行政院國家科學委員會專題研究計畫　期中進度報告

子計畫二：適用於高速移動擷取網路的等化碼技術與高階調變位元軟性決策解碼技術之研究(1/3)

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1. 中、英文摘要及關鍵詞(keywords)

由於載具在高速移動時，傳輸通道會呈現極快速時變(time-varying)的特性，導致很難進行高速率傳輸。有別於其他相關研究是專注於設計更好的方法來更精準估計的時變通道參數，本計畫改採以通道編碼為導向的做法解決時變通道的傳輸效能。計畫的第一年，已由非即時解碼(off-line decoding)的理念推展至快速時變的多路徑衰減通道解碼的設計理論；另外，也推導出以符元為傳送單位的高速率傳輸調變的最佳軟性解碼位元量度(soft bit metric)。

關鍵詞：時變多路徑衰減通道、通道估量、通道等化、錯誤更正碼

The main difficulty for high-bit-rate transmission under high mobility is on the tracking of the fast time-varying channel characteristic due to movement. Different from other researches who mostly focus on enhancing the accuracy of the channel parameter estimation and equalization, this project aims at a pure channel-coding approach, i.e., to combine the channel estimation and equalization into channel code design. The first year study of this project began with an off-line decoding assumption, and obtained the code design philosophy for resisting time-varying multipath interference. In addition, the derivation of a general soft bit-decomposed metric formula for symbol-based modulation scheme has been established in this year.

Keywords: Time-varying multipath fading channel, Channel estimation, Channel equalization, Error correcting coding

2.報告內容

2.1 Introduction and motivations:

The organization of present typical receivers for wireless communications mostly includes channel estimation and channel equalization devices in order to compensate the channel effect. A milestone research in 2002 by Skoglund, Giese and Parkvall [1] however demonstrated that a communication scheme which jointly considers error correcting code and multipath fading effect can achieve markedly better performance than a typical communication system even if perfect channel estimation and equalization is assumed. This exciting result provides a prospect that makes possible to have high data transmission rate for highly mobile users (that results fast time-varying fading).

2.2 The research procedures in this project:

Based on the above motivation, there are two issues on which we concentrate in this year. The first issue is to establish the optimal criterion for decoding the equalizer codes. The second issue is to derivate the metric formulas of the bit-wise soft-decision decoding for the symbol-based modulation. We describe the details in the following subsections.

2.2.1 Equalizer code design:

In the first issue, our goal is to establish a framework of a systematic equalizer code in the time-varying environment. An important step in this phase is to find the optimal decoding metric. In addition, we also need to examine the properties of the derived metric in order to help the code construction and subsequent decoder design. The procedure of our research is separated into several parts. (1) The mathematical expression of the considered fading channels should be well-defined. (2) Based on the chosen channel model, what the maximum-likelihood (ML) criterion is under i.i.d. input information bit sequence. (3) Based on the derived ML criterion, how to construct a code, and design its feasible decoding algorithm for the code. (4) Finally, to derive the channel capacity of the channel, and to examine whether our code can achieve the capacity of the channel.

Two types of fast time-varying channel models are often adopted in the literature. The first is not analyzable [3], for example, Jakes' model, second-order Butterworth and rectangular spectrum…etc. These models are widely accepted as realistic fading channel models and are usually utilized in simulations; however, they are mostly not analytically tractable. The second type of fast time-varying channel models includes autoregressive (AR) model, time-independent model, polynomial model [8] and quasi-static channel, which are basically analyzable. In this project, we chose the first-order AR model as our research goal. This model is often named Gauss-Markov model, which is defined as:

$$r_k = \mathbf{a}_k^T \mathbf{h}_k + n_k \quad \text{and} \quad \mathbf{h}_k = \alpha \mathbf{h}_{k-1} + \mathbf{v}_k$$

where k=1,2,…,N, $\mathbf{v}_k$ is complex white Gaussian with mean $\mathbf{d}$ and covariance $\mathbf{C}$, $\mathbf{a}_k$ is the input, and α is a constant. The Gauss-Markov channel has been shown to be a good model to emulate a true fading channel [4]. The usual time-independent channel model and quasi-static channel model are its special cases.

