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Amplify-Forward Relay Network의 인터리버에 근거한 협동 다이버시티와 그 효과적 알고리즘

(Cooperative Diversity Based on Interleavers and Its efficient Algorithm in Amplify-And-Forward Relay Networks)

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요 약

참고문헌 [1]에서 full diversity를 달성할 수 있을 뿐만 아니라 각 relay node에서의 시간 지연을 제거하기 위해서 새로운 기법을 제안했다. 그러나 복호 후 전달 (DF) 모델 연산 모드는 relay에 더욱 많은 처리 부담을 준다. 본 논문에서는 [2],[3]에서 제안한 증폭 후 전달 (AF) model로 확장할 뿐만 아니라 source-relay link와 relay-destination link로 이루어진 모든 채널의 결합 채널 계수의 순서를 정하고 다음 복호 절차에서 이전의 복호된 심벌을 지우는 효율적인 복호 알고리즘을 제안한다. 컴퓨터 모의실험 결과를 통하여 제안한 알고리즘을 이용하면 높은 SNR 영역에서 참고문헌 [1]과 비교하여 상대적으로 2-3dB의 이득을 가져오기에 효율적으로 성능을 향상시킬 수 있다는 것을 알 수 있었다. 또한 제안한 알고리즘이 원래의 알고리즘과 비교하여 상대적으로 3dB 이상의 이득을 얻을 수 있는 복호 후 전달 (DF)에 유용하다는 것을 알 수 있었다.

Abstract

In [1], the authors have proposed a novel scheme to achieve full diversity and to combat the time delays from each relay node, but decode-and-forward (DF) model operation mode puts more processing burden on the relay. In this paper, we not only extend their model into amplify and forward (AF) model proposed in [2],[3], but also propose an efficient decoding algorithm, which is able to order the joint channel coefficients of overall channel consisting of source-relay link and relay-destination link and cancels the previous decoded symbols at the next decoding procedure. The simulation results show that this algorithm efficiently improves its performance achieving 2-3dB gain compared to [1] in high SNR region and also useful to DF achieving more than 3dB gain compared to an original algorithm.

Keywords : Interleaver, fading channels, relay networks, amplify-and-forward.

I. INTRODUCTION

Multiple-input multiple-output (MIMO) architecture can potentially enhance communication spectral efficiency in multipath wireless channels[4]. However, in rich multipath environment is characterized by wavelength-scale fast fading. The rapid channel variation for highly mobile nodes limits the quality of obtainable channel state information (CSI), which in turn reduces the channel capacity[5]. In fact, for an independent and identically distributed (i.i.d) Gaussian channel which remains constant over the block of length T. In contrast to the result for a constant channel, the capacity versus the number of transmit antennas achieves a maximum for T antennas[6].

More recently, cooperative diversity has attracted much attention for its wide applications in wireless communication networks[2, 7]. Cooperative relay of other users or terminals is used to achieve high

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diversity and high energy efficiency. Amplify forward and decoding forward are the commonly used protocols for cooperative diversity in wireless communication networks. In both amplify forward and decoding forward protocols, the signal/information received by the relay terminals are directly amplify-forwarded or decode-forwarded to the destination or to the next relay terminals. For amplify-forward protocol, an amplification of a received signal is transmitted from each relay terminal. In this protocol, the information signal as well as the received noise are amplified and forwarded to the destination. As remarked in [2], the AF operation mode puts less processing burden on the relay and hence, it is often preferable. In addition, we assume that AF relays actually outperform DF relays under certain conditions for the decoding forward protocol. The received signal at each relay terminal is first decoded and then transmitted after properly encoded. In this protocol, the received noise is not forwarded to the destination unless the information symbol can be correctly detected. In most of the available schemes for the decoding forward protocol, the detected information symbols are directly forwarded to the destination or used in a distributed space-time code design. The detected information symbols or the complex conjugate of the detected information symbols are sent to relay terminals. For convenience to discuss, only amplify forward is considered in the rest of our paper.

