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Description	

ProRATE: Obtaining the Potential Throughput from the System View in Two-way Relay Networks

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***Abstract*—The wireless condition is time-varying and difficult to predict, especially in two-way relay networks (TWRN) with physical-layer network coding (PLNC) performed. This paper proposes a prompt and optimized rate adaptation algorithm (ProRATE) in this situation, which performs the real-time rate adaptation based on data-flows instead of point-to-point links. By using the signal dispersions and the history of the ratio of successfully decoded units, ProRATE adjusts the transmission rates of the terminal nodes into a matching state, thus increases the system throughput. Simulation results verify the effectiveness of ProRATE by showing that it brings outstanding end-to-end throughput gain over the conventional statistical-SNR-based rate adaptation. Moreover, as a byproduct, the jitter of BER of the whole system has been alleviated.**

I. Introduction

According to Cisco Visual Network Index (VNI) [1] reports, mobile data traffic doubles every year due to the burst in the number of terminal users, and also due to the smart phone they use, which enable data-consuming applications. This causes severe interferences and the shortage of assignable frequency resources [2]. Therefore, relays are introduced to relieve the tight situation. Here comes the scenario of Two-way Relay Network(TWRN) [3], [4]. That is, whenever two terminal users exchange data through an common intermediate relay, these three form a TWRN. According to [5], [6], Physical-layer Network Coding (PLNC) can help save half of the time-frequency resources in TWRN. However, the problem is how to do rate adaption in TWRN, since the conventional point-to-point rate adaption way is not practical in TWRN whose sub-links are influenced by each other. Moreover, a dynamic rate adaptation is a must because the statistical or mathematical estimation is not always correct in the time-varying wireless channels [7]. Or they may cause a huge amount of packet loss or an unexploited channel capacity, thus lead to a low system throughput.

Here comes our motivation. By analyzing the existing point-to-point rate adaptation, we can find out why they are efficient in the wireless environment. By borrowing and combining their core ideas, we aim to design a rate adaptation scheme for TWRN, which can leverage the system throughput. Thus, we propose ProRATE (Prompt and Optimized Rate Adaptation) in this paper. The main contributions of our work are listed as follows:

- ProRATE utilizes the reciprocal characteristic of TWRN whereby the rates of both sides can be adjusted to proper values, and thus increases the spectrum efficiency.
- ProRATE offers a new theoretical way to estimate channel condition from an overall perspective, and adapt rates on data-flows instead of on point-to-point links.
- ProRATE increases the end-to-end throughput by performing a prompt and optimized rate adaptation.
- ProRATE alleviates the jitter of the Bit Error Rate (BER) caused by the variation of the average SNR.

The rest of this paper is organized as follows. Section II goes over the related work. Section III describes the system model. Section IV presents the algorithm of ProRATE in details, together with some further discussions. Section V demonstrates the performance of ProRATE through simulation evaluations. Section VI summarizes this paper and suggests some future works.

II. Related Work

To our best knowledge, there are three chief approaches to perform the point-to-point rate adaptation:

a) Based on the frame error history: Wong et al. have proposed an algorithm named RRAA (Robust Rate Adaptation Algorithm) [8], in which the short-term loss ratio can opportunistically be the guide to the decisions on changing rate. In ProRATE, the control unit has been downsized to several code units, which can offer a much more prompt response to the variation of the wireless environment.
b) Based on the statistical SNR: To improve the performance against the jitter caused by the wireless environment, Zhang et al. have proposed an algorithm, which can achieve rate adaptation based on the SNR evaluated by RTS/CTS (Request To Send/Clear To Send) [9]. Through the track of SNR, the sender can calculate the curve between SNR and the sending rate, and then get an appropriate sending rate according to the current SNR value. In ProRATE, we evaluate the SNR of the whole transmission flows.
c) Based on the signal dispersion: Sen et al. have proposed a rate adaptation algorithm, named AccuRate [10], which authorizes the receiver node to estimate a suitable

transmission rate. The receiver node calculates the possible constellation by comparing the minimal Euclidean distance of the constellation nodes with the dispersion of the received signal. The receiver node then feeds the sender back with the result of the above measurement, which contains both the appropriate highest modulation order and its suitable code rate. This method enables the sender to jump to the best transmission rate by replaying the dispersion. In ProRATE, the core idea of dispersion can be utilized to measure whether a higher or a lower order modulation scheme is needed.

