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Peak Power Reduction on OFDM Signals using BICM Technique

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Abstract – Recent mobile systems equipped to carry high volume and changeable bit rate transmission with bandwidth efficiency to maintain limited spectrum resource. One of the most severe problems in OFDM transmission is that, it shows a high peak-to-average ratio. In certain cases all the signal component can be put in phase and an enormous output is obtained and in several cases, they may revoke each other producing zero output. As the peak-to-average ratio (PAR) of the OFDM system is extremely huge. Thus Amplitude Clipping, Partial Transmit Sequence (PTS) and Selective Mapping (SLM) which affects the side information with huge high PAPR, to avoid this clipping without affecting side information by introducing Bit Interleaved Coded Modulation to reduce PAPR in orthogonal frequency division multiplexing (OFDM) systems. The intention of this paper is to describe the framework used to investigate the PAPR performance of OFDM with higher order modulation schemes and present simulation results for PAPR for Bit Interleaved Coded Modulation schemes.

Keywords: *Orthogonal Frequency Division Multiplexing (OFDM), Peak-to-Average Power Ratio (PAPR), Partial Transmit Sequence, Selective Mapping.*

1. INTRODUCTION

Orthogonal frequency division multiplexing (OFDM) has been adopted for high-speed wireless data transmission systems due to its inherent robustness against multipath fading channels. OFDM systems suffer from high peak to average power ratio (PAPR) which is one of the major drawbacks of all multicarrier transmission schemes. A wide dynamic range is required in the linear Power Amplifiers (PA) at the transmitter in order to transmit a signal with large PAPR. Owing to this, the difficulty of digital-to-analog converters (DAC) and the linear choice of high power amplifiers (HPA) have to be improved to avoid signal distortion, severely demoting system act.

Lately, various methods have been recommended to lessen the PAPR of OFDM signals in the literature, counting clipping, nonlinear companding transforms, coding technique, selected mapping (SLM), partial transmit sequence (PTS), time-domain symbol combining, adaptive projected sub gradient method, constellation modification, and tone reservation. In addition several methods are proposed to manage the PAPR in OFDM systems, like using fountain codes. Comparing these methods, the PTS and SLM methods are the mainly attractive and important methods as their fine performance in terms of PAPR reduction. Their essential idea is to produce numerous alternative OFDM signals by multiplying every original data sub block (i.e., cluster of some subcarriers) with different phase revolution vectors. Then, the formed signal with the least PAPR is selected to be transmitted, and its equivalent phase rotation vectors for sub blocks are named as optimal phase rotation combination. As a result, parts of sub carriers are occupied to construct the optimal phase rotation combination as side information to recover the original data at the receiver in the conventional PTS and SLM methods. Obviously, when the side information is infected since fading channel and noise, etc., the original data cannot be accurately retrieved. Therefore, the bit-error-rate (BER) performance is greatly corrupted. Therefore, some extra guard bits have to be borrowed for the protective transmission of the side information to make sure a reason skill BER performance, which grades in a decrease in spectrum efficiency. Thus, some extensions of the SLM and PTS methods have been optional to cut behind the quantity of the packed subcarriers for the side information transmission.

Here in our research, BICM is combined with OFDM and it was exposed in that the BICM-OFDM system is skill to obtain both the spatial and frequency diversities in autonomous and identically circulated Rayleigh fading channels. Furthermore studies of BICM-OFDM can be recognized in for generalized channel models. This can be accomplished just if the receiver has been fine trained all the way through an exact quantity of training or pilot data and therefore it assume the PAPR. Other

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than such (training) pilot-aided schemes usually work fine beneath slow time-varying channels, and are frequently implemented in current wireless communication systems, they may though no longer be financial beneath fast time-varying channels. With less coherence times, which implies extra recurrent training, both pilot overhead and CSI estimation error could become famous issues. Simulation results too propose that the proposed method has got the similar aptitude of the PAPR reduction as that of the predictable PTS and SLM methods with ideal side information, and could provide fine BER results.

