A Study on a New P-FTN Method for High Throughput Wireless Communication

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Abstract **—**To increase throughput efficiency, LDPC (Low Density Parity Codes) codes with various high coding rates are employed in DVB-S3 system. However, the more the coding rate increases, the more performance decreases sharply. This paper proposed a new P-FTN (Punctured-FTN) method which combine a punctured LDPC with FTN (Faster Than Nyquist) method to increase performance while maintain throughput as compared to high coding rates of LDPC codes. Furthermore, this paper present efficient receiver structure suitable for P-FTN with LDPC decoder. As a result of computer simulation, P-FTN method has better performance than the high coding rate LDPC method.

***Index Terms*—**LDPC, high coding rate LDPC, FTN, punctured, P-FTN

I. Introduction

LDPC(Low Density Parity Check) [1]-[4] codes employed on satellite communications have received tremendous attention in the coding community because of their excellent error correction capability and near-capacity performance. Some randomly constructed LDPC codes, measured in BER(Bit Error Rate), come very close to the Shannon's limit for the AWGN(Additive White Gaussian Noise) channel with iterative decoding and very long block. Therefore many methods for increase of throughput are being researched while the bandwidth is limited. However, it is very difficult to improve both throughput and performance at the same time, that's why the two are in a trade-off relationship. Therefore, it is the most important to develop methods which can maintain the performance to the maximum, while increasing the throughput. As adjusting the parity check matrix in LDPC codes, various coding rates are generated in order to satisfy required bandwidth of satellite communications. However, the more the coding rate increases, the more the performance decreases. To overcome the drawbacks of high coding rates, some researchers study puncture LDPC codes [5], [6] which can improve throughput as delete coding symbols of mother coding rate according to appropriate rules. The others, FTN (Fast Than Nyquist) method [7]-[10] which transmits faster than Nyquist rate is considered. This paper propose a punctured-FTN method which combine punctured LDPC codes and FTN method to improve throughput efficiency and performance simultaneously. At the receiver side, there are two methods to improve performance of FTN signals. First, it is the SIC (Successive Interference Canceller) method that removes the FTN interference by obtaining it from subtracting the LDPC decoded bit stream at the receiving end. Decoded signals through remapping them using the FTN mapper and again subtracting interference from the received signal [11]-[13]. It is hard to get the accurate amount of interference at the stage of remapping because the first method performs the LDPC decoding using signals that are distorted from high interference. Second, when a signal has much interference, BCJR (Bahl, Cocke, Jelinek and Raviv) equalization method [14]-[17], which removes the interference firstly and then equalized symbols are passes through the LDPC decoder demonstrates a better performance. However, it has a disadvantage that, with iterative decoding using the conventional BCJR equalization scheme, its performance does not improve significantly due to inefficient iteration, applying the same extrinsic input that is applied as the input value for equalization to the LDPC decoding signal. In this paper, to improve the efficiency of the iterative decoding scheme of conventional FTN decoding, we propose a scheme, which separates the LDPC decoding signals and applies them as extrinsic input, and investigate the performance of the iterative decoding scheme proposed in this paper by comparing it with the conventional scheme in a simulation according to the increase of the throughput. Combined with proposed decoder structure, we confirmed that the performance of the proposed method is better than high coding rate methods through computer simulations.

II. High Throughput Algorithms

A. *High Coding Rate*

As described in [3], the next generation satellite broadcasting standard such as DVB-S3(Digital Video Broadcasting-Satellite Third Generation) requires the large block sizes ($\boldsymbol{N}$ = 64800, 16200) and numerous iterations. There are ten code rates ($\boldsymbol{R}$ =1/4, 1/3, 1/2, 3/5, 2/3, 3/4, 4/5, 8/9, 9/10) and four modulation methods (QPSK, 8-PSK, 16-APSK, 32-APSK) are applied to increase bandwidth efficiency. High coding rate LDPC

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method is a technique that can generate various coding rate in the process of existing LDPC encoding. Design the parity check matrix used for LDPC encoding so that it can have various coding rates. For a given high code rate, these parity check matrices become increasingly as the length of a code-word, hence the number of columns and rows of parity checks increases. The more the coding rate increases, the more the performance decreases. As the parity check matrix is different according to various coding rates, structure of memory becomes complicated.