In the above three existing methods, the first two rely on mainly the statistical information, which may lose some efficiency when considering the swift variation of the wireless channel conditions. The third algorithm can respond to the variation of the wireless channel immediately.

Adaptive Modulation and Network Coding (AMNC) [11] was proposed as a rate adaptation against the TWRN background, which jointly considers modulations and network coding based on the Channel State Information (CSI). At the relay node, an optimized constellation with a mapping based on Closest-Neighbor Clustering Algorithm [12] is employed. The performance gain is, to some extent, obtained by the cost of computational complexity.

III. System Model

The aforementioned TWRN includes three nodes (data exchanger A, B and relay R), and four physical links (the unidirectional uplinks A-R and B-R, and the unidirectional downlinks R-A and R-B), as shown in Fig. 1. We assume that the links between A and R are independent with that between B and R, while the uplink and the downlink with the same end-nodes are reciprocal. We also assume that the direct links between A and B can be ignored because of the extremely low SNR. All links in this system are assumed to be Rayleigh-fading channels. The network model and the 2-stage date exchange protocol are given in Fig. 1.

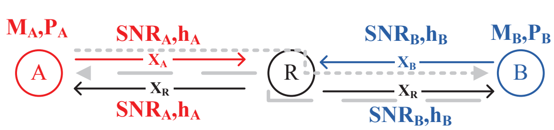

Fig. 1. The system model of ProRATE. The data flow from A to B is represented by the dotted arrow (gray) starting from A, while the one from B to A is by the dashed arrow (gray) starting from B.

Exchanging data with PLNC in our system can be divided into 3 parts: the *multiple access (MA) stage*, the *denoising treatment*, and the *broadcast (BC) stage*. In some literature such as [12], this protocol is also called a *2-stage protocol*.

1) The MA Stage: As shown in Fig. 1, the original information S_k is modulated with the scheme M_k, and transmitted at the power level P_k. Thus, the signals $x_k = \sqrt{P_k} M_k(S_k), (k \in \{A, B\})$ from the terminal nodes are sent simultaneously and overlapped at R as

$$y_R = h_A x_A + h_B x_B + n_R \tag{1}$$

In this equation, n_R is the additive noises which satisfy the complex circular symmetric Gaussian distribution with mean zero and variance N_0. This stage finishes with R getting the signal y_R.

2) The Denoising Process: Since denoise-and-forward (D-NF) is performed at R in this paper, R removes the noise part from the received signal, before forwarding the superimposed signal. According to [12], conducting maximum likelihood (ML) joint detection as

$$\begin{cases} (\hat{x}_A, \hat{x}_B) = \arg\min |y_R - (h_A x_A + h_B x_B)|^2, \\ \hat{y}_R = h_A \hat{x}_A + h_B \hat{x}_B \end{cases} \tag{2}$$

enables us to map the received signal y_R to the nearest candidate point $\hat{y}_R$ on the joint constellation map. The signal, x_R, transmitted by R in this stage is represented as $x_R = \hat{\alpha} \hat{y}_R$, where

$$\hat{\alpha} = \sqrt{\frac{P_R}{|h_A|^2 P_A + |h_B|^2 P_B}} \tag{3}$$

denotes the normalized coefficient extended from [13]. Similarly, P_R is the transmission power at relay node R. Now, the output signal x_R is ready for transmission after this stage.

3) The BC Stage: In this stage, R broadcasts the prepared signal x_R to both A and B. The received signals at A and B are expressed as

$$y_k = h_k x_R + n_k, \quad (k \in \{A, B\}) \tag{4}$$

Similarly, n_A and n_B are equivalent baseband additive noises which satisfy the complex circular symmetric Gaussian distribution with mean zero and variance N_0.