In [1], a simple uncoordinated cooperation scheme is based on interleaving, where each cooperating termi- nal simply interleaves the detected bits using an interleaver. It is proposed in an asynchronous network when considering with fixed decode-interleaver-and-forward (FDIF). In this paper, we consider an Amplify-Forward model due to its efficient processing on relay nodes locating on inteleaver at each relay node, namely FAIF model. Also propose an efficient algorithm to improve its performance, which assumes that the CSI is known at receiver and cancels the previous decoded symbols for the next decoding procedure.

This paper is organized as follow. Section II describes the system model in consideration. Our proposed protocol and its efficient algorithm are introduced briefly in Section III. The proposed scheme is described and analyzed in Section IV. The performance is verified by computer simulation in Section V. Finally, Section VI concludes this paper.

II. SYSTEM MODEL

Consider a wireless network with M randomly placed relay nodes one $\mathbb{R}_i, i = 1,\cdots,M$, source node $\mathbb{S}$ and one destination node $\mathbb{D}$ as shown in Fig. 1. Every node has only single antenna that cannot transmit and receive simultaneously. The channel between each pair of node is assumed quasi-stationary Ray-leigh flat fading which is constant within one frame but may vary from frame to frame. Denote the channel from $\mathbb{S}$ to $\mathbb{D}$ as f, $\mathbb{S}$ to $\mathbb{R}_i$ as g_i and $\mathbb{R}_i$ to $\mathbb{D}$ as h_i respectively ; named with $f \in \mathcal{CN}(0,\sigma_f^2)$, $g_i \in \mathcal{CN}(0,\sigma_g^2)$ and $h_i \in \mathcal{CN}(0,\sigma_h^2)$. Throughout this paper, we assume an asynchronous case in all terminals.

The relay based transmission is usually divided into two phase. During phase I (broadcasting phase), the source broadcast its own information bits to all relays and destination. During phase II (cooperative phase), the relays would either choose to purely amplify and retransmit the information to the

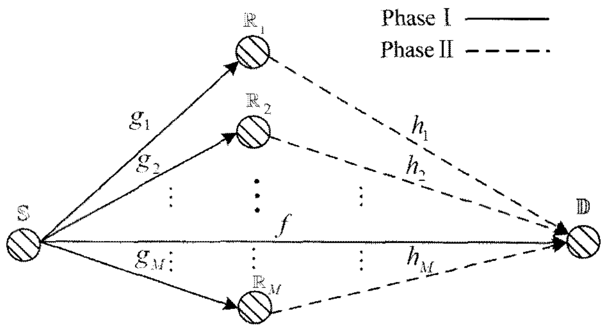

그림 1. M relay node를 갖는 relay networks
Fig. 1. A relay networks with M relay.

destination, or decode the information first and then transmit these information bits to the destination. The former process is referred as amplify-and-forward (AF) and latter it is referred as decode-and-forward (DF). As compare with [1], in this paper we only consider a simple different model amplify-interleave-and-forward protocol. In our case, all relay nodes are active in cooperative phase.

During phase Ⅰ, considering quadrature-phase-shift-keying (QPSK) signaling $x(n)=\{\pm1,\pm i\}$, the source node $\mathbb{S}$ transmits a information bits sequence with length N, where $x_s(n), n=0,1,\cdots,N-1$. The received signals at destination node in phase Ⅰ can be represented as

$$r_{s,d}(n)=fx_s(n)+n_{s,d1}(n), \tag{1}$$

and the received signals at each relay node $\mathbb{R}_i, i=1,\cdots,M$, can also be expressed as

$$r_{s,r_i}(n)=g_i x_s(n)+n_{s,r_i}(n), \tag{2}$$

where $n_{s,d}(n)$ and $n_{s,r_i}(n)$ are the complex AWGN processes with zero mean and variance σ^2 per dimension at destination node $\mathbb{D}$ during phase Ⅰ and at relay nodes $\mathbb{R}_i$, where $i=1,\ldots,M$. During phase Ⅱ, the power constraint of the transmitted sequence is $E\{x_s^H(n)x_s(n)\}=NE_s$, where E_s is the average transmitting power of source and "H" is the Hermitian of a vector or a matrix. Then $r_{s,r_i}(n)$ is scaled by real factor to keep the average power of $\mathbb{R}_i$ as E_n.