B. *Punctured Method*

Punctured method is to improve throughput as delete coding symbols of mother coding rate according to appropriate rules. It achieves a high coding rate without increasing the additional complexity. In such an approach, the transmitter systematically punctures bits in a coded block, and the locations of punctured symbols are known to receiver. Therefore it is important to choose the priority order to find the puncturing position. Priority is determined by the process of bit node and check node of LDPC decoding processing [5].

The procedures of puncturing are as follows.

Step 1. The number of puncturing bits, $N_p\left(=N\times\rho\right)$ is determined according to the desired throughput efficiency. N is length of encoded bits and ρ means the punctured rates.

Step 2. Set the initial counts of each m check node, $C_m\left(m=1,2,\cdots,M\right)$, are initialized to the number of degree of check nodes.

Step 3. Selected bit node connected to check nodes with maximum value of C_m .

Step 4. Puncturing the selected bit node

Step 5. Subtract N_p and C_m by 1.

Step 6. Iterate step 1 to step 4 until the $N_p=0$.

C. *Faster Than Nyquist Method*

FTN signaling is a technique of transmitting information at a rate higher than Nyquist limit. And this method has ISI (Inter-Symbol Interference) necessarily occurs.

Transmission signal with ISI can be given by (1).

$$s\left(t\right)=\sqrt{E_s}\sum_n a_n h\left(t-n\tau T\right),\quad \tau<1 \tag{1}$$

where a_n are encoded bit stream, E_s is the average symbol energy, and $h\left(t\right)$ is a unit-energy baseband pulse, which for this paper we will assume is orthogonal to shifts by T , τ is interference time. Interference ratio τ' is given by

$$\tau'\left(\%\right)=100\times\left(1-\tau\right) \tag{2}$$

The Fig. 1. show that there is no ISI generated as the transmission is run at the Nyquist rate when $\tau=1$. However, when $\tau=0.8$, the adjacent symbols affect each other due to FTN, and, as a result at each decision point of data, and so we know that there is a change in the waveform due to the interference. Although the signal's waveform gets distorted due to ISI, if this issue is overcome, it can be see that the transmission rate improves by 20% at the same time.

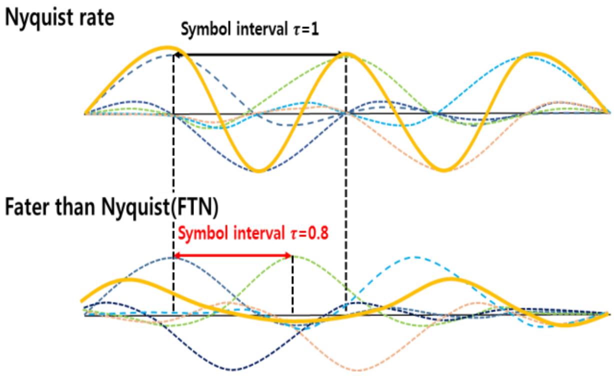


Fig. 1. FTN signal

III. Proposed P-FTN Method and Its Decoding Algorithm

We propose a P-FTN method combines punctured method and FTN method. In the existing FTN method, in order to obtain a desired throughput, interference rate according to the throughput is necessary. However, the proposed method punctures using the punctured steps as shown in previous Session II. Therefore, it is possible to obtain the throughput with a lower interference rate by the FTN processing. The transmitter block diagram of the proposed scheme is as shown in Fig. 2.

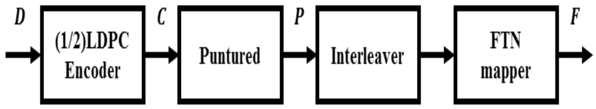


Fig. 2. Transmitter structure of P-FTN Method

Suppose that the binary bits is d_k delivered by source as follows:

$$D=\left\{d_1,d_2,\cdots,d_K\right\} \tag{3}$$

where K is length of source bits respectively. First, D is encoded by the $\left(N,K\right)$ LDPC encoder.

$$C=\left\{c_1,c_2,...,c_N\right\} \tag{4}$$

where C means encoded data by LDPC encoder. Encoded data of C is passed through punctured block. P means the output data of the punctured block according to punctured rate of ρ .

$$P=\left\{c_1,c_2,...,c_L\right\} \tag{5}$$

$L\left(=N\times\left(1-\rho\right)\right)$ is the length of the output data by punctured block.