This paper is organized as follows. Section 2 presents the PAPR formulation in OFDM systems. In Section 3, we analyze the combination of BICM and OFDM. Performance evaluation and comparisons are given in Section 4. Section 5 presents the conclusion.

2. PAPR FORMULATION IN OFDM SYSTEMS

To enhance estimate the transmitted OFDM signal, the discrete-time signal is obtained by employing the process of -point inverse discrete Fourier transform (IDFT) to the input unique data block with zero-padding, where the oversampling factor is an optimistic integer. Then, the OFDM signal is expressed as

$$x(n) = \frac{1}{\sqrt{N}} \sum_{k=0}^{N-1} X_k e^{j\frac{2\pi nk}{LN}}, \quad n = 0, 1, \ldots, LN-1 \tag{1}$$

Normally, the PAPR of OFDM signals is defined as the relation among the greatest immediate power and its average power, i.e.

$$\mathrm{PAPR} = \frac{\max\limits_{0 \le n \le NL-1} |x(n)|^2}{E[|x(n)|^2]} \tag{2}$$

2.1. Partial Transmit Sequence

For the conventional PTS method, the input data block is partitioned into disjointed subblocks, which is an integer. Noticeably, the mass of subblocks is the equivalent as that of the original data block. Furthermore, it satisfies

$$\mathbf{X} = \sum_{m=1}^{M} \mathbf{X}^m \tag{3}$$

So, the quantity of nonzero elements with IDFT operation, the partitioned subblocks are altered into time domain signals. After that, the substitute signal could be obtained by multiplying with phase rotation factors as

$$\mathbf{x} = \sum_{m=1}^{M} b^m \mathbf{x}^m \tag{4}$$

Then, the object of the PTS method is to locate the finest mixture of the phase rotation factors to reduce the PAPR of the alternative signal.

Usually, the phase rotation factors are limited to a set with elements. In the meantime, the receiver should have the information about the generation procedure of the transmitted OFDM symbol. Therefore, the finest phase rotation mixture with the least PAPR should be transmitted as the side information.

2.1. Selected Mapping

For the SLM method, phase rotation vectors are generated as

$$\mathbf{P}^u = \left[P_0^u, P_1^u, \ldots, P_{N-1}^u\right], \quad u = 1, 2, \ldots, U \tag{5}$$

Consequently, the input data block is multiplied by each one to attain alternative sequences as

$$\mathbf{X}^u = \left[P_0^u X_0, P_1^u X_1, \ldots, P_{N-1}^u X_{N-1}\right], \quad u = 1, 2, \ldots, U. \tag{6}$$

After that, the alternative sequences are altered into the time domain signals via IDFT process, and the signal with the least PAPR is searched to be transmitted.

Similar to the PTS method, it is frequent to limit the phase rotation factors to a set with elements in the SLM method. Usually, it is reason skill to suppose that the phase rotation vectors are identified for both the transmitter and receiver, and so, as side information, the index of the phase rotation vector with the least PAPR needs to be transmitted for the receiver to recover the data series.

2.3. Side Information

For the traditional PTS and SLM methods, the side information is required to inform the receiver of the optimal phase rotation vector.

In order to show why the side information needs to be transmitted in the PTS and SLM methods, we give some intuitive results of the traditional S-QAM denoted by. As shown in Fig. 1, in which and the minimum distance between the constellation points. When is multiplied by a phase rotation factor, we have. Likewise, is obtained when the phase rotation factor is. Clearly, equals to since is an integer multiple of, which implies that we cannot determine which value the phase rotation factor is without the side information. So, for the conventional PTS and SLM methods, the side information of the phase rotation factors has to be transmitted and protected by extra bits. Even though it has lot of advantages, its computational complexity is high as this method can't divide the frequency vector into sub-blocks.

3. BICM-OFDM

The system deploys just one transmit and one receive antenna (SISO). One OFDM symbol has subcarriers, where all subcarrier corresponds to a symbol from a constellation map as given in Section II, constellation size. A convolutional encoder is used to generate the binary code at the transmitter. For the rate convolutional code with a given number of states, the one with the highest minimum Hamming distance is picked from skills, e.g. the output bit of a convolutional encoder is interleaved and mapped onto the subcarrier at the location. The interleaver should be designed such that successive coded bits are:

1) Mapped on different symbols;
2) Transmitted over various subcarriers;
3) Interleaved within one OFDM symbol to avoid an additional delay constraint to start decoding at the receiver.