$$t_i(n)=a_i r_{s,r_i} \tag{3}$$

Basically, there are two different choices in a_i, which are listed as follows :

$$a_i=\sqrt{\frac{E_{ri}}{\sigma_{gi}^2 E_s}+N_0}, \tag{4}$$

or

$$a_i=\sqrt{\frac{E_{ri}}{|g_i|^2 E_s}+N_0}. \tag{5}$$

Here, we recommend (5) as a choice of a_i which is not a random value while keeps the power constraint from long term point of view. Otherwise g_i could be replaced by its estimated value which may not keep the average power exactly as E_n. If amplify-interleave-and-forward protocol is used here, then the scaled sequence can be interleaved by a random interleaver π_i, which is known at the source node $\mathbb{S}$ and destination node $\mathbb{D}$. Then the information bits at relay node can be expressed as

$$x_k(n)=\pi_m[x_s(n)], n=0,1,\cdots\cdots,N-1,$$

where $\pi_m[\cdot]$ is denoted as an interleaver operator. The scaled signal at each relay node can be rewritten as

$$t_{k,j}(n)=a_i g_i x_k(n)+a_i n_{s,r_i}(n). \tag{6}$$

Also the received signal at destination node $\mathbb{D}$ in the cooperative phase can be expressed as :

$$\begin{aligned} r_d(n) &= \sum_{i=1}^{M} a_i h_i t_{k,j}(n-\tau_m)+fx_s(n-\tau_s)+n_{s,d2}(n) \\ &= \sum_{i=1}^{M} a_i g_i h_i x_k(n-\tau_m)+fx_s(n-\tau_s)+n_d(n) \end{aligned} \tag{7}$$

where $n_d(n)=n_{s,d2}(n)+\sum_{i=1}^{M}a_i h_i n_{s,r_i}(n)$ with mean zero variance $\left(\sum_{i=1}^{M}|a_i h_i|^2+1\right)\sigma^2$, and τ_m,τ_s are delays from source node $\mathbb{S}$ and relay node $\mathbb{R}_i$, respectively.

Ⅲ. A SIMPLE DECODING ALGORITHM

A. Brief introduction to conventional algorithm

Similar to [1] and [8], we will introduce a simple symbol-by-symbol iterative algorithm based on Multi- user Detection concept. Here, initial prior information of $x_s(n)$ can be derived from (1) by using Maximum Likelihood Criterion, and (6) is similar to Multiuser signal model employs our proposed iterative algorithm to detect information bits of $x_s(n)$. We will use superscripts "Re" and "Im" or function notation $\mathrm{Re}(\cdot)$ and $\mathrm{Im}(\cdot)$ to indicate the real and imaginary parts.

The initial condition of information bits can be derived from (1) during phase Ⅰ. Then the initial log likelihood ratios (LLRs) can be computed by

$$L_1(x_s^{\mathrm{Re}}(n)) = 2\,\mathrm{Re}(f^* r_{s,d}(n))/\sigma^2,$$
$$L_1(x_s^{\mathrm{Im}}(n)) = 2\,\mathrm{Im}(f^* r_{s,d}(n))/\sigma^2. \tag{8}$$

