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Comparative Analysis between A-law & μ-law Companding Technique for PAPR Reduction in OFDM

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Abstract: In this paper, one of the categories of PAPR Reduction Technique i.e. Signal distortion techniques are being simulated & discussed using Matlab/Simulink Software version 7.14. Then comparison of A-law companding technique and μ-law companding technique which comes under above mentioned scheme is being discussed. Here companding techniques are applied in the OFDM system to minimize the PAPR and effects of value "A" & "μ" are discussed.

Keywords: Additive White Gaussian Noise (AWGN), Inverse Fast Fourier Transform (IFFT), Orthogonal Frequency Division Multiplexing (OFDM),Peak to Average Power Ratio (PAPR), Quadrature Phase Shift Keying (QPSK), Q-point.

I. INTRODUCTION

The demand of high data rate services has been increasing very rapidly and there is no slowdown in sight. We know that the data transmission includes both wired and wireless medium. Often, these services require very reliable data transmission over very harsh environment. Most of these transmission systems experience much degradation such as large attenuation, noise, multipath, interference, time variance, nonlinearities and must meet the finite constraints like power limitation and cost factor.

One physical layer technique that has gained a lot of popularities due to its robustness in dealing with these impairments is multi-carrier modulation technique. In multi-carrier modulation, the most commonly used technique is Orthogonal Frequency Division Multiplexing (OFDM); it has recently become very popular in wireless communication.

According to the demand of advance communication field there should be high data rate in addition to both power efficiency and lower bit error rate. This demand of high data rate can be fulfilled by the single carrier modulation with compromising the tradeoff between the power efficiency and bit error rate [1]. Again in the presence of frequency selective fading environment, it is very difficult to achieve high data rate for this single carrier modulation with a lower bit error rate performance.

With considering an advance step towards the multi carrier modulation scheme it is possible to get high data rate in this multipath fading channel without degrading the bit error rate performance. To achieve better performance using multi carrier modulation we should make the subcarriers to be orthogonal to each other i.e. known as the Orthogonal Frequency Division Multiplexing (OFDM) technique [5].

But the great disadvantage of the OFDM technique is its high Peak to Average Power Ratio (PAPR). As we are using the linear power amplifier at the transmitter side so it's operating point will go to the saturation region due to the high PAPR which leads to in-band distortion and out-band radiation.

This can be avoided with increasing the dynamic range of power amplifier which leads to high cost and high consumption of power at the base station.

II. PAPR PROBLEM IN OFDM

It is defined as the ratio between the maximum power and the average power for the envelope of a baseband complex signal $\tilde{s}(t)$ i.e.

$$PAPR\{\tilde{s}(t)\} = \frac{\max |\tilde{s}(t)|^2}{E|\tilde{s}(t)|^2} \quad (1)$$

Also we can write this PAPR equation for the complex pass band signal $s(t)$ as

$$PAPR\{s(t)\} = \frac{\max |s(t)|^2}{E|s(t)|^2} \quad (2)$$

A. *Effect of High PAPR*

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The linear power amplifiers are being used in the transmitter so the Q-point must be in the linear region. Due to the high PAPR the Q-point moves to the saturation region hence the clipping of signal peaks takes place which generates in-band and out-of-band distortion.
So to keep the Q-point in the linear region the dynamic range of the power amplifier should be increased which again reduces its efficiency and enhances the cost. Hence a trade-off exists between nonlinearity and efficiency [2]. And also with the increasing of this dynamic range the cost of power amplifier increases. As a communication engineer our objective should be to reduce this PAPR.

III. A-LAW COMPANDING TECHNIQUE

We know that in the companding technique, the compression of OFDM signals at the transmitter and expansion at the receiver [1]. In this companding method, the compressor characteristic is piecewise, made up of a linear segment for low level inputs and a logarithmic segment for high level inputs. Fig 1 shows the A-law compressor characteristics for different values of A.

Corresponding to A=1, we observe that the characteristic is linear (no compression) which corresponds to a uniform quantization. A-law has mid riser at the origin. Hence it contains non-zero value. The practically used value of “A” is 87.6. The A-law companding is used for PCM telephone systems.