IV. ProRATE Algorithm

The core idea of ProRATE is to obtain a potential performance gain by dynamically changing the transmission rates in real-time after theoretically estimating the system from an overall perspective. For each terminal node in TWRN, there are two relative *transmission flows*, sending flow and receiving flow. The *sending flow* of a certain terminal node is a transmission flow in which this node acts as a sender while its counterpart works as a receiver; the *receiving flow* of a certain terminal node is a transmission flow in which this node works as a receiver while its counterpart works as a sender. Since the uplink from A(B) to R has the similar channel qualities as the downlink from R to A(B), the sending flow and the receiving flow of A(B) have some reciprocal characters. The details will be illustrated in the rest of this section.

A. Purpose Description

We define the *consecutive period* as the duration of performing ProRATE once. If the modulation scheme $M_{i,k}$ is employed during the ith consecutive period of ProRATE, the number of the points on the corresponding constellation map is $Q_{i,k}$, and the number of bits contained in this transmitted symbol is $\log_2(Q_{i,k})$, in which k represents A or B. We define the *code unit* as a basic unit in which the channel coding with the code rate (rate of channel coding) $R_{c,i,k}$ can be conducted. Therefore, the *end-to-end throughput* (the same as system throughput) is defined as

$$\begin{aligned} T = &\frac{1}{2t}\sum_{i=1}^{N}(\log_2(Q_{i,A}) \times R_{c,i,A} \times \hat{N}_{cu,i,A}) \\ &+\frac{1}{2t}\sum_{i=1}^{N}(\log_2(Q_{i,B}) \times R_{c,i,B} \times \hat{N}_{cu,i,B}) \end{aligned} \tag{5}$$

The parameter t above is the length of the observation period, and the parameter N represents the number of times ProRATE is conducted during the observation period. During a consecutive period of ProRATE, the number of the code units sent by a specific terminal node is presented by $N_{cu,i,k}$ and number of the successfully decoded code units is presented by $\hat{N}_{cu,i,k}$. Therefore, the *decoding ratio* can be written as

$$R_{d,i,k} = \frac{\hat{N}_{cu,i,k}}{N_{cu,i,k}}, \quad (k \in \{A, B\}) \tag{6}$$

We also define another measurement BER as its normal definition:

$$BER_{sys} = \frac{1}{2t}\sum_{i=1}^{N} BER_{i,A} + \frac{1}{2t}\sum_{i=1}^{N} BER_{i,B} \tag{7}$$

where $BER_{i,k} = \frac{Error\_bits\_number_{i,k}}{Frame\_size_{i,k}}, \quad (k \in \{A, B\})$.

B. The Prompt and Optimized Rate Adaptation

In this scenario, we assume that the relay node R is able to probe the coefficients h_A and h_B. We also assume that the dispersions of all the received signals can be measured immediately at the receiver nodes. To keep it simple, we firstly put the rate adaptation at the relay node R, which naturally possesses almost all information in this system. We mark all the parameters in the ith consecutive period with the same subscript i. We decide a new pair of transmission rates for A and B after every consecutive period.

The *transmission rate* of the terminal node is defined as

$$R_{t,i,k} = \log_2(Q_{i,k}) \times R_{c,i,k}, \qquad (k \in \{A, B\}) \tag{8}$$

Obviously, the transmission rate can be adjusted by changing the code rate $R_{c,i,k}$ or by adjusting the modulation scheme, which is embodied by $Q_{i,k}$. We dynamically split data stream into code units according to the currently suitable code rate $R_{c,i,k}$. The parameter $L_{cu,i,k}$ is the minimum number of symbols, whereby a code unit can be constructed. For instance, we assume $R_{c,i,k} = 1/2$ and the current modulation scheme at node k is QPSK, which means $Q_{i,k} = 4$. Since two bits will be contained in one symbol, the code rate can be reached by assigning one bit as data and one as redundancy. Therefore, the length of the basic code unit is the same as the length of a symbol, i.e. $L_{cu,i,k} = 1$. However, if $R_{c,i,k} = 3/4$, the code rate can be reached only by assigning 3 bits as data and 1 as redundancy. Therefore, $L_{cu,i,k} = 2$, which means at least two symbols have to be involved into one code unit. After an integral multiple of code units has been transmitted, the receiver will be able to decide the percentage of the successfully decoded code units. Since our algorithm is based on the status how many previous decoding operations are successful, it can only be conducted after transmitting $n \times L_{cu,i,k} (n = 1, 2, 3, ...)$ times. Because PLNC is employed in our algorithm, A and B should transmit their signals simultaneously. In this way, we can only perform our algorithm when $N_{cu,i}$ times of transmissions have been happened, where

