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PAPR Reduction in OFDM using Joint Hybrid Techniques

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***Abstract*:- The Orthogonal Frequency Division Multiplexing (OFDM) has been known since 1966, but it only reached sufficient maturity for deployment in standard systems during 1990s. OFDM is an attractive modulation technique for transmitting large amounts of digital data over radio waves. One major disadvantage of OFDM is that the OFDM signal which is a sum of several sinusoids leads to high peak to average power ratio (PAPR).Number of techniques have been PROPOSED in the literature for reducing the PAPR in OFDM systems. In this work we are using the various hybrid techniques with joint algorithms are proposed for reducing the PAPR and the selection criteria for choosing these techniques have been discussed. The goal is we are mixing the different hybrid techniques and then see the results This is done primarily to give an overview of the various hybrid techniques known today for PAPR reduction.**

1. INTRODUCTION

New Technologies and thereby new applications are emerging not just in wired environment but also in the wireless arena. The next generation mobile systems are expected to provide a substantially high data rate to meet the requirements of future high performance multimedia applications.

The minimum target data rate for the 4G system is expected to be at 10-20 Mbps and at least 2 Mbps in the moving vehicles. To provide such a high data rate with high spectral efficiency, a new modulation scheme is to be used. A promising modulation technique that is increasingly being considered for adoption by 4G community is OFDM . Existing 3G systems uses single carrier modulation technique whereas OFDM which is otherwise known as Multicarrier Modulation (MCM) / Discrete Multitone Technique (DMT) sends a high speed data stream by splitting it up to multiple lower speed stream and transmitting it over a lower bandwidth subcarriers in parallel. OFDM has several favorable properties like high spectral efficiency, robustness to channel fading, immunity to impulse interference, uniform average spectral density, capacity to handle very strong echoes and less non-linear distortion.

OFDM is the modulation technique used in many new broadband communication systems. In recent years OFDM has emerged as the standard of choice in a number of important high data applications.

OFDM is employed in Digital Television Broadcasting (such as the digital ATV Terrestrial Broadcasting), European Digital Audio Broadcasting (DAB) and Digital Video Broadcasting Terrestrial (DVB-T), Wireless Asynchronous Transfer Mode (WATM) and numerous Wireless Local Area Networks (e.g. IEEE 802.11a operating at 5 GHz) and European Telecommunications Standard Institute (ETSI) Broadband Radio Access Networks (BRAN)'s High Performance Radio Local Area Network (HIPERLAN) Type-2 standard.

2. ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING (OFDM):

OFDM is a Multicarrier Transmission technique which divides the available spectrum into many carriers each one being modulated by a low data rate stream. OFDM is similar to Frequency Division Multiple Access (FDMA) in that the multiple user access is achieved by sub-dividing the available bandwidth into multiple channels, which are then allocated to users.

However OFDM uses the spectrum much more efficiently by spacing the channels more closer together. This is achieved by making all the carriers orthogonal to one another, preventing interference between the closely

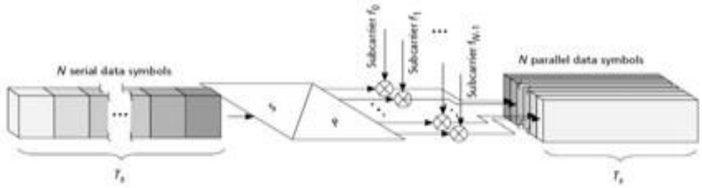


Fig.1. Multicarrier Transmission Technique

In FDMA each user is typically allocated a single channel which is used to transmit all the user information. The bandwidth of each channel is typically 10-30 kHz for voice communication. However, the minimum required bandwidth for speech is only 3 kHz. The allocated bandwidth is made wider than the minimum amount required to prevent channels from interfering with one another. This extra bandwidth is to allow for signals of neighboring channels to be filtered out and to allow for any drift in the center frequency of the transmitter or receiver. In a typical system up to 50% of the total spectrum is wasted due to the extra spacing between channels. This problem becomes worse as the channel bandwidth becomes narrower and the frequency band increases.

Time Division Multiple Access (TDMA) overcomes this problem by using wider band width channels which are used by several users. The subcarriers in an OFDM signal are spaced close as is theoretically possible which maintain orthogonality between them.

