response of CPFSK and is given by

$$f(t)=\begin{cases} 0, & t \leq 0 \\ t/2T, & 0 < t \leq T \\ 1/2, & t > T \end{cases} \tag{5}$$

The minimal implementation of the modulator given the minimum number of states in the corresponding trellis diagram is obtained by defining the frequency

$$f_1 = f_0 - \frac{(M-1)h}{2T} \tag{6}$$

and rewriting the signal $s(t)$ as

$$s(t, U)=\sqrt{\frac{2E}{T}}\cos(2\pi f_1 t+\Psi(t, U)+\theta_0) \tag{7}$$

where

$$\Psi(t, U)=4\pi h\sum_{i=0}^{\infty} U_i f(t-iT), \ t \geq 0$$

and $U_i=\frac{(a_i+(M-1))}{2}$

The information sequence $U=(U_0, U_1, \ldots)$, $U_i \in \{0,1,\ldots,(M-1)\}$ is an M-ary sequence. The transmitted signal (7) of CPFSK can be generated from a system composed of a continuous phase encoder(CPE) and an memoryless modulator(MM) as shown in Fig.1. The physical tilted-phase of CPFSK over the nth signaling interval as a function of the MM input is

$$\begin{aligned}\overline{\Psi}(t, U) &= R_{2\pi}[\Psi(t, U)] \\ &= R_{2\pi}[\frac{2\pi K}{P}(X_{2,n}+X_{1,n}\frac{(t-nT)}{T})] \\ &, nT \leq t \leq (n+1)T\end{aligned} \tag{8}$$

$$\text{with} \quad X_{1,n}=U_n \tag{9}$$

$$\text{and} \quad X_{2,n}=V_n \tag{10}$$

$$\text{where} \quad V_n=R_P\left[\sum_{i=0}^{n-1} U_i\right] \tag{11}$$

is the state of the encoder.

From (8)~(11), it is easy to see that over the nth symbol interval the initial value of the physical tilted-phase is proportional to the state V_n of the CPE whereas the slope of the physical tilted-phase is proportional to the current CPE input digit U_n[6,7].

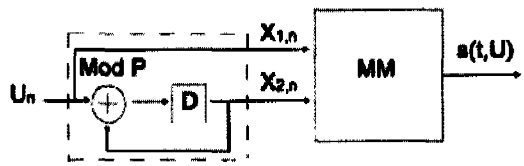

Fig. 1 CPFSK decomposed into CPE & MM

The CPE of Fig. 1 cannot be combined in a natural way with an external convolutional encoder(CE). The combination of a CE with rate $r=(\log_2 M - 1) / \log_2 M$ and the CPFSK system described in Fig. 1 requires a mapper between the CE and the CPE to transform the CE outputs into an M-ary alphabet. Because of this mapper and of the fact that the CE has modulo-2 adders whereas the CPE has a modulo-P adder, the combination of the CE and the CPE is no longer a combinational encoder. An equivalent CPFSK model can be obtained by placing a scrambler at the CPE input. Consider the linear device with M-ary input sequence $\{U_n\}$ and M-ary output sequence $\{U_n'\}$ where

$$U_n'=R_M[U_n-R_P[U_{n-1}]] \tag{12}$$

This inverse operation is given by

www.dbpia.co.kr

$$U_n = R_M[U_n' - R_P\ [U_{n-1}\ ]] \tag{13}$$

If we feed the scramble output U_n' into the CPE with transfer function (9) and (10) we obtain the new input/output relations

$$X_{1,n} = U_n' = R_M\ [U_n - R_p\ [U_{n-1}\ ]] \tag{14}$$

and

$$X_{2,n} = R_P\ [\sum_{i=0}^{n-1} U_i'\ ] = R_P\ [U_{n-1}\ ] \tag{15}$$

The equivalent encoder with (14) and (15) as transfer function is shown in Fig. 2.

In this figure the memoryless mapping represented by the dashed box can be introduced into MM to form an MMM(mapped memoryless modulator).