Regard as a frequency-selective channel with taps given by. Each tap is assumed to be statistically free and modeled as a zero-mean complex Gaussian random variable with unit variance. The fading model is assumed to be quasi-static, i.e., the fading coefficients are constant above the transmission of one packet, other than independent from one packet transmission to the next. It is unspecified that the taps are spaced at integer multiples of the symbol periods, which is the worst-case scenario in terms of designing full diversity codes

A cyclic prefix (CP) of suitable span is added to every OFDM symbol. Adding CP converts the linear convolution of the transmitted signal and the -tap channel keen on a spherical convolution. while CP is removed and fast Fourier transform (FFT) is taken at the receiver, the received signal is set by

$$y(k) = H(k)x(k) + n(k), \quad 0 \le k \le K-1 \tag{7}$$

Where the transmitted signal on the subcarrier is complex additive white Gaussian noise (AWGN) with zero mean and variance SNR, and is given by

$$H(k) = \underline{\mathrm{W}}_K^H(k)\mathbf{P}\underline{\mathrm{h}} \tag{8}$$

Where the vector with an oblique matrix with, for, on the main diagonal representing the PDP of the frequency-selective channel. PDP matrix entries' are real and severely positive. Note that the transmitted symbols are assumed to have common energy of 1. As a result, with the channel, PDP, and AWGN models described now, the received signal-to-noise ratio is SNR.

Bit-interleaved coded modulation (BICM) is a flexible modulation/coding scheme which allows the designer to select a modulation constellation separately of the coding rate. This is since the output of the channel encoder plus the input to the modulator is divided by a bit-level interleaver. In order to increase spectral efficiency, BICM be able to combined with high-order modulation schemes such as quadrature amplitude modulation (QAM) or phase shift keying. BICM is mainly well suitable for fading channels, and it simply introduces a small penalty in terms of channel volume while compared to the coded modulation volume for together additive white Gaussian noise (AWGN) and fading channels. Moreover, if the so-called BICM with iterative decoding (BICM-ID) is used, the didapper and decoder iteratively swap information, improving the system performance. On the receiver's side of BICM, the reliability metrics are designed for the coded bits under the shape of logarithmic likelihood ratios, or simply L-values. These metrics are later deinterleaved and further used by the soft-input channel decoder. This thesis deals with the probabilistic classification of the L-values designed by the didapper while BICM is used in combination with high order QAM schemes. Three assistances are included in this thesis. In Paper A the issue of the probabilistic modeling of the extrinsic L-values for BICM-ID is addressed. First with a easy piece-wise linear model of the L-values obtained via the max-log estimate, expressions for the possibility density functions (PDFs) for Gray-mapped 16-QAM are originate. The developed logical expressions are then used to proficiently calculate the so-called extrinsic information transfer functions of the didapper, and they are as well compared with the histograms of the L-values obtained during numerical simulations. In Paper B closed-form expressions for the PDFs of the L-values in BICM by Gray mapped QAM constellations are developed. Based on these expressions, two simple Gaussian mixture approximations that are analytically tractskill are also projected. The developments are used to powerfully compute the BICM channel volume and to develop bounds on the coded bit-error rate while a convolutional code is used.

The coded performance of a hybrid automatic repeat request based on constellation rescheduling is also evaluated. In Paper C closed-form expressions for the PDFs of the L-values in BICM transmissions with Gray-mapped QAM constellations over fully-interleaved fading channels are derived. The results are special for a Rayleigh fading channel, though, developments for the general case of a Nakagami-m case are also integrated. By means of the developed expressions, the performance of BICM transmissions using convolutional and turbo codes is efficiently evaluated. The BICM channel volume for dissimilar fading channels and constellation sizes is also designed.