$$N_{cu,i} = N_{cu,i,A} \times L_{cu,i,A} = N_{cu,i,B} \times L_{cu,i,B} \tag{9}$$

i.e. $N_{cu,i}$ is the lowest common multiple of $L_{cu,i,A}$ and $L_{cu,i,B}$. We define $N_{cu,i}$ as *the length of a consecutive period*. Obviously, $N_{cu,i}$ is dynamic. According to the 2-stage DNF, every 2 time slots, one pair of signals (from A and B, respectively) can be exchanged. Therefore, the length of a consecutive period equals to $(2 \times N_{cu,i})$ time slots.

After each consecutive period, A and B calculate decoding ratio $R_{d,i,k}$ in (6). Then, A and B send the ratio values to R. If the ratio value of the receiver is larger than a specific threshold R_{th}, we define that the receiver *has a good receiving flow*, wherefore rate-increasing is encouraged. Otherwise, rate-decreasing is recommended. The situations could be classified into 3 cases, which are listed as follows.

1) Both A and B have good receiving flows: This situation indicates that higher order transmission rates can be supported at both sides. Therefore, R will first check whether A and B can increase the transmission rates, $R_{t,i+1,A}$ and $R_{t,i+1,B}$, for the next consecutive period by only increasing their code rates, $R_{c,i+1,A}$ and $R_{c,i+1,B}$. If so, R will calculate $L_{cu,i+1,k}$ and a new consecutive period $N_{cu,i+1}$, and the corresponding $N_{cu,i+1,k}$ $(k \in \{A, B\})$ via (9).

However, if the code rate $R_{c,i,k}$ is already the highest one, R needs to do one more step to decide whether to adjust the node k to a higher order modulation or to give up this operation of rate-increasing. Here we introduce *dispersion* to depict the quality of the channels. We introduce the concept of dispersion from [10] and redefine it as $\overrightarrow{d_i} = \overrightarrow{r_i} - \overrightarrow{x_i}'$, where $\overrightarrow{r_i}$ represents a received vector on the constellation map while the $\overrightarrow{x_i}'$ is the nearest candidate vector. We illustrate this concept in Fig. 2.

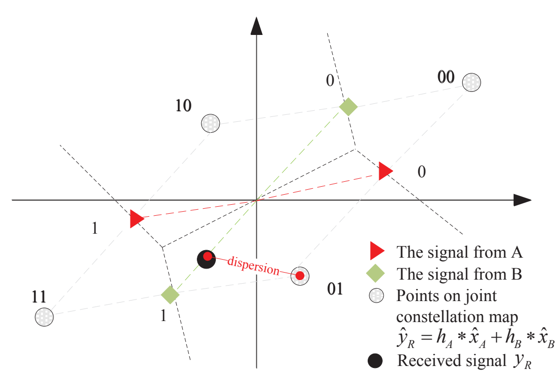

Fig. 2. If the constellation used at A and B are both BPSK (Binary Phase-Shift Keying), the joint constellation map will be like the structure of the gray points. The numbers marked beside the gray points are the combination of what A and B send, of which the former number is from A and the latter from B. If the received signal is presented with the black point, the distance between the received one and the correct point is defined as "dispersion".