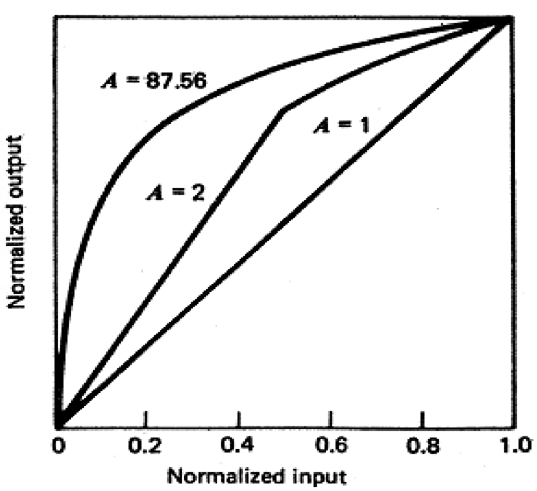

Fig. 1. A-law Compressor Characteristics

The linear segment of the characteristic is for low level inputs whereas the logarithmic segment is for high levelinputs. It is mathematically expressed as,

$$y(x) = \begin{cases} y_{max} \dfrac{A\frac{|x|}{x_{max}}}{(1+A)} sgn(x) & ;\ 0 < \dfrac{|x|}{x_{max}} \le \dfrac{1}{A} \\ y_{max} \dfrac{\left[1+\ln\left(A\frac{|x|}{x_{max}}\right)\right]}{(1+\ln A)} sgn(x) & ;\ \dfrac{1}{A} < \dfrac{|x|}{x_{max}} \le 1 \end{cases} \quad (3)$$

Where

- x=input signal.
- y=output signal.
- Sgn(x) =sign of the input (+ or -).
- $|x|$=absolute value (magnitude of x).
- A=87.6 (defined by CCITT).

This A-law companding technique is used in Europe, Asia, Russia, Africa, China, etc.

IV. MU-LAW COMPANDING TECHNIQUE

In the μ-law companding, the compressor characteristic is piecewise, made up of a linear segment for low level inputs and a logarithmic segment for high level inputs. Fig 2 shows the μ-law compressor characteristics for different values of μ. Higher the value of ‘μ’, more is the compression.

Corresponding to μ=0, we observe that the characteristic is linear (no compression) which corresponds to a uniform quantization. μ-law has mid tread at the origin. Hence it contains zero value. The practically used value of “μ” is 255. The μ-law companding is used for speech & music signals.
This μ-law companding technique is used in United States (U.S.), Canada, Japan, etc. The early BELL digital transmission system uses a 7-bit PCM Coder with μ=100. But the most recent digital transmission system uses a 8-bit PCM code with the value of μ=255.

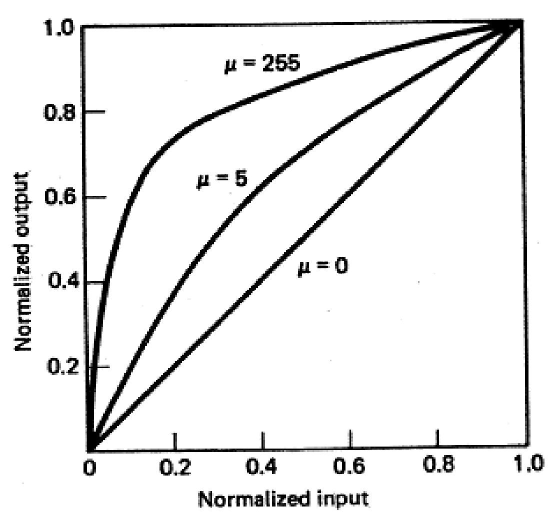

Fig. 2.μ-law Compressor Characteristics

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It is mathematically expressed as,

$$\frac{C|x|}{x_{max}} = \frac{\ln[1+\frac{\mu|x|}{x_{max}}]}{\ln[1+\mu]} \;;0 \leq \frac{|x|}{x_{max}} \leq 1 \quad (4)$$

Where

- x= Amplitude of the input signal at a particular instant of time.
- x_{max} =Maximum uncompressed analog input amplitude
- $C|x|$=Compressed output amplitude.
- $|x|$=Absolute value (magnitude of x).