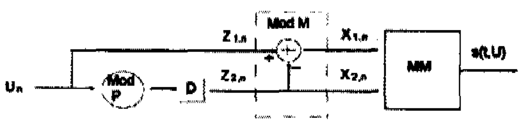

Fig. 2 Equivalent CPE & MM

The input to the MMM is

$$\boldsymbol{Z}_n = (Z_{1,n}\ ,\ Z_{2,n}) \tag{16}$$

From Fig.2, we have

$$X_{1,n} = R_M\ [Z_{1,n} - Z_{2,n}\ ] \tag{17}$$

$$\text{and} \quad X_{2,n} = Z_{2,n} \tag{18}$$

We obtain the physical tilted-phase in the nth interval as a function of $\boldsymbol{Z}_n$,

$$\overline{\Psi}\ (t,\ U) = R_{2\pi}[\frac{2\pi K}{P}(Z_{2,n} + R_M[Z_{1,n} - Z_{2,n}]\frac{(t-nT)}{T})]$$
$$,\ nT \le t < (n+1)T \tag{19}$$

In this paper, the 2/3-rate convolutional code for an 8-ary CPFSK with modulation index h=1/4 is used. The design of this trellis coded 8-ary CPFSK scheme is based on the CPM decomposition technique. Fig.3 shows the block diagram of a trellis-coded 8-ary CPFSK system.

The transmitted signal passes through a slow fading Rician channel with AWGN. Here, w(t) is the zero-mean white Gaussian noise and c(t) is a complex-valued process that describes the multiplicative distortion of the mobile satellite channel. The slow fading Rician channel is discussed in detail Appendix A of reference[9]. For our purpose here it is sufficient to assume that the effect of the fading on the phase of the received signal is fully compensated by tracking it with some form of phase-locked loop. Thus our results will reflect only the degradation due to the effect of the fading on the amplitude of the received signal. This amplitude is modeled by Rician statistics with parameter K representing the ratio of the power in the direct or line-of-sight plus specular components to that in the diffuse component. If shadowing is severe, or if we are dealing with a terrestrial channel, a Rayleigh statistical model becomes appropriate, which can be locked upon as the limiting case of a Rician channel when K approaches zero. Of course, the case of no fading corresponds to a Rician channel with K approaching infinity. Mathematically speaking, the statements correspond to a pdf for the normalized amplitude fading random variable, ρ, given by $p(\rho) = 2\rho(1+K)\exp[-(K+1)\rho^2 - K\ ]I_0(2\rho\sqrt{K(1+K)})$; $\rho \ge 0$, where $I_0(x)$ is the zero-order modified Bessel function of the first kind.

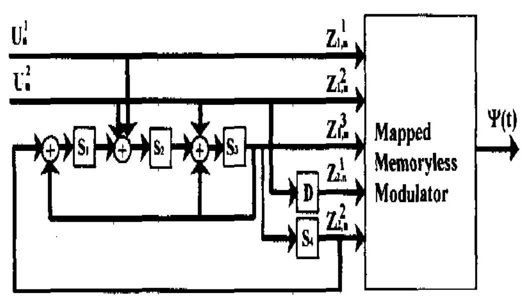

Fig. 3 A trellis-coded 8-ary CPFSK system (h=1/4)

Ⅲ. Reduced-state sequence detection

A digital transmission system operating over a time-dispersive channel is shown in Fig.4. The

www.dbpia.co.kr

complex symbols x_k are modulated, filtered, and transmitted over a time-dispersive channel corrupted by AWGN. The system employs a whitened matched filter whose outputs, $\{y_k\}$, are applied to a sequence estimator to obtain an estimate $\{\widehat{x_k}\}$ of the transmitted sequence. The transmit filter, channel, matched filter, and noise whitening filter can be modeled by the L+1 tap symbol-spaced transversal filter shown in Fig.5.