To each data exchange option, two data flows are involved. The first one is from A to R to B, while the second is from B to R to A. Both of them are composed of two links. One link is between R and A, and the other link is between R and B. Therefore, to finish a transmission from one terminal node to another, there are two point-to-point transmissions, wherein one dispersion happens at R and another at the terminal receiver. The ratio of dispersion to the minimum Euclidian distance of a specific constellation map can express how big the channel influence is. The closer the value of this ratio is to 1/2, the more likely the symbol will be demodulated to a wrong point. Therefore, for the jth symbol transmission during the ith consecutive period, we define the parameters $\hat{\eta}_{i,j} = \left| \frac{2 \times d_{i,j}}{Ed_{\min,i,j}} \right|$ and $\hat{\eta}_{i,j,k} = \left| \frac{2 \times d_{i,j,k}}{Ed_{\min,i,j,k}} \right|$ to describe the condition at R and the terminal receiver (A or B), respectively. Here, $Ed_{\min,i,j}$ and $Ed_{\min,i,j,k}$ are both Euclidean distances for their maps, respectively. $\hat{\eta}_{i,j}$ is calculated by the relay node R, while the $\hat{\eta}_{i,j,k}$ by the terminal receiver. The second step done by the terminal receiver is to calculate $\eta_{i,j,k} = \max\{\hat{\eta}_{i,j}, \hat{\eta}_{i,j,k}\}$, which can help describe the condition of the weaker transmission channel. After $N_{cu,i}$ times of transmissions, the performance of the terminal receiver can be evaluated via

$$\eta_{i,k} = \frac{\sum_{j=1}^{N_{cu,i}} (1 - \eta_{i,j,k})}{N_{cu,i}} \tag{10}$$

This parameter can be used to describe the capability of increasing the transmission modulation order. For example, in Fig. 2, the dispersion $d_{i,j}$ is marked while the $Ed_{\min,i,j}$ is the distance between '01' and '10'. The parameter $\eta_{i,j}$, to some extent, describes the quality of the two uplinks in the MA stage. The closer $\eta_{i,k}$ is to '1', the more severe the channel influence, and so as the parameter $\eta_{i,j,k}$. Under such a circumstance, changing to a higher order modulation scheme may lead to the reduction of the Euclidean distance, hence the received signal is more likely to be demodulated to a wrong candidate. Therefore, we adjust the sender node to a higher-order modulation scheme at the probability of $(1 - \eta_{i,k})$. We need to clarify that the parameter $\eta_{i,A}$ is used to evaluate whether B can change to a higher order modulation scheme while $\eta_{i,B}$ is used for A.

2) Either A or B has a good receiving flow: Without loss of generality, we assume that B is suitable for increasing rate while A is not. Therefore, B increases the code rate $R_{c,i+1,B}$ and gives up rate-increasing when $R_{c,i+1,B}$ is the highest code rate. B should never raise its modulation order since A fails to decode lots of code units and the option of raising modulation order can at the same time increase the density of the joint constellation at R. On the other hand, A decreases the rate. Here A will first check the parameter $\eta_{i,B}$ and directly jump to a lower order modulation at the probability of $(1 - \eta_{i,B})$, or only reduce the code rate ($R_{c,i,A}$) at the probability of $\eta_{i,B}$. Therefore, B is likely to recover the data in the next period either because of the sparser constellation or of the less information carried.

TABLE I
MODULATION MODE & CODE RATE VS. SNR [14]

Modulation Mode & Code Rate	**Received SNR** (dB)
BPSK 1/2	3
QPSK 1/2	6
QPSK 3/4	8.5
16QAM 1/2	11.5
16QAM 3/4	15
64QAM 2/3	19
64QAM 3/4	21

3) Both A and B have bad receiving flows: In this case, both A and B decrease their code rates ($R_{c,i+1,A}$ and $R_{c,i+1,B}$) for the next period. Similarly, they will directly jump to a lower order modulation scheme at the probability of $(1 - \eta_{i,k})$; and at the probability of $\eta_{i,k}$, they will only reduce the code rate.

C. Data flow Equivalence and Initial Rate

As mentioned before, we perform rate adaptation on data flow instead of point-to-point link. The way we estimate a data flow is by finding an equivalent point-to-point link model with the same channel condition. From a statistic view, the link determines the modulation scheme, together with its code rate, by the quality of the wireless environment, which is mostly embodied by SNR. The relation is shown in Table I [14]. Therefore, the first step is to find the equivalent link.