After puncturing and interleaving, $\boldsymbol{X}$ symbols are generated according to interference rate $\boldsymbol{\tau'}$ of FTN processing, which is given by

$$\boldsymbol{X} = \{x_1, x_2, ..., x_F\} \tag{6}$$

$\boldsymbol{F}\left(= \boldsymbol{L} \times \left(1 - \frac{\tau'}{100}\right)\right)$ means the length of FTN output symbols. Total coding rate of P-FTN method cab be expressed as

$$\boldsymbol{R_T} = \frac{\boldsymbol{T_D}}{\boldsymbol{T_F}} \tag{7}$$

where $\boldsymbol{T_D}$ and $\boldsymbol{T_F}$ means the duration of the input data and FTN signals respectively.

Transmitted signal x_j may be written in the form:

$$x_j(t) = \sqrt{E_s}\sum_n c_j(n)h(t - n\tau T),\ \tau < 1\ \ (j = 1, 2, \cdots, N) \tag{8}$$

In FTN transmission, conventional method which used only LDPC codes can't guarantee performance requirement suggested in standard. Many schemes are announced in order to combat to ISI, among these schemes, SIC and BCJR equalizations are recommended in DVB-S3 standard.

However, these two schemes also can't guarantee performance requirement as increasing interference ratio. Therefore we propose iterative decoding method using LLR (Log Likelihood Ratio) symbol separation method to increase bit error performance.

A. Decoder Type-I: SIC Model

Fig. 3. shows iterative receiver structure based on SIC method.

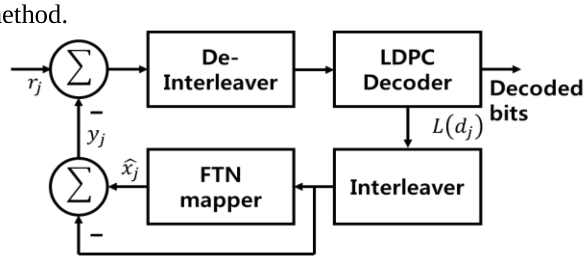


Fig. 3. Structure of iterative decoder based on SIC model

The received signal r_j can be expressed as (9).

$$r_j = x_j + \eta_j \tag{9}$$

where η_j means AWGN at time j.

Received signals, r_j are fed to LDPC decoder after de-interleaving. In LDPC decoding process, LLR can be obtained as shown in (10) [18].

$$L(d_j) = \sum v_{m,n} \tag{10}$$

The $v_{m,n}$ denote LLR of bit node unit in the direction of $\boldsymbol{m}$ check nodes to $\boldsymbol{n}$ bit nodes. LLR for d_j, $L(d_j)$ has the sign of plus or minus values according to d_j is "1" or "0".

$L(d_j)$ are calculated in LDPC decoder include interference of τ. It is interleaved and remapped in FTN mapper. As shown in (1), take into account acceleration factor τ, quantities may be written in the form:

$$\hat{x}_j = \sqrt{E_s}\sum_n L(d_j)h(t - n\tau T),\ \tau < 1\ . \tag{11}$$

The symbol $\hat{x}_j$ at the output of the mapper represents the amount of information the soft symbol $L(d_j)$ and the interference due to FTN signal. Therefore, the soft output $L(d_j)$ when subtracted from its corresponding output leaves behind the total interference y_j and it can be expressed as (12).

$$y_j = \sqrt{E_s}\sum_n h(t - n\tau T),\ \tau < 1 \tag{12}$$

Once an estimate of the interference is calculated, it can be readily cancelled out from the received symbols to give

$$\hat{r}_j = r_j - y_j \tag{13}$$

where $\hat{r}_j$ represents the interference cancelled symbols and y_j the estimate of the interference. And in the subsequent iterations, the interference cancelled symbols is decoded in LDPC decoder. However, if the interference ratio τ exceeds a certain level, distorted signals are input to the LDPC decoder; therefore, the amount of interference obtained from remapping these signals cannot be as accurate values. For these problems, we need another method that improves the decoding performance at a high value of τ.