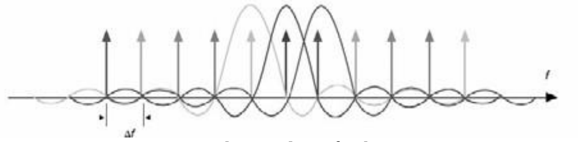


Fig.2. Orthogonality of subcarriers

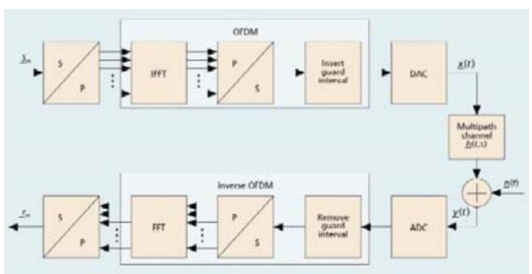
The orthogonality of the carriers means that each carrier has an integer number of cycles over a symbol period. Due to this the spectrum of each carrier has a null at the center frequency of each of the other carriers in the system. This results in no interference between the carriers, allowing them to be spaced as close as theoretically possible.

3. OFDM SYSTEM MODEL

To generate OFDM successfully the relationship between all the carriers must be carefully controlled to maintain the orthogonality of the carriers. For this reason, OFDM is generated by firstly choosing the spectrum required based on the input data, and modulation scheme used. Each carrier to be produced is assigned same data to transmit.

The required amplitude and phase of them are calculated based on the modulation scheme. The required spectrum is then converted back to its time domain signal using an Inverse Fourier Transform (IFT). In most applications, an Inverse Fast Fourier Transform (IFFT) is used.

The IFFT performs the transformation very efficiently and provides a simple way of ensuring the carrier signals produced are orthogonal.



The Fast Fourier Transform (FFT) transforms a cyclic time domain signal into its equivalent frequency spectrum. This is done by finding the equivalent waveform, generated by a sum of orthogonal sinusoidal components.

The amplitude and phase of the sinusoidal components represent the frequency spectrum of the time domain signal. The IFFT performs the reverse process, transforming a spectrum (amplitude and phase of each component) into a time domain signal. An IFFT converts a number of complex data points, of length that is a power of 2, into the time domain signal of the same number of points. Each data point in frequency spectrum used for an FFT or IFFT is called a bin.

The orthogonal carrier required for the OFDM signal can be easily generated by setting the amplitude and phase of each frequency bin, thus performing the IFFT.

4. PROBLEM OF PEAK-TO-AVERAGE POWER RATIO IN OFDM SYSTEMS

High Peak-to-Average Power Ratio has been recognized as one of the major practical problem involving OFDM modulation. High PAPR results from the nature of the modulation itself where multiple subcarriers / sinusoids are added together to form the signal to be transmitted. When N sinusoids add, the peak magnitude would have a value of N, where the average might be quite low due to the destructive interference between the sinusoids. High PAPR signals are usually undesirable for it usually strains the analog circuitry.

High PAPR signals would require a large range of dynamic linearity from the analog circuits which usually results in expensive devices and high power consumption with lower efficiency (for e.g. power amplifier has to operate with larger back-off to maintain linearity). In OFDM system, some input sequences would result in higher PAPR than others. For example, an input sequence that requires all such carriers to transmit their maximum amplitudes would certainly result in a high output PAPR.

Thus by limiting the possible input sequences to smallest sub set, it should be possible to obtain output signals with a guaranteed low output PAPR.

The PAPR of the transmit signal x(t) is the ratio of the maximum instantaneous power and the average power.

By definition

$$x = -\infty < x < \infty([x(t)]2)/E\{|x(t)|\}2$$

a signal is a sum of N signals each of maximum amplitude equal to 1 Volt, then it is conceivable that we could get a maximum amplitude of N Volts, that is, all N signals add at a moment at these maximum points.

For an OFDM signal, that has 126 carriers each with normalized power of 1W, then the maximum PAPR can be as large as 10 log10 126 or 21 db. This is at the instant when all 126 carriers combine at their maximum point unlikely but possible. The RMS PAPR will be around half of the number as 10-12 db. The large amplitude variation increases in-band noise and increases the Bit Error Rate (BER) which the signal has to go through amplification nonlinearities. The crest factor is widely used in the literature as well, which is defined as the square root of the PAPR. Crest Factor, FPAPR C = .. (2) High PAPR / crest factor

could cause problems when the signal is applied to a transmitter which contains non-linear components such as High Power amplifier (HPA) in the Transmitter chain. The PAPR has the worst case value PAPRWC which depends on the no. of subscribers N. The non-linear effects on the transmitted OFDM symbols are spectral spreading, inter-modulation and changing the signal constellation. In other words, the nonlinear distortion causes both in-band and out-of-band interference to signals.

