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Comparison of OFDM System using DWT, DCT based Image Processing

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Abstract— **Peak Signal-to-Noise Ratio (PSNR) is a serious issue in OFDM systems. Several techniques have been proposed by researchers to maintain PSNR. In this paper, the Peak Signal-to-Noise Ratio (PSNR) performance of conventional discrete cosine transform (DCT)-OFDM system is compared with discrete wavelet transform (DWT)-OFDM system in a Gaussian Noise environment along with error correcting codes like Reed Solomon, LDPC**

Keywords- **DCT, DWT, OFDM, Reed Solomon, LDPC.**

I. Introduction

Wireless communication was initially realized for analog sector now-a-days it is mainly applied in digital sector. In wireless communication multiple sub-carriers are employed to make the transmission process easier. OFDM is a type of Wireless communication which utilized multi-channel modulation structure, using Frequency Division Multiplexing (FDM) of orthogonal sub-carriers, each of them modulating a low bit-rate digital stream.

The concept of OFDM strikes during the mid-60, s when chang printed his report on the functionality of band limited signals for multichannel transmission [1]. He reveals a theory for the transmission of messages at the same time via a linear band limited channel without (ISI) intersymbol interference and (ICI) inter-channel. Saltzberg carried out a research of the functionality [2], where he observed that "the strategy of designing an efficient parallel system should concentrate more on reducing crosstalk between adjacent channels than on perfecting the individual channels themselves, since the distortions due to crosstalk tend to dominate". This is an important conclusion, which has proven a major improvement in the digital baseband processing that emerged a few years later.

A major contribution to OFDM was presented in 1971 by Weinstein and Ebert [3], who used the discrete Fourier transform (DFT) to perform baseband modulation and demodulation.

Another important contribution was due to Peled and Ruiz in 1980 [4], who introduced the cyclic prefix (CP) or cyclic extension, solving the orthogonality problem. Instead of using an empty guard space, they filled the guard space with a cyclic extension of the OFDM symbol.

OFDM is currently used in the European digital audio broadcasting (DAB) standard [5]. Several dab systems proposed for north America are also based on OFDM [6], and its applicability to digital to broadcasting is currently being investigated.

Now a days satellite, military and cellular systems are now commercially utilized by ever more demanding customers, who required smooth conversation from their residence to their workplace, to their vehicle, or even for outdoor activities. With this enhanced desire comes an ever-growing need to transmit data quickly, wirelessly and precisely. To cope with this requirement, communications manufacturer have merged systems ideal for high rate transmission with forward error correction (FEC) techniques.

Orthogonal Frequency Division Multiplexing (OFDM) is mainly used for high data rates necessary for data intensive applications that must now become routine. The OFDM schemes is able to functioning without a traditional equalizer, when communicating over depressive transmission media, like wireless channels, while easily maintaining the frequency and time domain channel quality variances of the wireless channel.

A. *OFDM system model*

Modulation

Modulation is a way by which the signal waves are transmitted over the communication channel so as to diminish the consequences of noise. This procedure is helpful in recreating the original data after demodulation. In an OFDM scheme, the information is broken down into small packets of information

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which are positioned orthogonal to one another. This is accomplished by means of modulating the data through a suitable modulation strategy. Following this, IFFT is conducted on the modulated signal that is additionally refined by passing via a parallel–to–serial converter. To avoid ISI, cyclic prefix is applied to the signal.

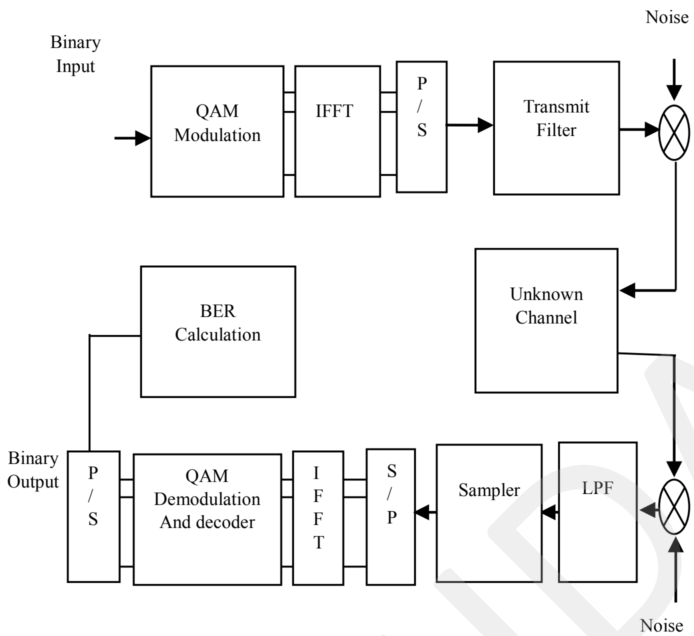

Fig.1: block diagram of OFDM

Communication Channel

This is the channel by which the information is moved. Noise Occurrence in this medium influences the signal and leads to distortion in its information content.