These initial information are served as prior information at phase Ⅱ, such as $L_2^1(x_k(n)) = L_1^0(x_s(\pi(n)))$, where the superscript i of $L_j^i(x_s(n))$ denotes the $i-th$ iteration and j of $L_j^i(x_s(n))$ denotes the transmission phase. Assume that channel coefficients of f, g_i and h_i are known at the destination and define a newly combined random variable $w_i = h_i \cdot g_i$. While detecting first symbol, we can select k^{th} symbol whose combined channel gain is maximum $w_{\max}$ among w_i for $i = 1,\cdots\cdots,M$ as the first desired symbol. Then, we rewrite (6) as

$$r_d(n+\tau_k) = a_k w_{\max 1} x_k(n) + \underbrace{\sum_{i\neq k} a_i w_i x_i(n+\tau_k-\tau_m) + n_d(n)}_{distortion\ \ \zeta(k)} \tag{9}$$

The distortion $\zeta(k)$ with respect to $x_k(n)$ consists of interferences and Gaussian noise. According to central limit theorem for large M, $\zeta(k)$ can be approximated as a complex Gaussian random variable with mean $E[\zeta(k)]$ and variance $Var[\zeta(k)]$. So,

$$E[\zeta(k)] = \sum_{i\neq k} a_i w_i E[x_i(n)]$$
$$Var[\zeta(k)] = \sum_{i\neq k} a_i^2 w_i^2 Var[x_i(n)] + \left(\sum_{i=1}^{M} |a_i|^2 |h_i|^2 + 1\right)\sigma^2, \tag{10}$$

where $E[\cdot]$ and $Var[\cdot]$ denote the functions of mean and variance. At each iteration, $E[x_k(n)]$ and $Var[x_k(n)]$ are updated using prior information $L_j^i(x_m(n))$. Therefore,

$$E[x_k^{\mathrm{Re}}(n)] = \tanh(L_j^i(x_s^{\mathrm{Re}}(n))/2)$$
$$Var[x_k^{\mathrm{Re}}(n)] = 1 - |E[x_k^{\mathrm{Re}}(n)]|^2. \tag{11}$$

The $E[x_k^{\mathrm{Im}}(n)]$ and $Var[x_k^{\mathrm{Im}}(n)]$ can also be approximated following by the same way. The procedure of this decoding is the same with [8]. So we are not going to introduce any thing more and please refer [1, 8].

B. An introduction to our proposed algorithm

In this section, we will introduce our proposed algorithm in detail. For this algorithm, we combine ordering and canceling techniques compared to the conventional algorithm[8] due to difference of channel coefficients. In first step of our proposed algorithm is, we order the channel vector $\mathbf{W} = [w_1,\cdots\cdots,w_{\max}]$ by multiplying a permutation matrix $\mathbf{P}$ and the corresponding transmitted signals are also ordered by the same permutation matrix. So, we can obtain $\tilde{\mathbf{W}} = \mathbf{PW}$ and $\tilde{\mathbf{X}} = \mathbf{PX}$. Then the first entry of $\tilde{\mathbf{W}}$ is the maximum channel coefficient in vector $\mathbf{W}$ and the last entry of $\tilde{\mathbf{W}}$ is the minimum channel coefficient in vector $\mathbf{W}$. After decoding the information symbol $\tilde{x}_0(n)$ with maximum channel gain $\tilde{w}_0$, the another information symbol $\tilde{x}_1(n)$ with the second maximum channel gain $\tilde{w}_1$ is waited for decoding. In second step, since $\tilde{\mathbf{X}} = \mathbf{PX}$ by ordering channel gains and canceling the information symbol $\hat{\tilde{x}}_0(n)$, the formula for second decoding, $\tilde{x}_1(n)$, can be written equivalently as

$$\begin{aligned} r_d(n+\tau_k)-\tilde{a}_0\tilde{w}_0\hat{\tilde{x}}_0(n+\tau_k) &= \tilde{a}_1\tilde{w}_1\tilde{x}_1(n) \\ &+ \sum_{i\neq 0,1}\tilde{a}_i\tilde{w}_i\tilde{x}_i(n+\tau_k-\tau_m) \\ &+ \tilde{a}_0\tilde{w}_0(\tilde{x}_0(n)-\hat{\tilde{x}}_0(n))+n_d(n) \end{aligned} \tag{12}$$