μ=Parameter used to define the amount of compression (unit less) & standard value taken is 255.

V. SIMULINK MODELS

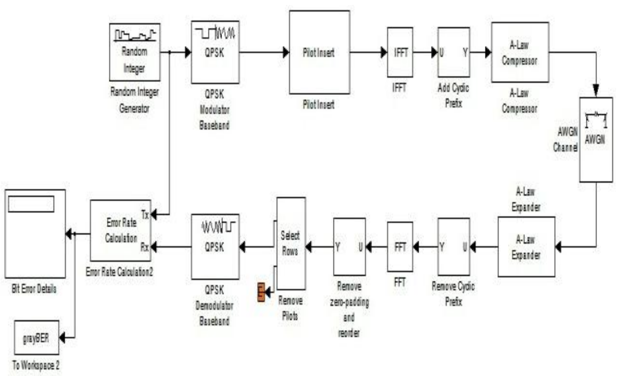

Fig. 3.Simulink model of OFDM system using A-law Companding technique

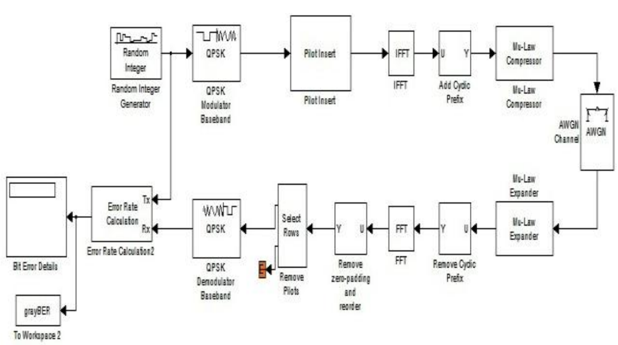

Fig. 4.Simulink model of OFDM system using μ-law Companding technique

Here, the Simulink models are designed in Fig 3 and Fig 4 for both A-law and μ-law Companding Techniques.

The various blocks are used for designing the models which are listed below:

- Random Integer Generator
- QPSK Modulator Baseband
- Pilot Insertion
- IFFT
- Adding Cyclic Prefix
- A-law Compressor
- AWGN Channel
- A-law Expander
- Pilot Removal
- Removing Cyclic Prefix
- FFT
- QPSK Demodulator Baseband
- Error Rate Calculation

A. Random Integer Generator:

The Random Integer Generator block generates uniformly distributed random integers in the range [0, M-1], where M is the M-ary number defined in the dialog box.

The M-ary number can be either a scalar or a vector. If it is a scalar, then all output random variables are independent and identically distributed (i.i.d.). If the M-ary number is a vector, then its length must equal the length of the Initial seed; in this case each output has its own output range.If the Initial seed parameter is a constant, then the resulting noise is repeatable.

B. QPSK Modulator Baseband:

The QPSK Modulator Baseband block modulates using the quaternary phase shift keying method. The output is a baseband representation of the modulated signal.

If you set the Input type parameter to Integer, then valid input values are 0, 1, 2, and 3. When you set Constellation ordering to Binary for input m the output symbol is $\exp(j\theta + j\pi m/2)$ where θ represents the Phase offset parameter (see the following figure for Gray constellation ordering). In this case, the block accepts a scalar or column vector signal.

If you set the Input type parameter to Bit, then the input contains pairs of binary values. For this configuration, the block accepts column vectors with even lengths. When you set the Phase offset parameter to $\frac{\pi}{4}$, then the block uses one of the signal constellations in the following figure, depending on whether you set the Constellation ordering parameter to Binary or Gray.This is shown in Fig. 5.

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C. Pilot Insertion:

The Multiport Selector block extracts multiple subsets of rows or columns from M-by-N input matrix u, and propagates each new sub matrix to a distinct output port. The block treats an un oriented length-M vector input as an M-by-1 matrix.