The inband interference increases the Bit Error Rate (BER) of the received signal, while the out-of band interference causes adjacent channel interference through spectral spreading. A better solution is to prevent the occurrence of such nonlinear distortion by reducing PAPR of the transmitted signal with some manipulation of the OFDM signal itself.

5. PAPR REDUCTION TECHNIQUES

Several techniques have been proposed in the literature to reduce the PAPR. These techniques can mainly be categorized in to signal scrambling techniques and signal distortion techniques. Signal scrambling techniques are all variations on how to scramble the codes to decrease the PAPR. Coding techniques can be used for signal scrambling.

Golay complementary sequences, Shapiro-Rudin sequences, M sequences, Barker codes can be used efficiently to reduce the PAPR. However with the increase in the number of carriers the overhead associated with exhaustive search of the best code would increase exponentially. More practical solutions of the signal scrambling techniques are block coding, Selective Level Mapping (SLM) and Partial Transmit Sequences (PTS). Signal scrambling techniques with side information reduces the effective throughput since they introduce redundancy.

The signal distortion techniques introduce both Inband and Out-of-band interference and complexity to the system. The signal distortion techniques reduce high peaks directly by distorting the signal prior to amplification. Clipping the OFDM signal before amplification is a simple method to limit PAPR.

However clipping may cause large out-of band (OOB) and in-band interference, which results in the system performance degradation. More practical solutions are peak windowing, peak cancellation, Peak power suppression, weighted multicarrier transmission, companding etc.

A. Signal Distortion

1) Clipping & Filtering: A threshold value of the amplitude is set in this process and any sub-carrier having amplitude more than that value is clipped or that sub-carrier is filtered to bring out a lower PAPR value. It introduces distortion and no power raise. It is one of the simplest technique to apply.

2) Peak Windowing: Peak windowing reduces PAPRs at the cost of increasing the BER and out-of-band radiation. In peak windowing method we multiply large signal peak with a specific window, for example; Gaussian shaped window, cosine, Kaiser and Hamming window.

B. Signal Scrambling Techniques

1) Selected Mapping: In this a set of different data blocks representing the information same as the original data blocks are selected. Selection of data blocks with low PAPR value makes it suitable for transmission.

2) Partial Transmit Sequence: This technique helps in reducing distortion.no power raise.It is complex in nature. Transmitting only part of data of varying sub-carrier which covers all the information to be sent in the signal as a whole is called Partial Transmit Sequence Technique.

3) Interleaving: The notion that highly correlated data structures have large PAPR can be reduced, if long correlation broken down. The basic idea in adaptive interleaving is to set up an initial terminating threshold. PAPR value goes below the threshold rather than seeking each interleaved sequences.

4) Tone Reservation (TR) The main idea of this method is to keep a small set of tones for PAPR reduction. This can be originated as a convex problem and this problem can be solved accurately. Tone reservation method is based on adding a data block and time domain signal. A data block is dependent time domain signal to the original multicarrier signal to minimize the high peak. The advantages reduces distortion effect and power gets raised in this technique.it is less complex in nature

5) Tone Injection (TI): in this technique reduces distortion effect and PAPR reduction is observed without data rate loss. It is based on additive method for PAPR reduction. Using an additive method achieves PAPR reduction of multicarrier signal without any data rate loss. It uses a set of equivalent constellation points for an original.

C. Coding

1) Block Coding: The fundamental idea is that of all probable message symbols, only those which have law peak power will be chosen by coding as valid code words for transmission.

TECHN-IQUE	DISTOR-TION	POWER	DATA LOSS	BER IMPROVED
PTS	YES	NO	YES	YES
BLOCK COADING	YES	NO	YES	YES
SLM	YES	NO	YES	YES
TI&TR	YES	YES	NO	YES
CLIPPING	NO	NO	NO	NO
PW	NO	YES	NO	NO

An overview of SLM and quantization clipping scheme:

As one of the most appealing techniques, selective mapping (SLM) provides an effective solution for Peak to-average Power Ratio (PAPR) reduction in Orthogonal Frequency Division Multiplexing (OFDM) systems, however, results in large computational complexity simultaneously. As compared to SLM, clipping provides a more simple way towards a better PAPR performance, whereas results in degradation of the Bit Error Rate (BER) performance. In this paper, we proposed a joint algorithm called SLM and Quantization Clipping (SLM-QC) scheme, which combined the conventional SLM technology with a modified quantization clipping scheme. The proposed scheme achieves a better exploitation of advantages brought about by each technique alone, with the tradeoff between the PAPR performance and the BER performance being well balanced.