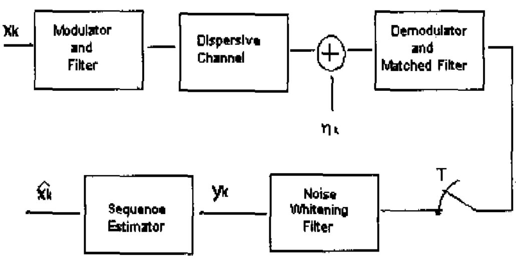

Fig. 4 Transmission system model

With this model, the received signal y_k is

$$y_k = \sum_{m=0}^{L} p_m x_{k-m} + \eta_k \tag{20}$$

where $\{\eta_k\}$ is a sequence of i.i.d. zero-mean complex Gaussian random variables with variance

$$\sigma_{\eta_k}^{\ 2} = \frac{1}{2} E\ [\ |\ \eta_k |^{\ 2}\ ] = N_0 \tag{21}$$

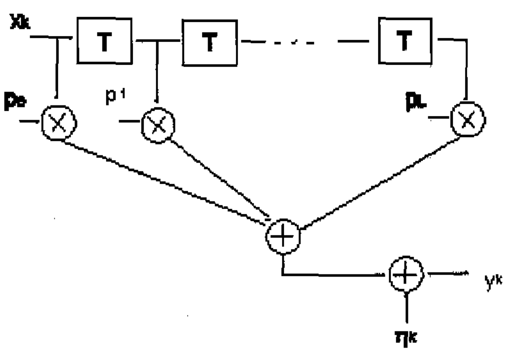

Fig. 5 Discrete-time white noise channel model

If the signal constellation has an alphabet size M, the channel is an M^L-state finite state machine. The system state at the time k is defined by $s_k^{(i)} = (x_{k-1}, x_{k-2}, \ldots, x_{k-L})$ for i=0, ..., M^L-1. To reduced to number of system states, each x_{k-i} in s_k is associated with a set partition in which the signal constellation is partitioned into J_i subsets, according to set partitioning principles. The system subset-state is defined as

$$t_k = [\, a_{k-1}(1), a_{k-2}(2), \ldots, a_{k-l}(l)\ ] \tag{22}$$

where $a_{k-i}(i)$ is the subset to which the transmitted symbol x_{k-i} belongs. Since $a_{k-i}(i)$ can only assume J_i possible values, there are $\prod_{i=1}^{l} J_i$ states in the subset-trellis which could be much less than M^L. Note that $J_1 < M$, there are parallel transitions associated with each subset-transition. The VA can be used for searching the subset-trellis as in MLSD, except for a different branch metric and possibility of parallel transitions associated with the subsettransitions. The performance of RSSD is worse than MLSD, because the reduction in the number of system states results in early merging. However, since an Ungerboeck-like set partitioning scheme has been employed, a good performance/complexity trade-off can always be expected. With RSSD, the branch metric given is not uniquely determined by the associated pair of subset states in the reduced state trellis, because of the reduction in the number of system states. To solve this difficulty, a decision feedback mechanism is introduced, and the branch metric for a particular parallel transition associated with subset transition (t_k, t_{k+1}) is calculated by

$$\widehat{r_k} = |\ y_k - \sum_{m=0}^{L} p_m \widetilde{x_{k-m}}\ |^{\ 2} \tag{23}$$

where $\widetilde{x_k}$ is the source symbol corresponding to this particular parallel transition, and $\{\widetilde{x_{k-i}}\}$ is the sequence corresponding to the path leading to subset-state t_k.

Ⅳ. A proposed adaptive RSSD receiver

In order to overcome the previously mentioned

www.dbpia.co.kr

problems of the conventional MLSD, an alternative algorithm is proposed in the following.

1) An adaptive RSSD with per-survivor method is used for the purpose of reducing the complexity and tentative decision delay of the MLSD.
2) A symbol-aided method is used for the improvement of tracking performance on time-varying channel estimation.

Fig. 6 shows the block diagram of the transmission model and Fig. 7 draws the flowchart of the proposed adaptive RSSD receiver.