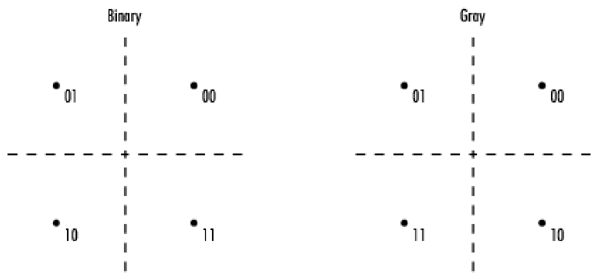

Fig. 5. Constellation Diagram of QPSK for Binary & Gray Coding

D. IFFT:

The IFFT block computes the inverse fast Fourier transform (IFFT) of each row of a sample-based 1-by-P input vector, or across the first dimension (P) of an N-D input array. When you specify an FFT length not equal to the length of the input vector, (or first dimension of the input array), the block implements zero padding or modulo-M, (FFT length) data wrapping. This occurs before the IFFT operation.

E. Adding Cyclic Prefix:

The Selector block generates as output selected or reordered elements of an input vector, matrix, or multidimensional signal. A Selector block accepts vector, matrix, or multidimensional signals as input. The parameter dialog box and the block's appearance change to reflect the number of dimensions of the input. Based on the value you enter for the Number of input dimensions parameter, a table of indexing settings is displayed. Each row of the table corresponds to one of the input dimensions in Number of input dimensions. For each dimension, you define the elements of the signal to work with. Specify a vector signal as a 1-D signal and a matrix signal as a 2-D signal. When you configure the Selector block for multidimensional signal operations, the block icon changes.

F. A-law Compressor:

The A-Law Compressor block implements an A-law compressor for the input signal.

G. AWGN Channel:

The AWGN Channel block adds white Gaussian noise to a real or complex input signal. When the input signal is real, this block adds real Gaussian noise and produces a real output signal. When the input signal is complex, this block adds complex Gaussian noise and produces a complex output signal. This block inherits its sample time from the input signal. This block accepts a scalar-valued, vector, or matrix input signal with a data type of type single or double. The output signal inherits port data types from the signals that drive the block.

H. A-law Expander:

The A-Law Expander block recovers data that the A-Law Compressor block compressed.

I. Pilot Removal:

The Multiport Selector block extracts multiple subsets of rows or columns from M-by-N input matrix u, and propagates each new sub matrix to a distinct output port. The block treats an un oriented length-M vector input as an M-by-1 matrix. When you set the Select parameter to Rows, the block uses the one-dimensional indices you specify to select matrix rows, and all elements on the chosen rows are included. When you set the Select parameter to Columns, the block uses the one-dimensional indices you specify to select matrix columns, and all elements on the chosen columns are included. A given input row or column can appear any number of times in any of the outputs, or not at all. When an index references a nonexistent row or column of the input, the block reacts with the action you specify using the Invalid index parameter

J. Removing Cyclic Prefix:

The Selector block generates as output selected or reordered elements of an input vector, matrix, or multidimensional signal. A Selector block accepts vector, matrix, or multidimensional signals as input. The parameter dialog box and the block's appearance change to reflect the number of dimensions of the input.

K. FFT:

When you specify an FFT length not equal to the length of the input vector, (or first dimension of the input array), the block implements zero padding or modulo-M, (FFT length) data wrapping. This occurs before the IFFT operation.

L. QPSK Demodulator Baseband:

The QPSK Demodulator Baseband block demodulates a signal that was modulated using the quaternary phase shift keying method. The input is a baseband representation of the modulated signal. The input must be a complex signal. This block accepts a scalar or column vector input signal.

M. Error Rate Calculation:

The Error Rate Calculation block compares input data from a transmitter with input data from a receiver. It calculates the error rate as a running statistic, by dividing the total number of unequal pairs of data elements by the total number of input data elements from one source.

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Use this block to compute either symbol or bit error rate, because it does not consider the magnitude of the difference between input data elements. If the inputs are bits, then the block computes the bit error rate. If the inputs are symbols, then it computes the symbol error rate. The same case is there with respect to μ-law companding technique.