In order to use this decoding method, we have to evaluate $E[(\tilde{x}_k(n)-\hat{\tilde{x}}_k(n))]$ and $Var[(\tilde{x}_k(n)-\hat{\tilde{x}}_k(n))]$. After cancelling the estimated information bit, the mean can be evaluated by

$$E[\tilde{x}_k(n)-\hat{\tilde{x}}_k(n)]=E[\tilde{x}_k^{i-1}(n)]-E[\hat{\tilde{x}}_k^{i}(n)], i=0,1,\cdots,l-1,$$

where i is number of iteration times and $E[\tilde{x}_k^{i-1}(n)]$ can be evaluated by the extrinsic LLRs using (11). For example : while cancelling the first iteratively estimated information bits, the mean is estimated, where the statistics of the first information is available statistics with mean zero and variance one. Then the variance can also be evaluated by $\tilde{x}_m(n)$

$$Var[\tilde{x}_k(n)-\hat{\tilde{x}}_k(n)]=Var[\tilde{x}_k^{i-1}(n)]-Var[\hat{\tilde{x}}_k^{i}(n)],$$

$i=0,1,\cdots\cdots,l-1,$ and the principle of evaluating variance is similar to the evaluation of mean. Then, we can generalize this decoding algorithm to detect and estimate the m^{th} transmitted signal with m^{th} maximum channel gain from the relay node to destination.

$$\begin{aligned} & r_d(n+\tau_k)-\sum_{k\le m}\tilde{a}_k\tilde{w}_k\hat{\tilde{x}}_k(n+\tau_k) \\ &=\tilde{a}_m\tilde{w}_m\tilde{x}_m(n)+\sum_{i>m}\tilde{a}_i\tilde{w}_i\tilde{x}_i(n+\tau_k-\tau_m) \\ &\quad+\sum_{k\le m}\tilde{a}_k\tilde{w}_k(\tilde{x}_k(n)-\hat{\tilde{x}}_k(n))+n_d(n), \end{aligned} \tag{13}$$

where the index i means that these transmitted symbol has been decoded by (1), and the index k consists of undecoded symbols. Denote $y_m(n)$ as

$$r_d(n+\tau_k)-\sum_{k\le m-1}\tilde{a}_k\tilde{w}_k\hat{\tilde{x}}_k(n+\tau_k).$$

The central limit theorem can also be accustomed to approximate the mean and variance of the term $\sum_{k\le m-1}\tilde{a}_k\tilde{w}_k\hat{\tilde{x}}_k(n+\tau_m-\tau_k)$ and the residual part of (13). Here, denotes $E[\tilde{\tilde{x}}_k(n)]=E[(\tilde{x}_k(n)-\hat{\tilde{x}}_k(n))]$ and $Var[\tilde{\tilde{x}}_k(n)]=E[(\tilde{x}_k(n)-\hat{\tilde{x}}_k(n))]$, and the distortion after ordering and canceling is $\tilde{\zeta}_m(k)$ including the interference terms of undecoded part in (13) and decoded part in (13) plus noise $n_d(n)$. The mean and variance of $\tilde{\zeta}_m(k)$ can also be approximate by central limit theorem similar to (10).

$$\begin{aligned} & E[\tilde{\zeta}_m(k)]=\sum_{k\le m-1}\tilde{a}_k\tilde{w}_kE[\tilde{\tilde{x}}_k(n)]+\sum_{i>m}\tilde{a}_iw_iE[\tilde{x}_i(n)] \\ & Var[\tilde{\zeta}_m(k)]=\sum_{k\le m-1}\tilde{a}_k^2\tilde{w}_k^2Var[\tilde{\tilde{x}}_k(n)]+ \\ & \sum_{i>m}a_i^2w_i^2Var[\tilde{x}_i(n)]+\left(\sum_{j=1}^{M}|a_j|^2|h_j|^2+1\right)\sigma^2, m=2,3,\cdots,M \end{aligned} \tag{14}$$