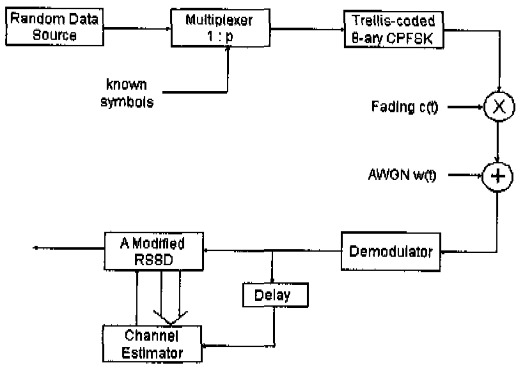

Fig. 6 The block diagram of the transmission model

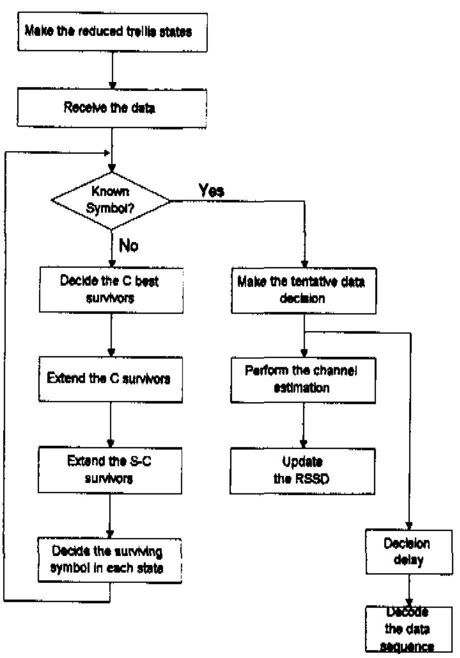

Fig. 7 The flowchart of the proposed adaptive RSSD receiver

Among the various suboptimal methods to reduce complexity of the MLSD, we concentrate on the RSSD combined with a per-survivor processing. The RSSD can nearly obtain the performance of the MLSD at a significantly reduced complexity. The primary idea is the construction of trellis with a reduced number of states. These states are formed by combining the states of the maximum likelihood trellis using Ungerboeck-like set partitioning principles. The RSSD is then implemented using the VA to search this reduced trellis. By using the RSSD, the trellis states are reduced in size by partially representing the ISI characteristics into the trellis diagram. Note that in MLSD, path histories usually contain the surviving state sequences leading to the current states. In RSSD, however, it may be more appropriate to store the actual surviving symbols since there is no one-to-one correspondence between state and symbol sequences.

Fig. 8, 9, 10 show the subset trellis diagrams. The procedures of an adaptive RSSD technique are described as follows :

1) A trellis diagram with reduced states is made by the Ungerboeck-like set partitioning principles.
2) A decoding process is performed through a per-survivor processing using the symbol-aided method.

Let S denote the number of survivors and C denote the number of ISI canceler. At each step of the decoding process :

① determine the C best survivors among the S survivors;
② extend each of the C best survivors;
③ extend each of the remaining S-C survivors in accordance to the code sequence of the best survivor.
④ make the tentative decisions for the channel estimation after the reception of a known symbol.

To be specific, consider the Fig. 11 which shows a few signal constellations at consecutive symbol intervals.

www.dbpia.co.kr

At $t=kT$, we have S candidate sequences, say $\widehat{a_k}(i)$, $i=1,2,\ldots,S$, which represent best guesses about true transmitted sequences $\boldsymbol{a_k}=(a_0,\ a_1,\ a_2,\ \ldots\ ,a_k)$. A metric is associated with each $\widehat{a_k}(i)$.

The $\widehat{c_k}(i) = [\,\widehat{c_0}(i), \widehat{c_1}(i), \ldots, \widehat{c_k}(i)\,]$ is the set of fading predictions corresponding to $\widehat{a_k}(i)$. For example, the last component of $\widehat{c_k}(i)$ is $\widehat{c_k}(i) = f\,[\,\widehat{a_{k-1}}(i)\,]$. The cumulative metric of $\widehat{a_k}(i)$ is defined as

$\Gamma[\widehat{a_k}(i)] = \sum_{j=0}^{k} |\, \widehat{r_j} - \widehat{c_j}(i)\,\widehat{a_j}(i)\,|^2$,where r_j is the input sequences of the MLSD.