B. Decoder Type-II: BCJR Equalization Model

Equalization, which is an effective method to deal with ISI, is necessary and is the most important part in the FTN systems. Therefore, a lot of efforts have been spent on equalization and advanced detection algorithms at the receiver side. At the receiver side, an LLR computer is needed to convert the equalizer output, assumed to follow Gaussian distribution, into extrinsic LLRs regarding the code bits, by using a priori LLRs from the previous decoding iteration. This updated set of soft information about the code bits is then deinterleaved and provided as a priori LLRs, for the next decoding iteration.

The computation of the LLRs pertaining to the equalizer uses the FTN rate and roll-off values in reconstructing ISI in each carrier. Generally, the decoding method of trellis type as shown in Fig. 4. is BCJR algorithm with soft decision value [16]. BCJR algorithm is a well-known maximum a posteriori probability decoding algorithm which has been proposed earlier for point to point communication applications.

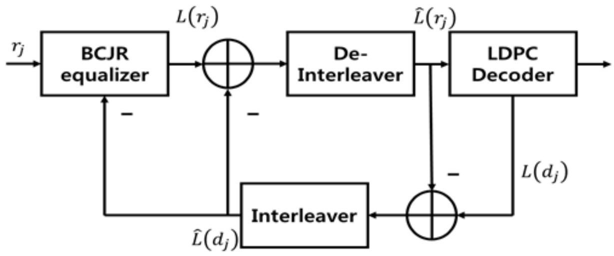


Fig. 4. Structure of BCJR equalization model

The value of $\hat{L}\left(r_j\right)$ after interleaving is computed $L\left(r_j\right)-\hat{L}\left(d_j\right)$, then input LDPC decoder. The estimated extrinsic value of $L\left(d_j\right)$ at decoder output is given by

$$L\left(d_j\right)=\log\frac{p\left(x=+1\right)}{p\left(x=-1\right)} \tag{14}$$

The extrinsic value $L\left(d_j\right)$ of which calculates the post probability is error correction terms. The re-interleaving of computed value as $L\left(d_j\right)-\hat{L}\left(r_j\right)$ is input to BCJR equalizer, then $\hat{L}\left(d_j\right)$ is updated in order to compensate for the errors.

At the receiver, we resort to powerful turbo equalization algorithms that iteratively exchange probabilistic information between inner decoder and outer decoder, thereby reducing the error rates significantly. In turbo equalization, two decoders are concatenated in the serial fashion. The inner decoder is BCJR decoder to cancel ISI induced from FTN signaling, and outer codes are LDPC decoder. The symbols of outer decoder are then subtracted from the input and interleaved. The interleaved symbols are cancelled a posteriori from the proceeding received symbol. Interleaving helps receiver convergence.

C. Decoder Type-III: Proposed P-FTN Decoder Model

In order to efficiently eliminate interference by FTN, symbol bit separation block is added to the receiver, which will enable performance has improved and whole iteration. Fig. 5. shows structure of the proposed BCJR equalization with bit separation. Furthermore, Fig. 5. shows the transceiver model of P-FTN method combines punctured method and FTN method.

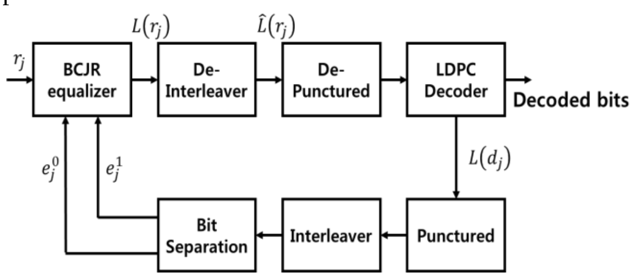


Fig. 5. Structure of the proposed model

The forward states metric is $\alpha_j^{s^j}$, backward states metric is $\beta_j^{s^{j-1}}$, and branch metric is $\gamma_j^{s^{j-1},s^j}$ at time j and s^j state. It is as follows:

$$\alpha_j^{s^j}=\max{}^*\left[\alpha_{j-1}^{s^j}+\gamma_j^{s^{j-1},s^j}\right] \tag{15}$$

$$\beta_j^{s^{j-1}}=\max{}^*\left[\beta_j^{s^j}+\gamma_j^{s^{j-1},s^j}\right] \tag{16}$$