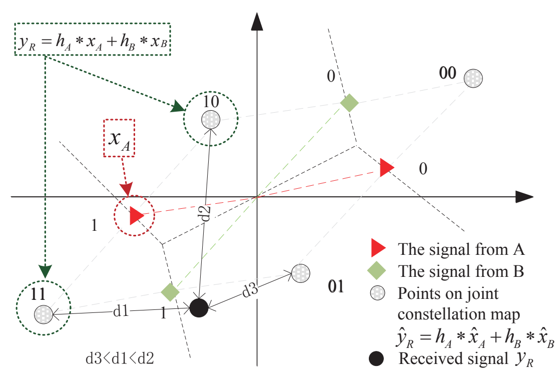

Fig. 3. A sends x_A = '1', which is the red triangle marked by a red dotted circle. The overlapped signals that A expects to receive are the gray points marked by the green dotted circles. The received signal is presented with the black point. Even when the distances $d1$, $d2$, $d3$ satisfy $d3 < d1 < d2$, the receiver A will demodulate the signal to '11'.

Since the signals are overlapped naturally at R, when demodulating the received signals, the terminal nodes can first downsize the scale of the joint map by filtering the points with their own information. Usually, the amount of the points on the joint map can be downsized to the quantity of the constellation points of the sender node (e.g. Fig. 3), i.e.

$$\begin{aligned} \tilde{y}_A &= \hat{\alpha} h_A h_B \hat{x}_B + n_A \\ \tilde{y}_B &= \hat{\alpha} h_B h_A \hat{x}_A + n_B \end{aligned} \tag{11}$$

Therefore, it is reasonable to design an SNR according to (11).

$$\begin{aligned} \mathrm{SNR_{AB}} &= \frac{\hat{\alpha}^2 |\mathrm{h_A}|^2 |\mathrm{h_B}|^2 \mathrm{P_A}}{\mathrm{N_0}} \\ \mathrm{SNR_{BA}} &= \frac{\hat{\alpha}^2 |\mathrm{h_A}|^2 |\mathrm{h_B}|^2 \mathrm{P_B}}{\mathrm{N_0}} \end{aligned} \tag{12}$$

The values in (12) can be used as indices to describe the performances of the overall transmission data flows.

When the conversation starts, R probes the channel conditions, h_A and h_B, and broadcasts the evaluation coefficient $\beta = \widehat{\alpha}^2 |h_A|^2 |h_B|^2$ to A and B. Thus, A and B can calculate (12) with β. Moreover, initial transmission rates of each flow can be determinated by checking Table I with SNRs in (12).

D. Rate Adaptation at the Terminal Nodes

In the previous narration, the rate adaptation is performed at the relay node R because it is easier to estimate the whole system at the "joint" point R. However, some extra transmissions are inevitable if rate adaptation is performed at R (e.g. the decoding rates at terminal nodes). Therefore, we modify the workflow of ProRATE and perform it at terminal nodes.

After the relay node calculate the parameter $\eta_{i,k}$ in (10), it will send the value to the terminal nodes. At the same time, R will tell A and B which flow is better, the transmission flow A-to-R-to-B or the one B-to-R-to-A. This measurement could be embodied by the $\mathrm{SNR_{AB}}$ and the $\mathrm{SNR_{BA}}$ in (12). Here we involve some prediction. If the receiver of a better transmission flow cannot recover the original information, neither can the receiver of the transmission flow with lower SNR. To illustrate this, we assume that in one specific transmission period, $\mathrm{SNR_{BA}}$ is greater than $\mathrm{SNR_{AB}}$, which means A is the receiver of a better transmission flow.