Extend the survivors to t=(k+1)T. We should distinguish according to whether time (k+1)T corresponds to a data symbol or a known symbol.

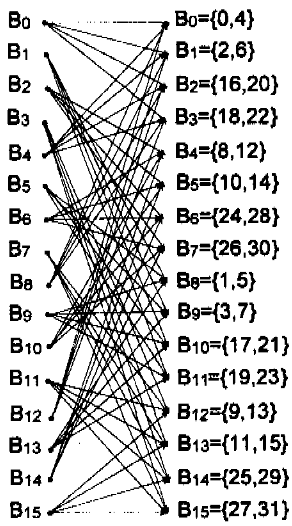

Fig. 8 Subset trellis diagram of the RSSD (16 states)

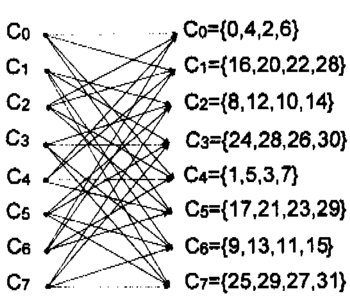

Fig. 9 Subset trellis diagram of the RSSD (8 states)

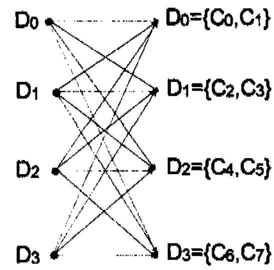

Fig. 10 Subset trellis diagram of the RSSD (4 states)

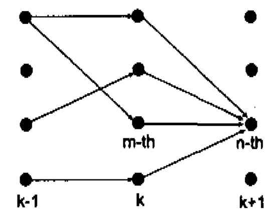

Fig. 11 An example of the extension of survivors utilizing the symbol-aided method

In the first case, for the C best survivors, we choose the minimum metric paths at t=(k+1)T. For example consider the n-th node at t=(k+1)T and call $\widetilde{a_{k+1}}(n)$ the associated symbol. The survivor reaching such a node is both the extension of $\widehat{a_k}(i)$ which satisfies

$$\widehat{a_k}(i) \leftarrow Min_{j=1,2,\ldots C}\{\Gamma[\widehat{a_k}(i)] + |\, r_{k+1} - f[\widehat{a_k}(j)]\,\widetilde{a_{k+1}}(n)\,|^2\} \qquad (24)$$

and the extension of the remaining S-C survivors in accordance to the code sequence of the best survivor. The $|\, r_{k+1} - f[\widehat{a_k}(j)]\,\widetilde{a_{k+1}}(n)\,|^2$ is the branch metric for the transition from the m-th node to the n-th node.

In the second case, the (k+1)T corresponds to a known symbol. This amounts to knowing the specific node. Then as the only survivor at t=(k+1)T ,we choose the best extension of the $\widehat{a_k}(i)$ to the specific node. We choose $\widehat{a_k}(i)$ such that

$$\widehat{a_k}(i) \leftarrow Min_{j=1,2,\ldots S}\{\Gamma[\widehat{a_k}(i)] + |\, r_{k+1} - f[\widehat{a_k}(j)]\,\widetilde{a_{k+1}}\,|^2\} \qquad (25)$$

Through the above mentioned symbol-aided

www.dbpia.co.kr

method, the tentative data decisions can be made for channel estimation every period between known symbols.