LLRs can be calculated as (17).

$$\begin{aligned}L\left(r_j\right)=&\max{}^*\left[\alpha_{j-1}^{s^{j-1}}+\gamma_j^{s^{j-1},s^j}+\beta_j^{s^j}\right]\left(s^{j-1},s^j\right):d_j=0\\&-\max{}^*\left[\alpha_{j-1}^{s^{j-1}}+\gamma_j^{s^{j-1},s^j}+\beta_j^{s^j}\right]\left(s^{j-1},s^j\right):d_j=1\end{aligned} \tag{17}$$

$L\left(d_j\right)$ is LLRs value calculated in (10) and employed to extrinsic value of LDPC decoder. It is updated to BM by (18).

$$\hat{\gamma}_j^{s^{j-1},s^j}=\gamma_j^{s^{j-1},s^j}+L\left(d_j\right) \tag{18}$$

The BM $\hat{\gamma}_j^{s^{j-1},s^j}$ is applied to calculate LLRs as (19).

$$\begin{aligned}L\left(r_j\right)=&\max{}^*\left[\alpha_{j-1}^{s^{j-1}}+\hat{\gamma}_j^{s^{j-1},s^j}+\beta_j^{s^j}\right]\left(s^{j-1},s^j\right):d_j=0\\&-\max{}^*\left[\alpha_{j-1}^{s^{j-1}}+\hat{\gamma}_j^{s^{j-1},s^j}+\beta_j^{s^j}\right]\left(s^{j-1},s^j\right):d_j=1\end{aligned} \tag{19}$$

The extrinsic value $L\left(d_j\right)$ is removed to LLRs $L\left(r_j\right)$, which was shown in (20).

$$\hat{L}\left(r_j\right)=L\left(r_j\right)-L\left(d_j\right) \tag{20}$$

However, with this type of method, $L\left(d_j\right)$ indicates the probability value of the input bit d_j; because it shows a probability of plus value when d_j is "1" and probability of minus value when d_j is "0", updating LLR values of decoded signals without distinguishing between the cases where the state S_j is "0" and "1", respectively, to all BM values is not efficient in making significant differences among BM values from iterative decoding and thus does not improve the performance even with iteration [16]. The iterative decoding scheme proposed by this paper, different from the present method in which the LLR value $L\left(d_j\right)$ of LDPC decoder is added to the BCJR encoder's BM value, uses $L\left(d_j\right)$ as external input value by dividing the bit according to "0" or "1."

If we define the probability for "0" as e_j^0 and for "1" as e_j^1, when $L\left(d_j\right)$ as external input value is smaller than 0, e_j^0 is equal to $L\left(d_j\right)$ minus its absolute value, and e_j^1 is the absolute value of $L(d_j)$. In contrast, if $L(d_j)$ is greater than "0", e_j^0 takes $L(d_j)$ value, and e_j^1 is equal to $L(d_j)$ minus its absolute value. It can be obtained by (21) and (22).

$$\begin{aligned} e_j^0 &= L(d_j) - \left|L(d_j)\right| \\ e_j^1 &= \left|L(d_j)\right| \qquad \left(L(d_j) < 0\right) \end{aligned} \tag{21}$$

$$\begin{aligned} e_j^0 &= \left|L(d_j)\right| \\ e_j^1 &= L(d_j) - \left|L(d_j)\right| \qquad \left(L(d_j) \geq 0\right) \end{aligned} \tag{22}$$

At this time, $L(d_j)$ is normalized to its maximum value in order to keep the values as close as possible to "1" and "0". The calculated values, e_j^0 and e_j^1, like the (21), (22) are updated to BSM, separated as the state S^j is "0" and "1", respectively.

$$\hat{\beta}_j^{s^j} = \beta_j^{s^j} + e_j^0 : d_j = 0 . \tag{23}$$

$$\hat{\beta}_j^{s^j} = \beta_j^{s^j} + e_j^1 : d_j = 1 \tag{24}$$

Using the updated BSM values, $\hat{\beta}_j^{s^j}$ according to (23) and (24), we can obtain an equation for $L(r_j)$ as follows.