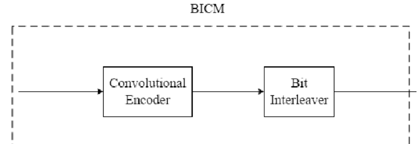

Fig 1 Block diagram of BICM

3.1. Convolution Encoder

The essential functioning principle of a convolutional encoder is that the encoder performs a convolution of the input stream with encoder's urge response. As a result, a binary information series *u* is split into l information symbols *dk dk*,, *dk* 0 1 = f _ ^ h ^l - hi. In case of binary input, l = 1, hence *u d d k k k* 0 = = ^ h. The information symbols are later mapped by the encoder on coded symbols *xk xk*,, *xk* 0 1 = f _ ^ h _h - ii of h bits. So, the extent of idleness can be described by the code rate *R* = h l. The code rate is one of the two parameters convolutional codes are generally classified. The second parameter is the constraint length *Kc* which denotes the time span of bits that each input bit affects. A more solid explanation of the constraint length is *K M* 1 *c*= + where *M* indicates the length of the shift register or "memory length" of the encoder. The constraint length is regularly limited to *K* 9 *c*= or less owing to the exponentially rising decoding computational supplies [3].

Fig. 1 depicts a rate 1/2, limitation length *K* 3 *c*=, non-systematic, non-recursive convolutional encoder. The encoder is non-recursive since it does not use linear response

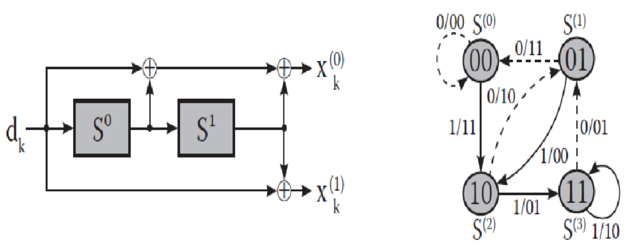

Fig. 2 Non-recursive, Non-systematic Convolutional Encoder (NSC)

Fig. 2. Shows that state diagram corresponds to the shift-register execution to the left. For convolutional codes, it is more suitable to demonstrate encoders by their equivalent state diagrams. As the encoder consists out of *M* shift registers, there is a finite number of 2M states *S(m)* with *m* 0,, 2 1 *M* = f -, hence the encoder can be measured as a 2M-state finite state machine. Each state *S(m)* is represented by a node, each state transition is illustrated by a directed edge. The edges are labeled with the input bit *dk* and the output symbols *xk*, in the example shown above *d* /*x x*( ) *k k k* ^0h 1. Dashed edges show state transitions where *d* 0 *k*=, solid edges represent state transitions where *d* 1

The trellis diagram depicted in Fig. 3 represents the state diagram of Fig. 1 unrolled over occasion. For every information symbol *dk*, the trellis diagram contains every probable state transitions arising from the preceding encoder state. Note that as in the state diagram, dashed edges show state transitions where *d* 0 *k*=0, solid edges stand for state transitions where *d* 1 *k*= 1.

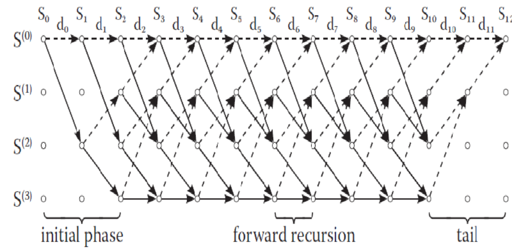

Fig. 3 Trellis Diagram for 4-State-NSC

The trellis illustrated in Fig. 4 assumes that the primary state of the encoder is state *S(0)*. The trellis diagram can be rip in three phases: For a message of the length *L*, after the initial phase of *M* encoded bits *dk*, there are *L* - *M* identical trellis segments. After *L* encoded bits, the encoder is required to the all nothing state by adding a tail of *M* symbols. So, the coded message length is amplified to *L L M T*= +, the code rate is decreased to *R R L L T T* =. This method improves the error security of the information symbols at the end of encoded message.