1.Traditional SLM Scheme

The main idea of the SLM technique is to generate a set of sufficiently different candidate data blocks by multiplexing the frequency domain signal X with U different phase sequences, and then select the most favorable one for transmission.

In conventional SLM scheme, the distinct random phase rotation sequences $P^u = [P_0^u, P_1^u, \cdots, P_{N-1}^u],\ 0 \le u \le U - 1$ is generated with the unit-magnitude complex number. As shown in $[10]$, P_n^u are usually selected from {±1} or {±1,±j}. P^0 is usually all 1 sequence to include the original input sequence. U alternative OFDM symbol sequences are generated by component-wise multiplication of the input symbol sequence X_k and U phase sequences P^u, that is

$$X^u = X \otimes P^u = X_0^u, X_1^u, \cdots$$

$\otimes$ represents component-wise multiplication of two vectors. IFFT should be performed for each of the U input symbol sequences $\{X^1, X^2, ..., X^U\}$ to generate U alternative signal sequences as

$$x^u = IFFT(X \otimes P^u), 0 \le u \le U-1 \qquad (2)$$

2. Traditional Clipping Scheme

The fundamental principle of clipping algorithm is to detect the input signal's peak before the time domain OFDM signal is supplied to the power amplifier. If any part of the signal exceeds the thresholda non-linear processing should be applied to limit the amplitude within the preset threshold; Otherwise let the signal through without interference [5]. The clipped time domain signal can be written as:

$$\hat{x}(t) = \begin{cases} x(t), & |x(t)| \le A \\ Ae^{j\phi(t)}, & |x(t)| > A \end{cases} \qquad (3)$$

where $x(t)$ represents the time domain signal, A is the clipping threshold, and $\varphi(t)$ represents the phase of input $x(t)$. The most important parameter for the performance of clipping scheme is Clipping Ratio (CR), which is defined as

$$\mathrm{CR} = 20 \log \frac{A}{\sigma} \qquad (4)$$

where σ is the root mean square of signal power and can be expressed as:

$$\sigma = \sqrt{\frac{[x_0(t)]^2 + [x_1(t)]^2 + \cdots + [x_{N-1}(t)]^2}{N}} \qquad (5)$$

$x_0(t), x_1(t), \cdots, x_{N-1}(t)$ is the input signal, N is the number of subcarriers. As A denotes clipping threshold, a bigger CR corresponds a higher threshold, and the effect of reducing PAPR gets worse. And vice versa.

Study of SLM and Quantization Clipping Scheme:

In order to achieve a good PAPR reduction performance with low computational complexity, the conventional SLM and clipping schemes are jointly used in [11]-[13]. The SLM technique used in the first step can control the PAPR within a reasonable value, and then the Clipping technique is applied to further improve the PAPR reduction performance. The clipping operation following the SLM scheme can further improve the PAPR reduction performance, but it will increase the BER of the whole system.

To eliminate the negative effect brought by the clipping, we proposed a modified clipping scheme called Quantization Clipping (QC) as an alternative for the conventional one. The QC scheme can limit the distortion caused by the clipping scheme to a smaller level, thus reduces the bit error rate and band radiation of the whole system effectively. Fig. 1 is the block diagram of the SLM and quantization clipping scheme.

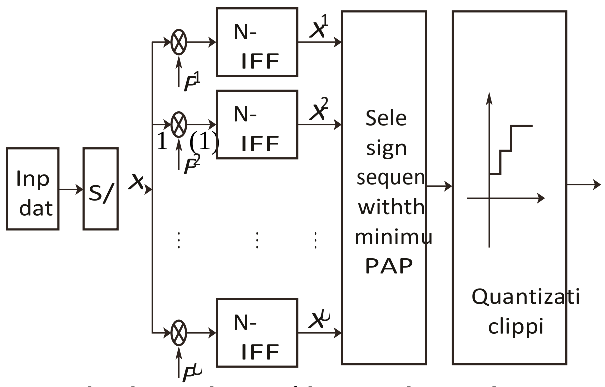


Fig. 1: The schematic diagram of the proposed SLM and QC joint scheme

Similar to the SLM and clipping scheme, the traditional SLM technique is used in the first step, and then the modified QC scheme is applied to further improve the performance of the whole system. Unlike the traditional clipping algorithm with only one threshold value, the quantized clipping algorithm contains several different preset thresholds so that the input signal would firstly be processed in a new way, in which the signal value between different intervals will be quantized to different threshold values.