a) If $R_{d,i,A} < R_{th}$, it is most likely that $R_{d,i,B} < R_{th}$, as well. Therefore, the wise option of A is to decrease the transmission rate $R_{t,i,A}$.
b) If $R_{d,i,A} \geq R_{th}$, then $\eta_{i,A}$ will be checked. If $\eta_{i,A}$ is quite close to '1', the noise in the MA stage is loud, which indicates that the noise in the BC stage may also be loud. In this way, B may not be able to recover the information. On the other hand, if $\eta_{i,A}$ is close to '0', B is likely to recover the information. Thereby, we design the option like this: A increases the transmission rate $R_{t,i,A}$ at the probability of $(1 - \eta_{i,A})$, and keeps the current rate at the probability of $\eta_{i,A}$.
c) If $R_{d,i,B} < R_{th}$, opposite to previous item, B decreases the transmission rate $R_{t,i,B}$ at the probability of $\eta_{i,B}$, and keeps the current rate at the probability of $(1-\eta_{i,B})$.
d) If $R_{d,i,B} \geq R_{th}$, it is most likely that A can also recover the data. Hence, B increases the transmission rate $R_{t,i,B}$.

However, performing rate adaptation at the terminal nodes makes the system more susceptible to errors, because the predictions will not always be correct.

V. Simulation Results and Analysis

We evaluated the performance of ProRATE by MATLAB simulations. We built the structure according to the system model, with two terminal nodes, A and B, and one relay node, R, in the middle. We used DNF at the relay node. Throughout the simulations, the distance between the terminal nodes, A and B, was fixed. We conducted the experiments with Rayleigh fading i.i.d. channels to observe the performance gain on the end-to-end throughput defined in (5). We also evaluated the system BER in (7). During the experiments, 10 simulation scenarios with different random seeds were averaged. The frame size was set to 1024. The mean value of SNR traversed from about 0 dB to 30 dB, which was controlled by changing the transmission power according to the channel fading. The mean consecutive period for ProRATE in the experiments was about 2000 time slots.

In the experiments, when the sender transmitted its symbols, there were four possible modulation schemes: BPSK (Binary Phase-Shift Keying), QPSK (Quadrature Phase-Shift Keying), 16-QAM (Quadrature Amplitude Modulation) and 64-QAM. The code rates supported in these experiments were the typical ones: 1/2, 2/3, 3/4, 5/6, 7/8, etc. We compared ProRATE with the conventional rate adaptation method over the statistical SNR, whereby the relay probes the channels condition periodically and changes the transmission rates according to Table I.

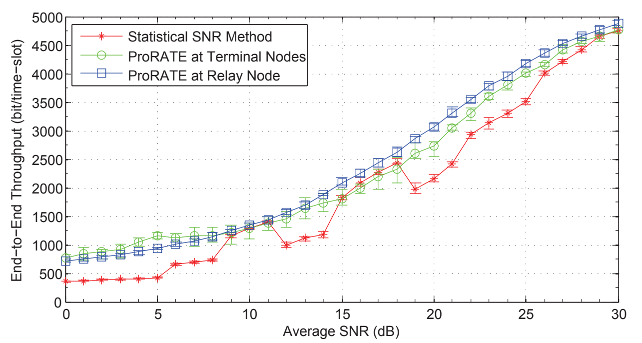

Fig. 4. End-to-end throughput as a function of average SNR in Rayleigh fading channels.

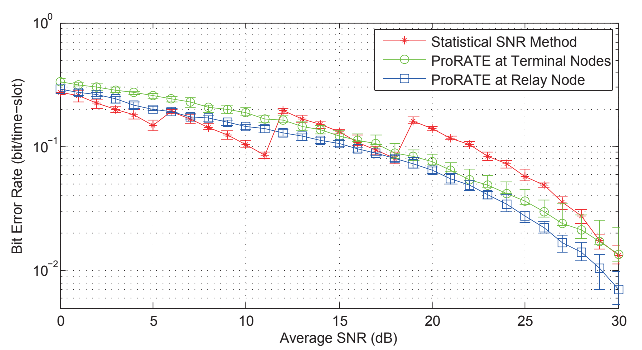

Fig. 5. The BER of different value of average SNR in Rayleigh fading channels.

In Fig. 4 the end-to-end throughput, which is defined in (5), achieved by ProRATE is expressed in the square-dotted line and the circle-dotted line. The figure compares ProRATE with the conventional rate adaptation over the statistical SNR, which is drawn with the star-dotted line. Similarly, Fig. 5 shows the BER of the whole system.