Then, the real part of LLR of $\tilde{x}_m(n)$ can be computed as

$$LLR(\tilde{x}_m^{\rm Re}(n))=\frac{2\tilde{w}_m(y_m^{\rm Re}(n)-E[\tilde{\zeta}_m^{\rm Re}(k)])}{Var[\tilde{\zeta}_m^{\rm Re}(k)]}, \tag{15}$$

and the imaginary part of LLR of $\tilde{x}_m(n)$ is also computed by the same way. From Fig. 1, we have $\tilde{x}_s(n)=\tilde{x}_m(\pi^{-1}(n))$ where π^{-1} is a deinterleaving operation. Then the i-th LLRs is combining the LLRs from phase Ⅱ with LLRs from phase Ⅰ initial LLRs.

$$L^i(\tilde{x}_s(n))=\sum_{m=1}^{M}L_2^i(\tilde{x}_m(\pi^{-1}(n)))+L_2^i(\tilde{x}_s(n))+L_1^0(\tilde{x}_s(n)) \tag{16}$$

For each iteration, it can be updated from last iteration by

$$\begin{aligned} & L^{i+1}(\tilde{x}_s(n))=L^i(\tilde{x}_s(n))-L_2^i(\tilde{x}_s(n)) \\ & L^{i+1}(\tilde{x}_m(n))=L^i(\tilde{x}_s((\pi^{-1}(n)))-L_2^i(\tilde{x}_m(n)). \end{aligned} \tag{17}$$

Then, using (11) to compute mean and variance of $\tilde{x}_m(k)$ by (9), and computation of mean and variance of distortion $\tilde{\zeta}_m(k)$ can also be updated by (13). Compared with conventional receiver in [1], the computation of LLRs of our proposed algorithm is slightly more complicated and additional work is to estimate distortion signal $\tilde{\zeta}_m(k)$ for decoding the transmitted signal $\tilde{x}_m(k)$ and reconstructing signal $y_m(n)$ at each decoding step and ordering the values of w_i, $i=0,1,\cdots\cdots,M$.

Ⅳ. PERFORMANCE ANALYSIS OF OUR PROPOSED ALGORITHM

To analysis the performance of our communication system, the result of a BER performance is good way for it, but finding a BER formula could be a quite complicate issue for our proposed protocol. For a normal communication system, the BER is a monotonically decreasing function of variable SNR. Similarly, we can define the BER performance for detecting $x_m(n)$ from the m^{th} relay node as a function of variable SINR. Denote that the BER performance function is $f(\cdot)$ with variable SINR. In general, there is no closed form expression for $f(\cdot)$, but it can be easily obtained by the Monte Carlo method. Due to the SINR property for BER performance function, if the SINR of our proposed algorithm has little improvement compared with conventional algorithm, then our algorithm has efficiently improved its BER performance in all iteration.

At first, we only consider ordering $\mathbf{w}$. Then the m^{th} SINR can be expressed as

$$SINR_m = \frac{|\tilde{w}_m|^2}{|\tilde{w}_m|^2 + E[|\hat{\zeta}_m(k)|^2]} \tag{18}$$

Denote a w_{old} for the conventional algorithm without ordering the channel coefficients $\mathbf{w}$ and a w_{new} after ordering $\mathbf{w}$, also $w_{new} \geq w_{old}$ for detecting m^{th} transmitted signal. Obviously, the distortion $E[|\zeta_{m\_new}(k)|^2]$ with ordering $\mathbf{w}$ is smaller than $E[|\zeta_{m\_old}(k)|^2]$ and $SINR_{m-new} \geq SINR_{m-old}$. The BER performance could be better due to little improvement on SINR. Secondly, we discuss the SINR evolution by cancelling $\hat{x}_m(k)$ at both side of (9), and m^{th} SINR can be expressed as

$$SINR_m = \frac{|\tilde{w}_m|^2}{|\tilde{w}_m|^2 + \sum_{k\leq M}|a_i w_i|^2 E[(\tilde{x}_k(n)-\hat{\tilde{x}}_k(n))^2] + E[|\hat{\zeta}_m(k)|^2]} \tag{19}$$