$$\begin{aligned} L(r_j) = &\max{}^*\left[\alpha_{j-1}^{s^{j-1}} + \gamma_j^{s^{j-1},s^j} + \hat{\beta}_j^{s^j}\right]\left(s^{j-1}, s^j\right) : d_j = 0 \\ &- \max{}^*\left[\alpha_{j-1}^{s^{j-1}} + \gamma_j^{s^{j-1},s^j} + \hat{\beta}_j^{s^j}\right]\left(s^{j-1}, s^j\right) : d_j = 1 \end{aligned} \tag{25}$$

At this time, after adding e_j^0 and e_j^1 again to the respective max values for "0" and "1", we calculate the value of LLR as in (26).

$$\begin{aligned} L(r_j) = &\max{}^*\left[\alpha_{j-1}^{s^{j-1}} + \gamma_j^{s^{j-1},s^j} + \hat{\beta}_j^{s^j} + e_j^0\right]\left(s^{j-1}, s^j\right) : d_j = 0 \\ &- \max{}^*\left[\alpha_{j-1}^{s^{j-1}} + \gamma_j^{s^{j-1},s^j} + \hat{\beta}_j^{s^j} + e_j^1\right]\left(s^{j-1}, s^j\right) : d_j = 1 \end{aligned} \tag{26}$$

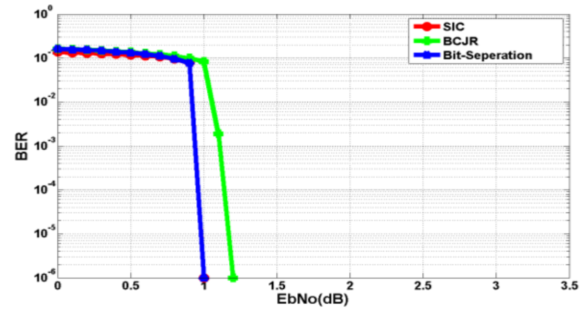
The $\max^*$ operator is defined as

$$\max{}^*(x, y) = \max(x, y) + \log(1 + e^{|x-y|}) \tag{27}$$

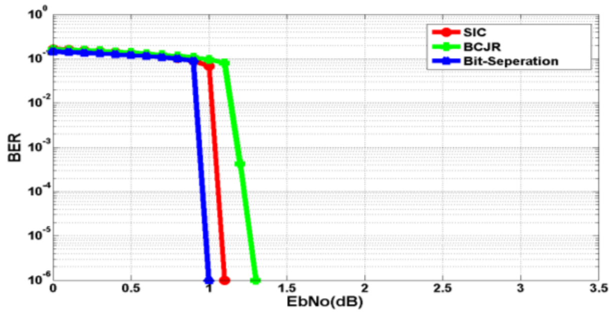
Iterating these steps, the values of the external input separated in accordance to the status of S_j continues to be added, and, as the number of iteration increases, the updated error correction value to be sent gets closer to the raw signal, and consequently the performance improves.

IV. Simulation Results

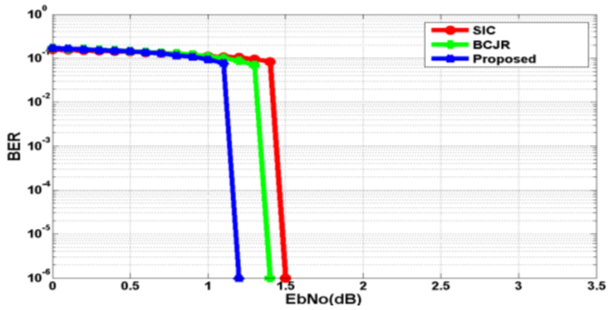
To illustrate the performance of the proposed scheme, we compared its BER curve to the ones of SIC and BCJR equalization according to τ'. The LDPC encoder employed K=32400, as the size of the source bit, and R=1/2, as the rate of encoding. The SRRC filter with roll-off factor of 0.35 and number of sampling per bit of 24 and the number of tap is 1 are used. We fixed number of iteration of LDPC decoder set to 60 and of total iteration set to 5. In Fig. 6, three types of methods mentioned in this paper are compared in the view of BER according to τ`.