And by means of a per-survivor processing, the number of branches calculated in each state is reduced significantly. The trade-off between the complexity and the receiver performance depends on the choice of the number of ISI cancelers C . The concept of the symbol-aided method multiplexes periodically known symbols with the data stream for the purpose of reliable channel estimation. After the periodic reception of known symbols, the tentative data decisions of the Viterbi processor are used to the input of the channel estimator. The estimator approximates the actual channel with a linear finite state machine. The estimated channel path gains are adjusted by the steepest descent algorithm to minimize the mean-square error between the actual received sequence and a received sequence estimate formed from the output of the finite state machine model. The channel estimator using these tentative data decisions can track a time-varying channel fast and reliably with a small decision delay compared to using long decision delay sequences. An insertion period of the known symbols should be determined so that a time-varying channel can be tracked correctly.

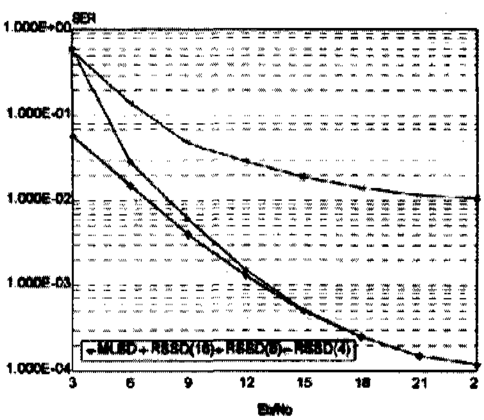

Fig. 12 The comparison of the MLSD and RSSD in the mobile satellite channel

In order to verify the proposed RSSD algorithm, we have been used computer simulations. For the Rician parameter, we set K=10. Fig.12 shows the comparison of the MLSD and RSSD in the mobile satellite channel. Through the RSSD using the set partitioning principle, we can reduce significantly the complexity of the MLSD. With 16 states, there is no noticeable degradation relative to the MLSD. With 8 states, a small degradation is shown.

Fig 13 indicates the comparison of the MLSD and RSSD with symbol-aided method in the mobile satellite channel. As the symbol period is decreased, the improvement of performance is obtained but the power loss is increased. The symbol period in this paper is set to five. By using the symbol-aided technique which exploits both data and known symbols to estimate channels, the improvement of the performance is obtained. With 8 and 16 states, there is no noticeable degradation compared to the MLSD.

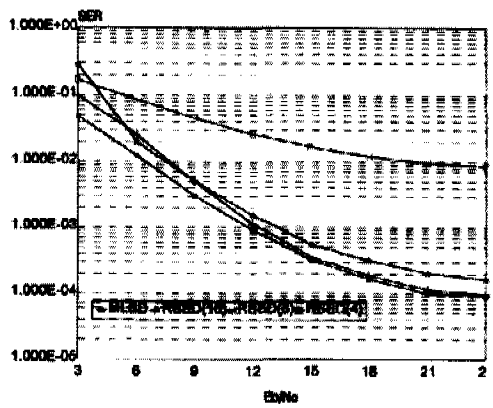

Fig. 13 The comparison of the MLSD and RSSD with symbol-aided method in the mobile satellite channel

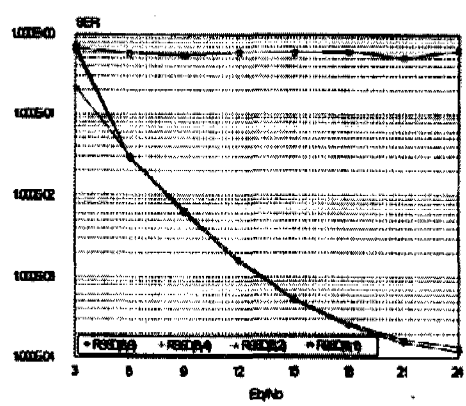

Fig. 14 An adaptive RSSD without symbol-aided method in the mobile satellite channel

www.dbpia.co.kr

Fig. 14, 15 are an adaptive RSSD without and with symbol-aided method in the mobile satellite channel. For S=8, the performance of the optimal (8, 8) receiver is obtained in the (8, 2) receiver with a degradation of a fraction of dB.