Demodulation

The procedure by the means of which the original information is retrieve from the modulated signal, which is obtained at the receiver end after the transmission. In this procedure, the received information is primary feed a low pass filter in order to remove the cyclic prefix. Then the resultant signal is made to pass through a serial to parallel converter so as to perform FFT on the signal. A demodulator is utilized, to obtain back the initial signal.

The bit error rate (BER) and the signal–to–noise ratio (SNR) is measured by taking into account the unmodulated signal data and the data at the receiving end.

B. *Low-density parity-check codes*

The essential idea of forward error correction is to deliberately introduce redundancy into a digital message so that the message can be correctly inferred at the receiver, even when some of the symbols are corrupted during transmission or storage. More specifically, a $q-ary$ $[n,k,d]$ block error correction code with rate $R = k/n$, maps a message of k symbols into a code word of $n > k$ symbols where each symbol is one of q possible elements. The decoder receives a length n vector, which is not necessarily a codeword, and uses the structure of the code to determine which message was sent. The gains to data reliability afforded by employing error correction can be used to reduce the required transmission power or bandwidth, or increase data storage efficiency.

The Hamming distance is the distance between two codewords is the number of symbols in which they differ. The minimal distance of the code, d, is the most basic Hamming distance between any pair of codewords in the code and is one measure of the error correction capability of the code. In general, for a code with minimum distance d, t bit errors can always be corrected by choosing the closest codeword, in Hamming distance, to the received vector whenever

$$t \leq [(d-1)/2] \quad (1)$$

Where |x|is the largest integer that is at most,x. To illustrate, a simple linear binary code, with elements from the binary Galois field, is defined to have the following structure:

$$c = c1\, c2\, c3\, c4\, c5\, c6, \quad (2)$$

Where each symbol ci is either 0 or 1, and the codeword, c, is constrained by three parity check equations:

$$c1 \oplus c2 \oplus c4 = 0$$
$$c2 \oplus c3 \oplus c5 = 0$$
$$c1 \oplus c2 \oplus c3 \oplus c6 = 0$$
$$(3)$$

The notation ⊕ represents modulo-2 addition, which is equal to 1 if the ordinary sum is odd and 0 if the ordinary sum is even. The parity-check equations can be re-written in matrix form:

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$$\underbrace{\begin{bmatrix}1 & 1 & 0 & 1 & 0 & 0\\0 & 1 & 1 & 0 & 1 & 0\\1 & 1 & 1 & 0 & 0 & 1\end{bmatrix}}_{H}\begin{bmatrix}c1\\c2\\c3\\c4\\c5\\c6\end{bmatrix}=\begin{bmatrix}0\\0\\0\end{bmatrix} \quad (4)$$

Where the parity-check matrix, H, represents the parity-check equations which define the code. Thus a vector r = [r1 r2 r3 r4 r5 r6] is a codeword if and only if it satisfies the constraint.

$$Hr^T = 0. \quad (5)$$

To generate the codeword for a given message, the code constraints can be rewritten in the form

$$c4 = c1 \oplus c2$$
$$c5 = c2 \oplus c3$$
$$c6 = c1 \oplus c2 \oplus c3 \quad (6)$$

Where bits c1, c2, and c3 contain the 3-bit message, and the parity-check bits c4, c5 and c6 are a function of this message. Thus, for example, the message 110 produces parity-check bits c4 = 1 ⊕ 1 = 0, c5 = 1 ⊕ 0 = 1 and c6 = 1 ⊕ 1 ⊕ 0 = 0, and hence the codeword 110010. As the code is linear, matrix notation can again be used,

$$[c1 \quad c2 \quad c3 \quad c4 \quad c5 \quad c6] = [c1 \quad c2 \quad c3]\underbrace{\begin{bmatrix}1 & 0 & 0 & 1 & 0 & 1\\0 & 1 & 0 & 1 & 1 & 1\\0 & 0 & 1 & 0 & 1 & 1\end{bmatrix}}_{G} \quad (7)$$

Where the generator matrix of the code, G, represents a basis for the one-to-one mapping of messages onto codewords. An error correction code can be described by more than one parity-check matrix or generator matrix. A matrix H is a valid parity-check matrix for a code provided that (5) holds for all codewords in the code. Likewise two matrices generate the same code if they map every message to the same codeword. Two parity-check matrices for the same code need not even have the same number of rows; however the rank over GF (q) of both must be the same, since the number of message symbols, k, in a $q-ary$ code is

$$k = n - rank_q(H), \quad (8)$$

Where $rank_q$(H) is the number of rows in H which are linearly dependent over GF (q).