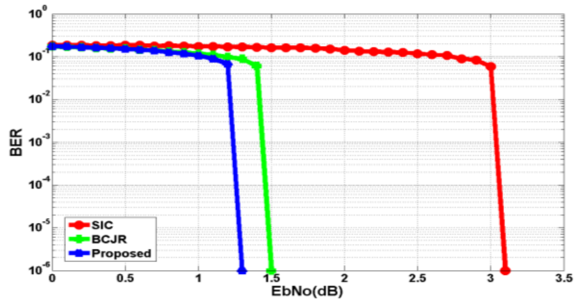
(a) $\tau' = 10\%$



(b) $\tau' = 20\%$



(c) $\tau' = 30\%$



(d) $\tau' = 40\%$

Fig. 6. Bit error performance according to various τ' (only FTN method)

When $\tau' = 10\%$, the performance proposed scheme is almost the same as that of SIC scheme. When τ' is higher, the performance of proposed scheme is being better that of the other scheme. As the interference increases, the SIC scheme cannot obtain the accurate amount of interference because it performs LDPC decoding using distorted signals, and so, as τ increases, shows drastic performance degradation compared to BCJR equalization scheme and the proposed method. Moreover, we know that the proposed method is more efficient because it updates external input value closer to the raw signal compared to that of BCJR equalization scheme.

In addition, we compared the performance of the proposed P-FTN method and the high coding rate LDPC method. As shown in (7), in order to compare same rate of 9/10 for LDPC codes, we set the parameters of P-FTN as shown in Table I.

TABLE I. THE CODING RATES ACCORDING TO THROUGHPUT

	High coding rate LDPC	P-FTN
R (coding rate)	9/10	1/2
K (source bits)	58320	32400
N (coded bits)	64800	64800
ρ (punctured rate)	-	1/3
τ' (interference ratio)	-	12.5%

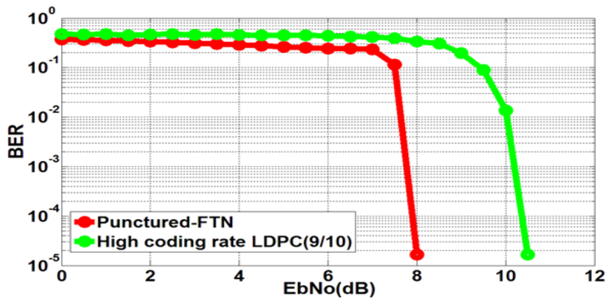


Fig. 7. The performance of P-FTN method and high coding rate LDPC method ($K = 32400$, $\rho = 1/3$ $\tau' = 12.5\%$, $N = 64800$)

Fig. 7. shows the result of BER performance comparison between P-FTN and high coding rate LDPC at BER of 10^{-4} with the same throughput efficiency.

According to Fig. 7, we confirmed that the performance of proposed P-FTN method improved to approximately 2 dB more than that of high coding rate LDPC.

V. CONCLUSIONS

Generally, in LDPC coded system, to increase throughput efficiency, high coding rates are employed in satellite and/or terrestrial wireless system. However, the more the coding rate increases, the more performance decreases sharply. The other representative method in recent is FTN method which transmit faster than Nyquist rate. Therefore, this paper proposed P-FTN method which combine punctured and FTN methods. In aspect to receiver side, the structure in order to compensate ISI induced by FTN processing is very important role to improve performance. Many schemes are announced in order to combat to ISI such as SIC and BCJR equalizations. Because SIC scheme needs accurate amount of ISI, BCJR equalizations which removes the interferences by trellis diagram is applied mainly. However, in whole iteration of BCJR equalizer and LDPC decoder, the performance does not improve due to inefficient extrinsic exchange. In feeding extrinsic information of LDPC decoder to BCJR equalizer, we added bit separation block which separates the LDPC decoding signals by symbol-by-symbol units and applies them as extrinsic input. As a result, as number of whole iteration is increased, we made the performance improved.

Based on proposed receiver model, we compared performance between high coding rate and P-FTN method through the computer simulation at the same transmission rate. P-FTN method has coding gain of 2 dB at BER of 10^{-4} compared to high coding rate LDPC method. Therefore, we confirmed the P-FTN method with proposed receiver structure is very useful algorithm in increasing throughput efficiency. In the future, we will evaluate performance and complexity between various interference rates and coding rates. We will suggest optimal receiver structure for high throughput wireless communications.

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