By denoting $\mathbf{A} = [\mathbf{a}_1,\ldots,\mathbf{a}_N]^T$ and $\mathbf{H} = [\mathbf{h}_1,\ldots,\mathbf{h}_N]$, we can derivate the ML decoding criterion for the model through:

$$\Pr(\mathbf{r} \mid \mathbf{A}) = \int_H \Pr(\mathbf{r} \mid \mathbf{A},\mathbf{H}) \Pr(\mathbf{H}) d\mathbf{H}$$

$$= \int_{h_N} \int_{h_{N-1}} \Lambda \int_{h_1} \left(\prod_{k=1}^{N} e^{-\frac{|r_k - \mathbf{a}_k^T \mathbf{h}_k|^2}{\sigma^2}}) \right) \prod_{k=1}^{N} \Pr(\mathbf{h}_k \mid \mathbf{h}_{k-1}) d\mathbf{h}_1 d\mathbf{h}_2 \Lambda \, d\mathbf{h}_N$$

The ML criterion for Gauss-Markov channel is

$$\Pr(\mathbf{r} \mid \mathbf{A}) \equiv e^{\mathbf{q}_N^H \mathbf{G}_N \mathbf{q}_N} \mid \mathbf{G}_N \mid \prod_{k=1}^{N-1} e^{(\mathbf{q}_k - \alpha^* \mathbf{C}^{-1}\mathbf{d})^H \mathbf{G}_k (\mathbf{q}_k - \alpha^* \mathbf{C}^{-1}\mathbf{d})} \mid \mathbf{G}_k \mid \qquad (1)$$

where "≡" means "equal" except for a multiplicative constant, and for 1<k<N

$$\begin{cases} \mathbf{G}_k = \left(\dfrac{\mathbf{a}_k^* \mathbf{a}_k^T}{\sigma^2} + (1+|\alpha|^2)\mathbf{C}^{-1} - |\alpha|^2 \, \mathbf{C}^{-1}\mathbf{G}_{k-1}\mathbf{C}^{-1} \right)^{-1} \\ \mathbf{q}_k = \dfrac{r_k \mathbf{a}_k^*}{\sigma^2} + \mathbf{C}^{-1}\mathbf{d} + \alpha \mathbf{C}^{-1}\mathbf{G}_{k-1}\mathbf{q}_{k-1} - |\alpha|^2 \, \mathbf{C}^{-1}\mathbf{G}_{k-1}\mathbf{C}^{-1} d \end{cases}$$

and

$$\begin{cases} \mathbf{G}_1 = \left(\dfrac{\mathbf{a}_1^* \mathbf{a}_1^T}{\sigma^2} + (1+|\alpha|^2)\mathbf{C}^{-1} \right)^{-1} \\ \mathbf{q}_1 = \dfrac{r_k \mathbf{a}_k^*}{\sigma^2} + \mathbf{C}^{-1}\mathbf{d} + \alpha \mathbf{C}^{-1}\mathbf{h}_0 \end{cases}$$

$$\begin{cases} \mathbf{G}_N = \left(\dfrac{\mathbf{a}_N^* \mathbf{a}_N^T}{\sigma^2} + \mathbf{C}^{-1} - |\alpha|^2 \, \mathbf{C}^{-1}\mathbf{G}_{N-1}\mathbf{C}^{-1} \right)^{-1} \\ \mathbf{q}_N = \dfrac{r_N \mathbf{a}_N^*}{\sigma^2} + \mathbf{C}^{-1}\mathbf{d} + \alpha \mathbf{C}^{-1}\mathbf{G}_{N-1}\mathbf{q}_{N-1} - |\alpha|^2 \, \mathbf{C}^{-1}\mathbf{G}_{N-1}\mathbf{C}^{-1}\mathbf{d} \end{cases}$$

Notably, our criterion is an extension of that in [2], in which zero-mean for $\mathbf{H} = [\mathbf{h}_1,\ldots,\mathbf{h}_N]$is assumed.

On the decoding algorithm, because $\mathbf{G}_k$ and $\mathbf{q}_k$ are functions of the entire information sequence, the standard Viterbi algorithm (VA) can not be applied. In [2], the authors employed the list Viterbi algorithm (LVA) to prune out some less likely data sequences, and showed that it is not beneficial to keep more than three survivor paths at each trellis node. In this project, we considered two alternative approaches: metric prediction and iterative decoding approach. On the metric prediction, we conjecture that the LVA (even with L = 3) performance can be achieved by the standard VA (L = 1) if a proper prediction metric can be added to the above derived metric. For example, in a flat fading environment, $\mathbf{G}_k$ and $\mathbf{q}_k$ in (1) are simply scalars can be reduced to scalars. Also, G_k remains almost the same for k = 1…N. Let the mean d=0. Then a prediction metric can be set as:

$$h(r_j) = \mu_j C_j + q_j \cdot 2BD_j$$

where

$\mu_j = q_j^2$, $C_j = \frac{1}{A^2} C_{j-1}$, $D_j = \frac{1}{A} D_{j-1} - |r_j|(1+C_j)$, $A = \frac{\alpha G}{\sigma_v^2}$ and $B = \frac{1}{\sigma^2}$.