To explain the VA in the next section more hands-on, the subsequent 12-bit information sequence *d* will be used: *d* =^0, 1, 0, 1, 1, 0, 0, 1, 1, 1, 0, and 0 h. This results in the trellis depicted in Figure 3:

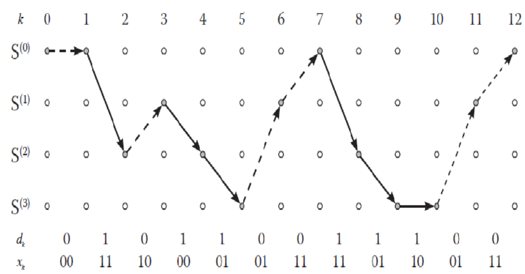

Fig. 4 Trellis Diagram for Exemplary Information Sequence

3.2. Bit Interleaving

In digital transmission a voice sample is changed to 8 binary bits. These bits will be sent in 3.9us to the other end all the way through the transmission media if the multiplexing used is CEPT (Committee of European Post and Telecom). It is necessary to send these 8 bits in serial in 3.9us. Or else if we use 8 media in 488ns the sample information will be received at the other end. The latter method is not used due to the following reasons.

1. As an alternative of one transmission media 8 transmission media to be used. Hence cost is high.
2. Even 8 transmission Medias deployed and it will not have comparable characteristics for the buts in transmission. Leading the receiver to install extra techniques to coordinate the bits for a word. At this point a word means a sample information of 8 bits.

Therefore in the essential PCM of 2.048 Mbps for all time 8 bits pertaining to sample information is sent in sequence. As there are 32 Time Slots (30 channels are multiplexed in the CEPT system). Every bits pertaining to a model are serially interleaved and sent in one transmission media. However while a multiplexing is passed out in the higher order the basic information to the higher order multiplexer is a bit pertaining to a primary tributary of 488ns and not 8 bits pertaining to a sample. Two interleaving methods can be adopted.

Our method can have the ability to divide the frequency vector into sub-blocks so that it can avoid the full IFFT (Inverse Fast Fourier Transform) operations. Therefore it is more advantageous than the existing methods, if amount of computational complication is limited.

4. SIMULATION RESULTS

Fig 5 shows the relationship of PAPR reduction in BICM with other methods in literature. The above figure proves that the proposed method can reduce PAPR level more than other methods.

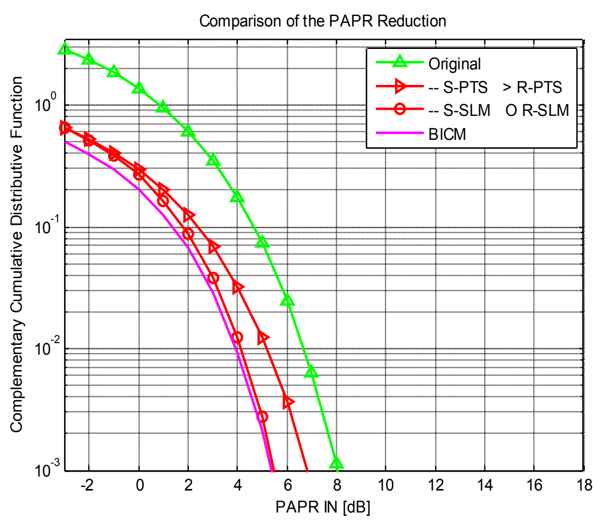

Fig. 5. Comparison of PAPR reduction

As shown in Fig. 6, the theoretical results of the SER performance are great agreement with the simulation results for the S-QAM and R-QAM constellations when, 16, 64, respectively. Furthermore, it is clear that the 4-ary R-QAM constellation requires about 0.7 dB bit SNR more than the 4-ary S-QAM to attain the same SER performance. However, the SER performance of the 64-ary R-QAM constellation is more or less the same as that of the 64-ary S-QAM constellation. That is since that the mean power of the 64-ary R-QAM constellation is only 1.3% slighter than that of the 64-ary S-QAM constellation.