Comparing (17), we only consider $\sum_{k\leq m-1}|a_i w_i|^2 E[(\tilde{x}_k(n)-\hat{\tilde{x}}_k(n))^2]$ part had influence on the $SINR_m$.

Obviously, combine with (11) and (13), we can easily compare

$$\sum_{k\leq M}|a_i w_i|^2 E[(\tilde{x}_k(n))^2] \geq \sum_{k\leq M}|a_i w_i|^2 E[(\tilde{x}_k(n)-\hat{\tilde{x}}_k(n))^2] \tag{20}$$

Repeating (13), we can track the SINR evolution for canceling each detected information symbol, $\sum_{k\leq M}|a_i w_i|^2 E[(\tilde{x}_k(n)-\hat{\tilde{x}}_k(n))^2]=0$ while perfectly decoding it. In this case, the interference can be totally cancelled and then the BER performance becomes better due to efficient improvement of SINR.

Ⅴ. NUMERICAL RESULTS

In this section, numerical results of the proposed protocol and its decoding algorithm are presented here. The channel is Rayleigh flat fading and also known as receiver. Let N be the number of information bits in a frame, M=2, 4, 6, 8 is the number of relay nodes and It denotes the number of iterations. QPSK signaling is always assumed. The delay τ_i at each relay is chosen randomly from 0 to

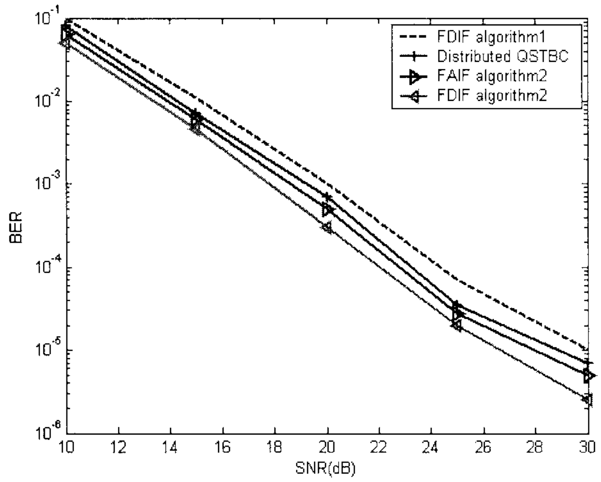

그림 2. 알고리즘에 따른 각 프로토콜의 비교

Fig. 2. Comparisons of different Protocols with different algorithm.

15 with uniform distribution. The number of iteration is set as 12.

In Fig. 2, the BER using our proposed protocol is compared with [1] using decode-interleave-forward protocol with 4 relay nodes. Also assume the synchronization of relay nodes when all relay nodes forward information bits to the destination, and the performances between these exits a trivial gap due to error-free in [1,3] at each relay node. The performance of our proposed algorithm used in FDIF protocol is also compared in Fig. 3 to the