A parity-check matrix is regular if each code symbol is contained in a fixed number, w_c of parity checks and each parity-check equation contains a fixed number, w_r of codeword symbols. If a code is described by a regular parity-check matrix it is called a (w_c, w_r)-regular code otherwise it is an irregular code. A regular parity-check matrix for the binary code of (4) with, $w_c = 2\ w_r = 3$ and $rank_2(H) = 3$ is:

$$H = \begin{bmatrix}1 & 1 & 0 & 1 & 0 & 0\\0 & 1 & 1 & 0 & 1 & 0\\1 & 0 & 0 & 0 & 1 & 1\\0 & 0 & 1 & 1 & 0 & 1\end{bmatrix} \quad (9)$$

An LDPC code is simply a block code with a parity-check matrix which is sparse, that is, the majority of entries must be zero. What sets LDPC codes apart from traditional codes is the way in which they are decoded which in turn has implications for which sparse parity-check matrices make good LDPC codes.

C. Reed-Solomon codes

RS codes are commonly defined as follows.

Definition 2.1: A Reed-Solomon (RS) code of length N and minimum Hamming distance d_{Hm} is a set of vectors, whose components are the values of a polynomial $C(x) = x^l \cdot C'(x)$ of degree $\{C'(x)\} \leq K-1 = N - d_{Hm}$, at positions $z^k$with z being an element of order N from an arbitrary number field.

$$c = (c0, \ldots, c_{N-1}), c_i = C(x = z_i) \quad (10)$$

ω_{Hm} And, d_{Hm}, the minimum Hamming weight and distance, respectively, are known to be

$$\omega_{Hm} = min\|c\|_0 = d_{Hm} = N-(K-1) = N-K+1 \quad (11)$$

Since the samples are chosen to be powers of an element of order N, i.e., z^k where z would be $e^{j2\pi/N}$in the complex case, the equivalent description is known to be

$$c_i = z^{il}\sum_{k=0}^{N-1} C_k z^{ik}, \quad i = 0, \ldots, N-1 \quad (12)$$

With C_{k+1}modulo N = 0 for K ≤ k ≤ N − 1. For, $l = 0$ (12) is a DFT, which can as well be formulated as

$$(c0, c1, \ldots cN-1) = 1/\sqrt{N} \cdot (C0, C1, \ldots, CN-1) \cdot W \quad (13)$$

With matrix elements $W_{ij} = z^{ij}$. We introduced the factor 1/√N to make it a unitary transform. Usually, the minimum Hamming distance is achieved by inspecting the number of linearly independent

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columns of the parity-check matrix. The generator matrix, however, can be used as well. Then, the minimum Hamming weight (distance) of N −K +1 = M +1 follows from K ×K non-zero minors from arbitrary columns of the K × N generator matrix. In our case, we extract the generator matrix as (cyclically) consecutive rows of the DFT matrix W.
Theorem 2.1: Any minor |F| of (any) size K × K of an N × N Fourier (DFT) matrix W with components, $W_{k,i} = z^{ik}$, $z = e^{\pm j2\pi/N}$ and cyclically adjacent rows (or columns) is nonzero. The non-singularity of the considered sub-matrices ensures that at most K − 1 zeros can be achieved, leaving at least N − K + 1 non-zero values in time domain, which is then the minimum Hamming weight (distance).
OFDM is known to consist of an IFFT and cyclic prefix (CP) extension at the transmitter and CP elimination and FFT at the receiver. The CP makes the channel convolution to appear as cyclic and the IFFT/FFT pair finally diagonalizes the channel. Practically, for band separation purposes or, in case of adaptive bit-loading, at low signal-to-noise ratios, usually, some consecutive carriers are left unused, thereby realizing an analog RS code.