Let $[A_1, A_2, ..., A_{m-1}, A_m]$ represents m thresholds and meet the inequality $A_1 < A_2 < \cdots < A_m$ and the equality $A_2 - A_1 = A_3 - A_2 = \cdots = A_m - A_{m-1} = s$, where s is the quantization step size. As with the conventional clipping scheme, the smaller the minimum threshold A_1, the better the PAPR reduction performance; nevertheless the system's BER will also increase. In the QC scheme with m thresholds, m+1 quantization intervals are generated. The input signal after clipping by a QC scheme with m thresholds can be expressed as:

$$\hat{x}(t) = \begin{cases} \tilde{x}(t), & |\tilde{x}(t)| \leq A_1 \\ \frac{A_1+A_2}{2} e^{j\phi(t)}, & A_1 < |\tilde{x}(t)| \leq A_2 \\ \ldots & \ldots \\ \frac{A_{m-1}+A_m}{2} e^{j\phi(t)}, & A_{m-1} < |\tilde{x}(t)| \leq A_m \\ A_m e^{j\phi(t)}, & |\tilde{x}(t)| > A_m \end{cases} \tag{9}$$

where ˆ$x(t)$ denotes the quantified signal, ˜$x(t)$ the time domain signal achieved from the SLM operation, and $\varphi(t)$ phase of input signal. Same with the conventional clipping scheme, the quantization clipping operation will only modify the amplitude of the input signal and remain the phase information unchanged after this processing.

For fair comparison, the minimum threshold A_1 of the QC scheme is set equal to the threshold of the conventional clipping scheme, that is, $A_1 = A$. The proposed SLM and quantization clipping scheme is not as good as the SLM-clipping scheme in reducing the system's PAPR, but the clipping noise caused by the nonlinear operation is much smaller. Let $x(t)$,$x\hat{}(t)$ represent the input and clipping processed signal respectively, and then the clipping noise caused by the nonlinear operation can be expressed as:

$$n(t)= x(t) - x\hat{}(t) \tag{10}$$

For conventional clipping scheme, the clipping noises $n_{clip}(t)$ can be written as:

$$n_{\text{clip}}(t) = \begin{cases} (|x(t)| - A)e^{j\phi(t)}, & |x(t)| > A \\ 0, & \text{else} \end{cases} \tag{11}$$

In order to get a better PAPR performance, the threshold A is always set to a small level which may results in serious degradation in BER performance. For our modified quantization clipping, the expression of clipping noise is:

$$n_{\text{QC}}(t) = \begin{cases} (|x(t)| - A_0)e^{j\phi(t)}, & A_0 \leq |x(t)| < A_1 \\ (|x(t)| - \frac{A_1+A_2}{2})e^{j\phi(t)}, & A_1 \leq |x(t)| < A_2 \\ \ldots & \ldots \\ (|x(t)| - \frac{A_{m-1}+A_m}{2})e^{j\phi(t)}, & A_{m-1} \leq |x(t)| < A_m \\ (|x(t)| - A_m)e^{j\phi(t)}, & |x(t)| \geq A_m \end{cases} \tag{12}$$

In the modified scheme, we set the lowest threshold value A_1 equal to the threshold A in traditional clipping scheme for comparison, and other thresholds satisfy $A = A_1 < A_2 <\cdots < A_m$ as mentioned in (9). After comparing equation (11) to (12), we can easily find that the clipping noise $n_{QC}(t)$ caused by the QC method is much smaller than the traditional one $n_{clip}(t)$. At the receiving end, assume that the received signal is $y(t)$ and can be expressed as:

$$y(t) = x\hat{}(t) + n_w(t) = x(t) - n(t) + n_w(t) \tag{13}$$

where$n_w(t)$ represents the Gaussian white noise. The frequency domain signal can be obtained by FFT operation on the received signal $y(t)$:

$$Y = \text{FFT}[y(t)] = \text{FFT}[x(t) - n(t) + n_w(t)] \tag{14}$$

Equation (14) implies that the smaller the clipping noise $n(t)$, the better the signal recovered. Based on the conclusion that $n_{QC}(t)$ is smaller than $n_{clipping}(t)$, it is easy to know the new QC scheme will have a better BER performance than the conventional clipping scheme.

CONCLUSION

In this paper we are discussed about the reduction techniques of PAPR. After brief study about the joint algorithms we observed that the better performance in OFDM system where as BER rate is increasing. And several techniques are proposed but for every technique there are several disadvantages are there. Our intention is to reduce the PAPR and complexity of the OFDM system.

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