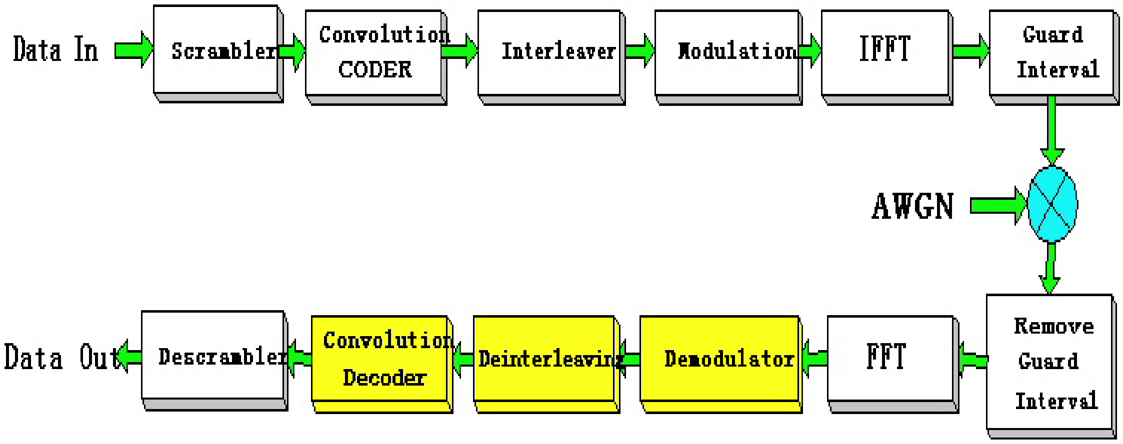
This result can gradually migrate to frequency-selective fading channels. In addition, the recursive property of the ML criterion can be easy to use in an iterative decoding algorithm, such as Soft-Output Viterbi algorithm (SOVA).

Our current focus is on the code construction (possibly non-linear) for the chosen Gauss-Markov channel. Our approach is basically to derive the pairwise error probability for two candidate codewords, and then uses simulated annealing algorithm to search for a good code.

Finally, the mutual information of the Gauss-Markov channel is examined so that it can be applied to the evaluation of the channel capacity of the Gauss-Markov channel with unknown channel state information (CSI).

2.2.2 Bit-wise decomposition of *M*-ary maximum-likelihood symbol metric:

Another result that we obtained in this year is a systematic recursive formula for bit-wise decomposition of *M*-ary symbol metric. The decomposed bit metrics can be applied to improve the performance of a system where the information sequence is binary-coded and interleaved before *M*-ary modulated. A straightforward receiver designed for certain system is to de-map the received *M*-ary symbol into its binary isomorphism so as to facilitate the subsequent bit-based manipulation, such as hard-decision decoding. With a bit-wise decomposition of *M*-ary symbol metric, a soft-decision decoder can be used to achieve a better system performance. The idea behind the systematic formula is to decompose the symbol-based maximum-likelihood (ML) metric by equating a number of specific equations that are drawn from squared-error criterion. It interestingly yields a systematic recursive formula that can be applied to some previous work derived from different standpoint. Simulation results based on IEEE 802.11a/g [6][7] standard show that at bit-error rate of 10^{-5}, the proposed bit-wise decomposed metric can provide 3.0 dB, 3.9 dB and 5.1 dB improvement over the concatenation of binary-demapper, deinterleaver and hard-decision-decoder for 16QAM, 64QAM and 256QAM symbols, respectively. Also, only 0.13 dB performance degradation is resulted by introducing 32-level quantization for 16QAM signals. The quantization impact for 64QAM signals under 64-level uniform quantization can even be reduced to 0.07 dB. No further performance degradation, in addition to that due to quantization, can be observed, when mismatch of AGC gain is limited to be within $\pm$ 40%. The robustness of the proposed bit-decomposed metric against phase noise is also examined. When the phase drift increases up to $\pm 6^{\circ}$, the BER due to our bit-decomposed metric will increase from 10^{-5} to around 4×10^{-5} at E_b/N_0 = 6.7 dB for 16QAM modulation. This phase drift tolerance reduces to $\pm 4^{\circ}$ at E_b/N_0 = 9.7 dB for 64QAM modulation, where E_b/N_0 is chosen such that the no-phase-drift BER is approximately 10^{-5}. The system architecture considered is as follows.