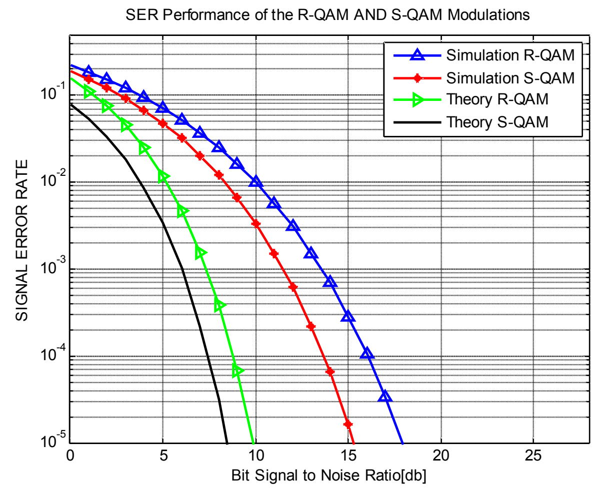

Fig. 6. Outage possibility of CSS under nakagami fading channels

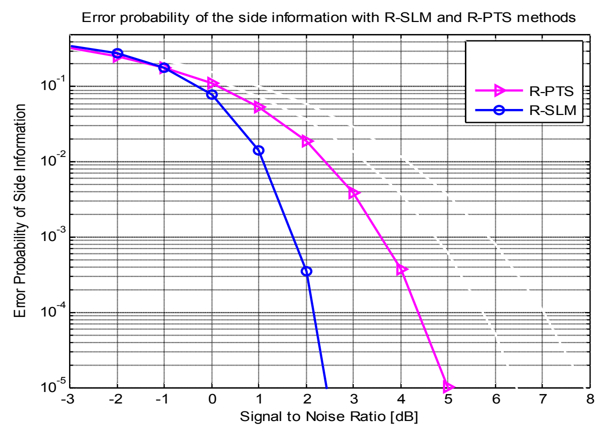

Fig. 7. Error probability of the side information with R-SLM and R-PTS methods

As shown in Fig. 7, the error probability of the side information with the R-SLM method is better than that of the R-PTS method, in which for the R-SLM method, for the R-PTS method, and the phase rotation factors are chosen from. However, the error probability of the R-PTS method is when dB, which is small enough to be accepted in practice.

Fig. 8 shows the comparison of bit interleaved coded modulation method and SLM & PTS methods.

5. CONCLUSION

OFDM has been seen as the vital method of the upcoming communication systems as it has several advantages. In contrast, the OFDM system suffers from various disadvantages. High PAPR results in decrease of effectiveness of the Power Amplifier. In our paper, we projected a novel BICM-OFDM method, to lessen the PAPR with no side information in OFDM systems.

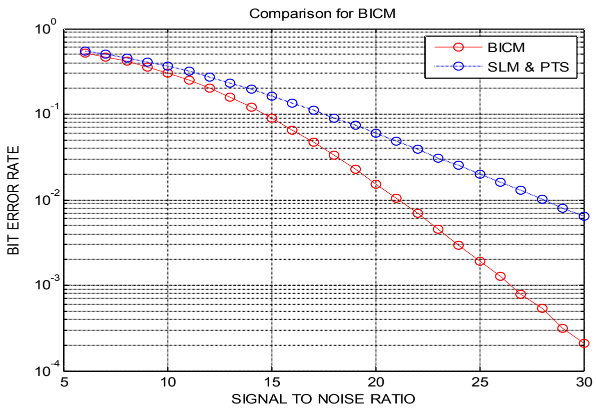

Fig. 8. Comparison betweem BICM with SLM & PTS

Our method separates the frequency vector into various sub-blocks by applying the phase alteration. So, some of the difficulties of numerous full IFFT operations can be avoided in BICM-OFDM method. Therefore it is more beneficial than the existing methods, if amount of computational complication is limited. Theoretical study and simulation results showed that the proposed method could attain the PAPR reduction than the PTS and SLM methods. The BER performance of the proposed methods is moreover improved than the existing methods with ideal side information in uncoded and coded OFDM systems over AWGN and fading channels, respectively. In future our system can be extended to a relay assisted distributed BICM - OFDM system, and a complexity-reduced implementation method. This system can lessen the PAPR further more so that the effectiveness of the system can be amplified.

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