VI. SIMULATION RESULTS

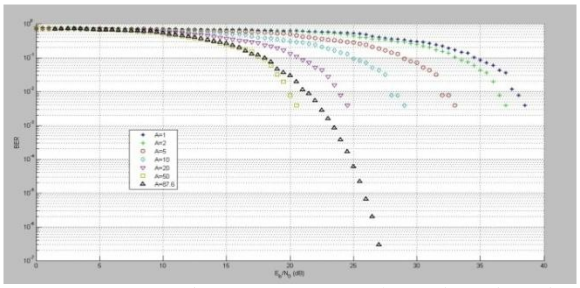

Fig. 6.Response of OFDM system to various values of "A" for A-law Compressor

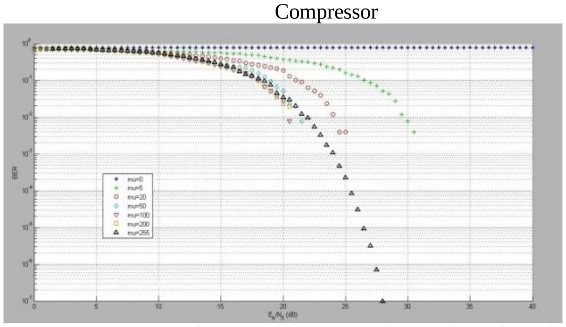

Fig. 7.Response of OFDM system to various values of "μ" for μ-law Compressor

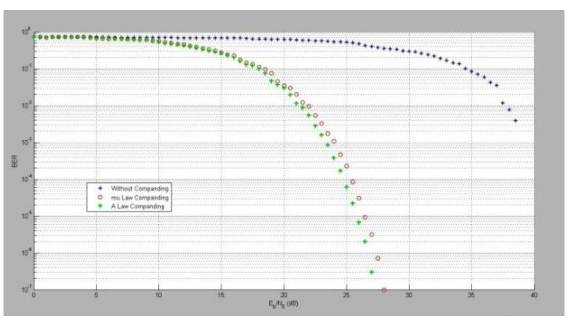

Fig. 8.Comparison between "A" law &"μ" law Companding Techniques

VII. CONCLUSION

The performance of the OFDM system is simulated by using A-law Companding as well as using μ-law Companding, it is observed that as the value of "A" and "μ" increases shown in Fig 6 & Fig 7, there is a significant reduction in the PAPR value. And it is clear that by choosing the appropriate value of A and μ, we can get the desired PAPR value. Also we found very much important observation that to achieve the Bit Error Rate (BER) of 10^{-3} shown in Fig 8, A-law technique requires approximately 0.7 dB less Signal to Noise Ratio (SNR) value as compared to the μ-law technique. So, after comparison we can say that A-law Companding Technique is somewhat better than μ-law Companding Technique.

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BIOGRAPHY

Vishwajit N. Sonawane received his Bachelor's degree in Electronics and Telecommunications engineering from PUNE University, Maharashtra, India, in 2010. Currently pursuing the M.Tech degree & working as Teaching Assistant in Electronics and Telecommunications engineering from Dr. Babasaheb Ambedkar Technological University, Lonere,Maharashtra, India. His Research area includes Wireless Communication, Mobile Communication, Information & Coding Theory, etc.

Sanjay V. Khobragade has been working as Assistant Professor in Dr. B.A. Technological University Lonere, Maharashtra, India from last 13 years. He is graduated from Nagpur University in 1996 and post graduated in Electronics Engineering from Mumbai University in 2008 and pursuing PhD form Rayalaseema University Kurnool, Andhra Pradesh. He has been involved in teaching a Microwave, Antenna & Wave Propagation and Electromagnetic Field. He has received Young Scientist Award in URSI 2004 in Pisa Italy, and Consolation Prize for best paper in ICMARS Jodhpur, 2008 and best Technical teacher award by ISTE sponsored by Maharashtra and Goa in 2010. He has around 70 papers at national and International conferences in his credit.