D. Discrete Cosine Transform

Similar to other transforms, the Discrete Cosine Transform tries to decorrelate the image information. Following decorrelation each transform coefficient is usually encoded separately without losing compression effectiveness. This portion talks about the DCT and some of its crucial attributes.
The DCT of a 1-D sequence of length N is

$$C(u) = \alpha(u) \sum_{x=0}^{N-1} f(x) cos\left[\frac{\pi(2x+1)u}{2N}\right] \qquad (14)$$

For u = 0, 1, 2… N-1. Likewise, the inverse transformation is described as

$$f(x) = \sum_{x=0}^{N-1} \alpha(u) C(u) cos\left[\frac{\pi(2x+1)u}{2N}\right] \qquad (15)$$

For x = 0, 1, 2… N −1. In both equations (14) and (15) α (u) is defined as

$$\alpha(u) = \begin{cases} \sqrt{\frac{1}{N}} & for\ u = 0 \\ \sqrt{\frac{2}{N}} & for\ u \neq 0 \end{cases} \qquad (16)$$

It is clear from (1) that for $u = 0$ $C(u = 0) = \sum_{x=0}^{N-1} f(x)$. Therefore, the primary transform coefficient is the average worth of the sample sequence.

The Two-Dimensional DCT is the expansion of the concepts introduced in the above section to a two-dimensional space. The 2-D DCT is a direct expansion of the 1-D case and is given by

$$C(u,v) = \alpha(u)\alpha(v) \sum_{x=0}^{N-1}\sum_{y=0}^{N-1} f(x,y) cos\left[\frac{\pi(2x+1)u}{2N}\right] cos\left[\frac{\pi(2y+1)v}{2N}\right] \qquad (17)$$

For u, v = 0, 1, 2… N −1 and α (u) and α (v) are defined in (16). The inverse transform is defined as

$$f(x,y) = \sum_{x=0}^{N-1}\sum_{y=0}^{N-1} \alpha(u)\alpha(v) C(u,v) cos\left[\frac{\pi(2x+1)u}{2N}\right] cos\left[\frac{\pi(2y+1)v}{2N}\right] \qquad (18)$$

For x, y = 0, 1, 2… N −1. The 2-D functioning can be produced by multiplying the horizontally oriented 1-D with vertically oriented set of the same functions.

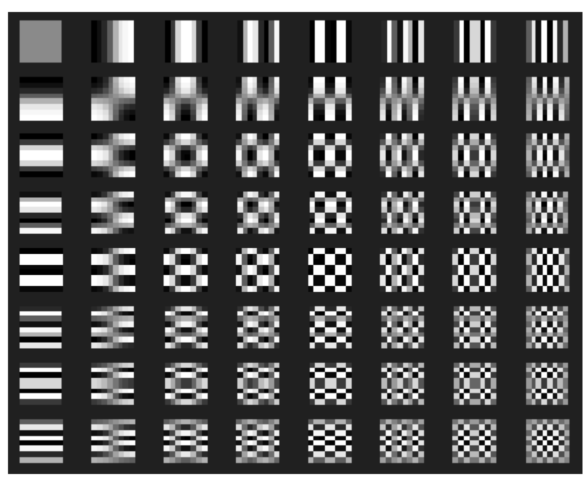

Figure 2. Two dimensional DCT basis functions (N = 8). Neutral gray represents zero, white represents positive amplitudes, and black represents negative amplitude [10].

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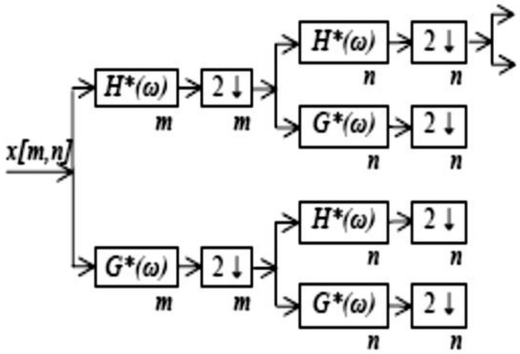
E. Discrete Wavelet Transform (DWT)