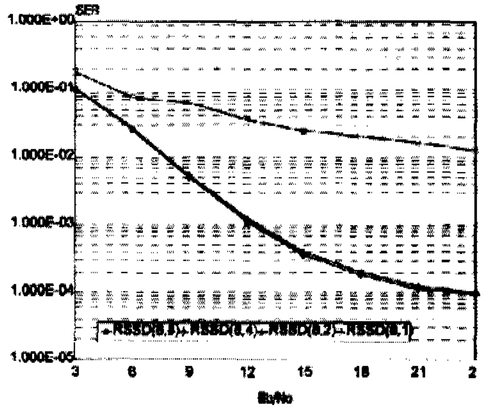

Fig. 15 An adaptive RSSD with symbol-aided method in the mobile satellite channel

V. Conclusion

In this paper, in order to overcome the problems of the conventional MLSD receiver, a new adaptive RSSD receiver is proposed. We propose an adaptive RSSD utilizing the symbol-aided technique for the purpose of reducing the complexity and tentative decision delay of the MLSD. The algorithm combines structures of the reduced-state VA and a decision feedback detection and provides a trade-off between complexity and performance. In spite of a suboptimal alternative receiver compared to the adaptive maximum likelihood detector, the proposed adaptive RSSD receiver is able to reduce the complexity significantly and track the time-varying channel fast and reliably. The used modulation method is a trellis-coded 8-ary CPFSK but the proposed adaptive RSSD receiver utilizing the symbol-aided method can be applied to other digital modulation techniques.

Reference

[1] T. Aulin and C-E Sundberg, "Continuous phase modulation-Part 1 : full response signaling", IEEE Trans. Commun., vol.COM-29, pp.196-209, Mar. 1981

[2] V. M. Eyuboglu and S. U. Qureshi, "Reduced-state sequence estimation with set partitioning and decision feedback", IEEE Trans. Commun., vol. COM-36. pp.13-20, January 1988

[3] G. D. Forney, Jr., "Maximum likelihood sequence estimation of digital sequence in the presence of intersymbol interference", IEEE Trans. Inform. Theory, vol. IT-18, pp.363-378, May 1972

[4] S. Haykin, Adaptive filter theory, New Jersey : Prentice-Hall, 1986

[5] J. G. Proakis, Digital Communications, New York : McGraw Hill, 1989

[6] Bixio E. Rimoldi, "A decomposition approach to CPM" , IEEE Trans. Inform. Theory, vol. 34, No.2, pp.260-270, March 1988.

[7] Richard H.-H. Yang, and Desmond, "Trellis-coded continuous-phase frequency-shift keying with ring convolutional codes", IEEE Trans. Inform. Theory, vol. 40, No.4, pp.1057-1067, July 1994.

[8] Brian D. Hart, and Desmond P. Taylor, "Maximum-likelihood synchronization, equalization, and sequence estimation for unknown time-varying frequency-selective Rician channels", IEEE Trans. Commun., vol.46, No.2, pp.211-221, Feb. 1998.

[9] E. Biglieri, D. Divsalar, P. J. McLane, and M. K. Simon, Introduction to trellis-coded modulation with applications, Macmillan, 1991.

[10] G. T. Irvine, and P. J. McLane, "Symbol-aided plus decision-directed reception for PSK/TCM modulation on shadowed mobile satellite fading channels," IEEE J. Select. Areas Commun., vol.10, pp.1289-1299, Oct. 1992.

[11] M. L. Moher, and J. H. Lodge, "TCMP - A modulation and coding strategy for fading channels," IEEE J. Select. Areas

www.dbpia.co.kr

Commun., vol.7, pp.1347-1355, Dec. 1989.

송 형 규(Hyoung-kyu Song) 정회원

1990년 2월 : 연세대학교 전자공학과 (학사)

1992년 2월 : 연세대학교 전자공학과 (석사)

1996년 2월 : 연세대학교 전자공학과 (박사)

1996년 1월~현재 : 전자부품연구소 시스템IC연구센터 선임연구원

<관심분야> 이동 및 개인 휴대 통신 시스템 및 디지털 신호 처리

www.dbpia.co.kr