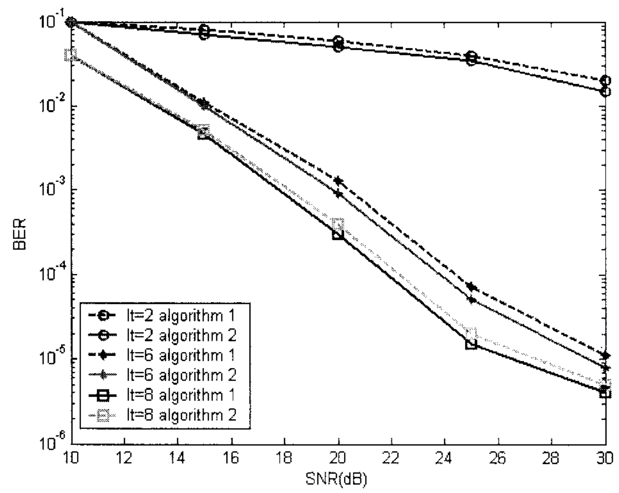

그림 3. 비동기 지연에서 반복 횟수에 따른 각 알고리즘의 비교

Fig. 3. Comparisons of different algorithms in asynchronous delay at each iteration.

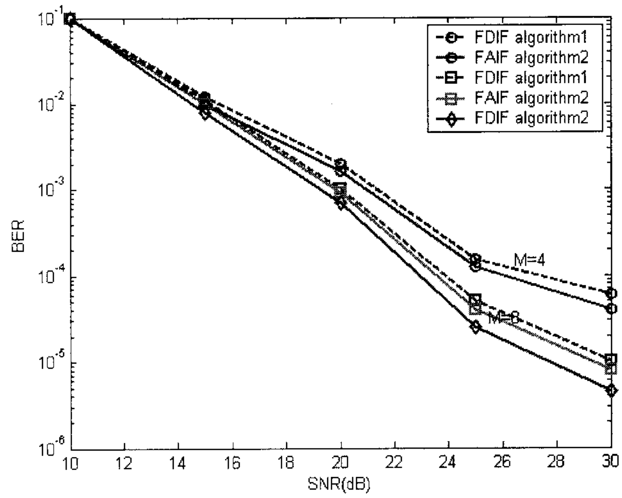

그림 4. 알고리즘에 따른 각 프로토콜과 비동기에서의 각 relay node 비교

Fig. 4. Comparison of different Protocols with its algorithm and different relay nodes in asynchronous case.

conventional algorithm using that FDIF in all iteration. The performance gap is going to gradually small until the last iteration exiting 1.5 dB difference compared to the first iteration exiting 4.5 dB and our proposed algorithm efficiently improves its performance in high SNR region 20-25 dB (where $BER \approx 0$) because of $E[(x_k(n) - \hat{x}_k(n))^2]$ near to 0. Similar to [1], we also compare the performance to employing QSTBC [2] and simulation result shows that in the lower SNR region 10 dB. These BER curve are almost same, but there exits a 2-3 dB gap while the SNR located in 20-25 dB region.

In Fig. 4, we consider asynchronous case in these relay network protocols with different relay nodes employing two different decoding algorithms. From this Fig. 4, it can be clearly observed BER in a given number of relay nodes and the BER gap in the high SNR region is becoming larger, which implies an increase in the diversity gain achieved by increasing the number of relay nodes. Comparing Fig. 4 to Fig. 2, in low SNR region, our proposed algorithm is also suitable for the FDIF protocol and efficiently achieves 2 dB improvement in the terms of BER performance. compared to Fig. 3.

Ⅵ. CONCLUSION

In this paper, we extend a decode-interleave-forward protocol into more general case of amplify-interleave-forward, due to its less processing burden on the relay. To combat its errors on relay, an efficient decoding algorithm is also presented in this literature consisting of an ordering operation and canceling operation. For this proposed algorithm, it can be also used in decode-interleave-forward protocol or some general cases with better performances compared to its conventional algorithm.

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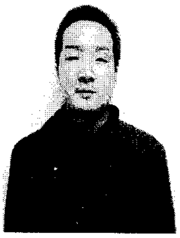

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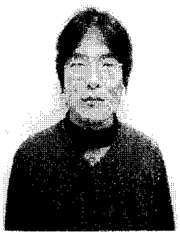

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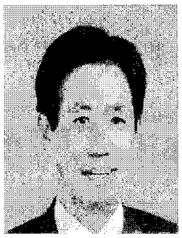

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