Our goal is to find the functions $f_i\,(c_i\,,\ r)$ to approximate symbol-based Maximum-likelihood metric:

$$\sum_{i=1}^{N} f_i(c_i,r) \approx \min_{s\in S}\sum_{i=1}^{K}(r_i - s_i)^2$$

The criterion we adopt is the minimization of average square error, namely,

$$\min_{f_1,f_2} E[(f_1(c,r)+f_2(\bar{c},r)-[r-s(c,\bar{c})]^2)^2]$$

We can get these sub-optimum functions in M^2-QAM systems

$$\begin{cases} f_1^{16QAM}(c,r) = c\,|\,r\,|\bullet \operatorname{sgn}(-r) \\ f_2^{16QAM}(c,r) = c(|\,r\,|-2) \end{cases}$$

$$\begin{cases} f_1^{64QAM}(c,r) = c(|\,r-4\,|+|\,r\,|+|\,r+4\,|-8)\bullet \operatorname{sgn}(-r) \\ f_2^{64QAM}(c,r) = f_1^{16QAM}(c,4-|\,r\,|) \\ f_3^{64QAM}(c,r) = f_2^{16QAM}(c,|\,r\,|-4) \end{cases}$$

$$\begin{cases} f_1^{(m)}(c,r) = c\bullet \operatorname{sgn}(-r)(\sum_{i=-(m-2)}^{m-2}(|\,r+2ui\,|-|\,2ui\,|)) \\ f_j^{(m)}(c,r) = f_{j-1}^{(m-1)}(c,(-1)^j[2^{m-2}u-|\,r\,|]),\ \text{for } 2\le j\le m \end{cases}$$

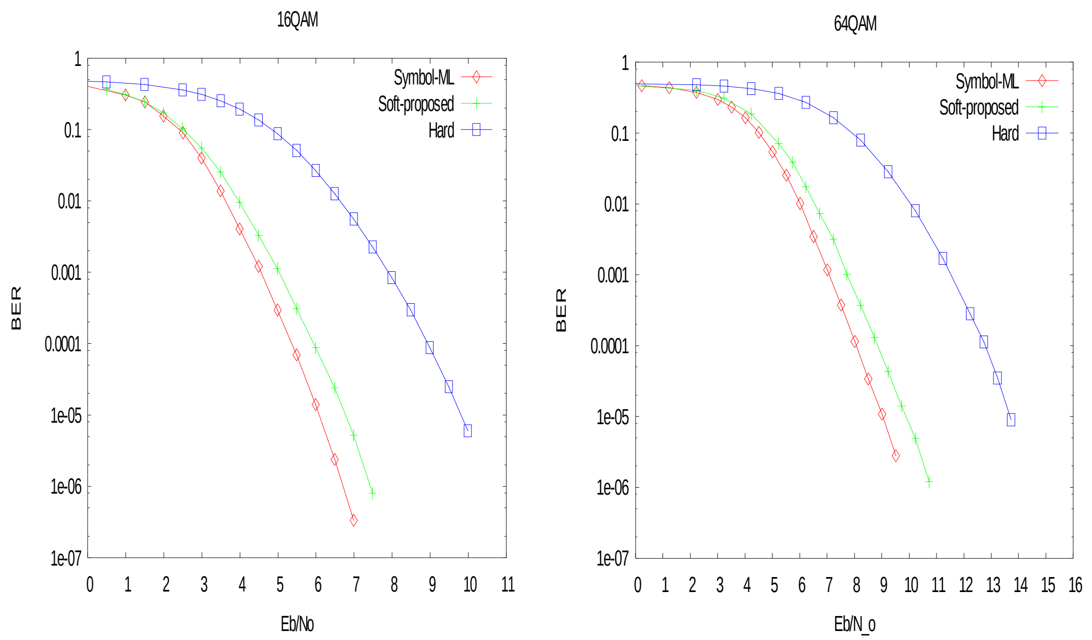
2.3 Achievement:

2.3.1 Equalizer codes technology:

We derived the ML metric for nonzero-mean-channel-response Gauss-Markov channel, and developed a suboptimum metric for off-line Viterbi algorithm, where the optimal heuristic function for the fast fading channels with Gauss-Markov model is showed. Our ongoing aim is on the design and performance evaluation of iterate decoding algorithm for time-varying Gauss-Markov channel.

2.3.2 Bit-wise decomposition of *M*-ary maximum-likelihood symbol metrics:

The two figures below reveal our Soft-proposed metric can further improve the performance of hard-decision, and approach to ideal symbol-ML performance. Further empirical study on system imperfection implies that the proposed bit-wise decomposed metric also improves the system robustness against gain-mismatch and phase-noise.



3. References

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4. 計畫成果自評

Accordingly to the project proposal, our aims in the first year are to develop and examine the design rule of equalizer codes in a time-varying multipath fading environment, and to derive the soft bit metric of symbol-oriented high-speed modulation. Both aims have been achieved in this year.

In addition, we demonstrated a low-complex suboptimum metric prediction approach and verified the performance of our soft bit metric by simulation based on IEEE 802.11 a/g standard. The result of the second part has already been published in APCCS, 2003, and has been submitted to IEEE trans. on wireless communications for reviews. In the first part, i.e., the part of equalizer codes design, our result will be verified by simulations in next year. If the BER approximate the performance of ML decision, it can considerably reduce the decoding complexity. We will also compare it with other known suboptimum ML approach, such as LVA.