We first pay attention to Fig. 5. The star-dotted line jitters a lot, because the modulation scheme is changing from BPSK to 64-QAM. Each item in Table I directs the time when the senders change the modulation scheme and the coherent code

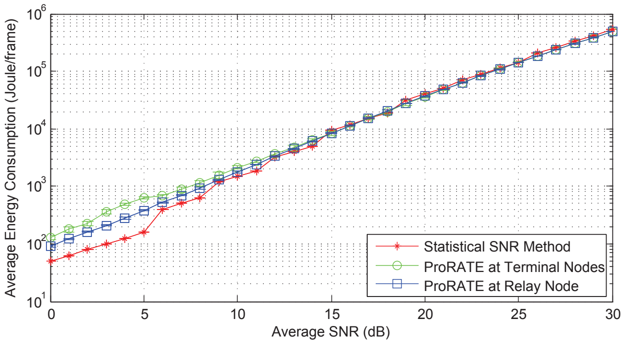

Fig. 6. The average energy consumed in transmission.

rate. With ProRATE, the jitter has been smoothed. There are two reasons here: 1) we calculate the theoretical SNR of the whole transmission flow (12), so that the SNR values are more plausible to estimate the real condition; 2) with dynamic rate adaptation, the system can respond swiftly to the channel variation.

As for Fig. 4, on the whole, the end-to-end throughput grows gradually when the average SNR is getting higher and usually the end-to-end throughput of performing ProRATE is higher than the conventional method. We can find a lot of jumps on the star-dotted line for the conventional method. These jumps happen at when A(B) changes its modulation scheme or its code rate. Combined with Fig. 5, we can find out that at every jump point where the end-to-end throughput declines, the BER line also mutates from a lower level to a higher one. This is because the modulation scheme used by A(B) changes from a lower order to a relatively higher one as aforementioned, which brings out larger bit-error probability but more bits contained in one transmission. The throughput ProRATE obtains in Fig. 4 is achieved by holding BER as high as that the code unit is just able to be decoded with its code rate. This is how ProRATE exploits the potential channel capability.

However, in some range of SNR (e.g., 13dB-15dB), the throughput gain is also outstanding, but the reason is different. In this kind of situation, the conventional rate adaptation method has a higher BER, which waste the bandwidth by sending wrong bits. However, in some regions, the conventional method has a better throughput than doing ProRATE at the terminal nodes (e.g. when SNR $\in$ (15,18) dB). This is because performing ProRATE at terminal nodes causes higher BER in these region. This is just a single time event, but can also reveals some truth: if we do ProRATE at the terminal nodes, more likely, we can expect a higher error rate and sometimes also lower throughput, since predictions are not the facts.

From the energy perspective (Fig. 6), the statistical SNR method consumes less energy when SNR is low. We can tell the reason when considering its performance in Fig. 5. With lower BER, lower order modulation scheme will be chosen, and less transmission energy will be wasted. When the SNR is high, however, the energy consumption is almost the same.

As a summary, to perform ProRATE at the relay node can raise the end-to-end throughput, and at the same time, can alleviate the jitter of BER, and it is normally better than performing ProRATE at the terminal nodes, especially when SNR is relatively high.

VI. Conclusion

In this paper, we have proposed ProRATE for TWRN. This algorithm utilizes the interaction of the four links in TWRN and performs rate adaptation from the data-flow perspective. ProRATE can response swiftly to the channel variation by analyzing the decoding ratio history and the signal dispersions. We have found out the initial rates for the terminal nodes from the perspective of data-flow. We have enhanced ProRATE by performing rate adaptation at the terminal nodes instead of at the relay node. The simulation results have verified that ProRATE is reasonable and efficient in TWRN. The performance gain brought by ProRATE is outstanding when SNR is less than 30 dB over the traditional SNR-based rate adaptation. Moreover, the energy consuming is almost the same and the BER is smoothed at the same time. One of our future work is to update ProRATE into some more general algorithm which fits for all kinds of data transmission networks with PLNC.

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