The fundamental concept of DWT for one-dimensional signals is briefly explained as fallows.
A signal is break down into two fragments, generally the low frequency part and the high frequency. This break down is named as decomposition. The edge aspects of the signal are generally enclosed to the high



frequencies part.
The signal is handed down through a number of high pass filters to evaluate the high frequencies, and then passed through a number of low pass filters to evaluate the low frequencies. Filters with cut-off frequencies are utilized to examine the signal at different resolutions. Let's guess that *x[n]* is the initial signal, having a frequency band of 0 to π rad/s. The signal *x[n]* is initially passed through a half band high pass filter *g[n]* and a low pass filter *h[n]*. Following the filtering, half of the trials can be eradicated based on the Nyquist's rule, as the signal now has the top frequency of $\pi/2$ radians rather than of π. The signal can consequently be subsampled by 2, merely by neglect every second sample. This comprises one level of decomposition and can arithmetically be stated as follows:

$$y_{high}[k] = \sum_n x[k]g[2k-n]$$
$$y_{low}[k] = \sum_n x[k]h[2k-n] \quad (19)$$

Where $y_{high}[k]$ and $y_{low}[k]$ are the yields of the highpass and low pass filters, correspondingly, fallowing the subsampling by 2. The above mentioned process can be continual for additional decomposition. The yields of the high pass and low pass filters are named as DWT coefficients. The original image can be reconstructed utilizing this DWT. The reconstructed method is known as the Inverse Discrete Wavelet Transform (IDWT).

The signals at every level are passed through the synthesis filters g' [n], and h' [n], and then added. The synthesis and analysis filters are alike to each other, except for a time reversal. So, the reconstruction formula becomes (for each layer)

$$x[n] = \sum_n y_{high}[k]g[2k-n] + y_{low}[k]h[2k-n] \quad (20)$$

The DWT and IDWT for a one-dimensional signal can be also described in the form of two channel tree structured filter banks. The DWT and IDWT for a two-dimensional image x [m, n] can be similarly defined by implementing DWT and IDWT for each dimension m and n separately $DWT_n[DWT_m[x[m,n]]$, which is shown in Figure 1.

Fig.3: DWT for two-dimensional images

An image can be decomposed into a pyramidal structure, which is shown in Figure 4, with various band information: low-low frequency band LL, low-high frequency band LH, high-low frequency band HL, high frequency band HH.

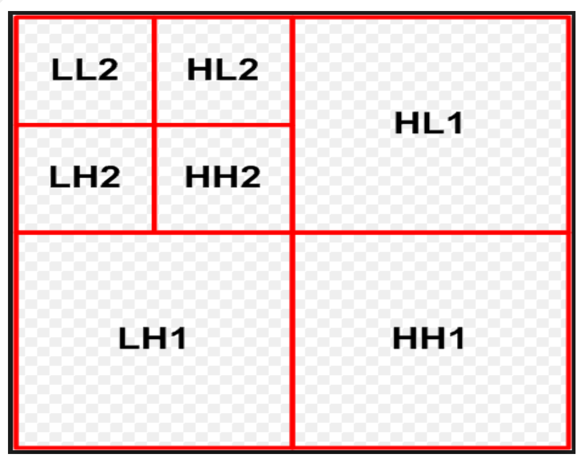


Fig.4: Pyramidal structure

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II. SIMULATION RESULTS

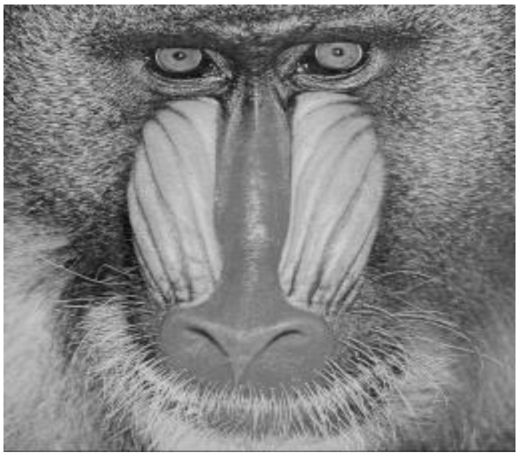

Figure 4: Input Image for DCT

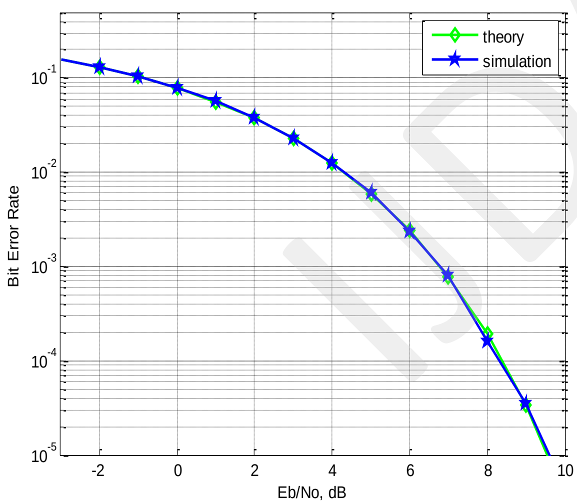

Figure 5: curve of Gaussian noise BER for DCT source coding with OFDM

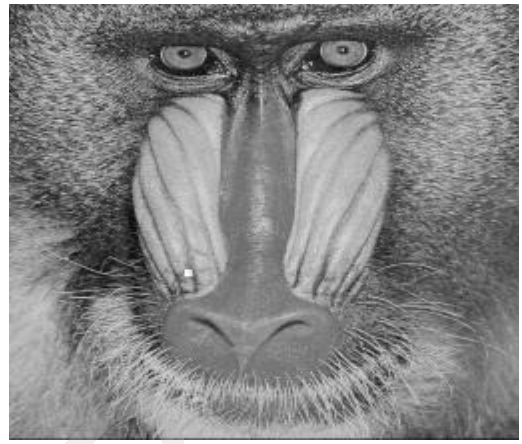

Figure 6: Output Image for DCT

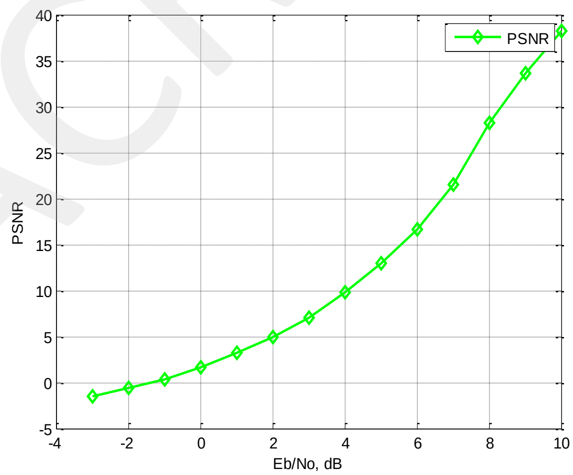

Figure 7: PSNR curve for DCT source coding with OFDM

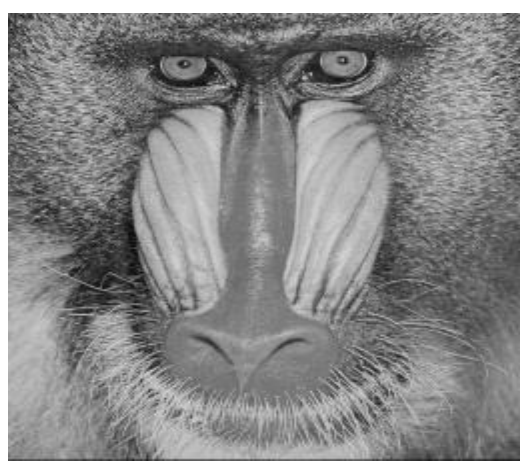

Figure 8: input Image for DWT

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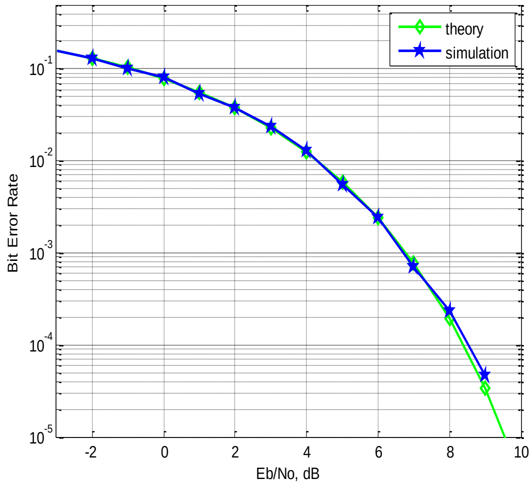

Figure 9: curve of Gaussian noise BER for wavelet source coding with OFDM

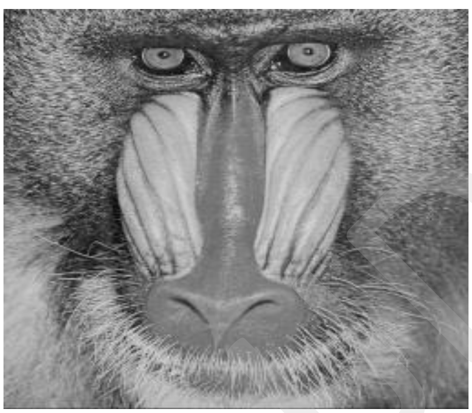

Figure 10: Output Image for DWT

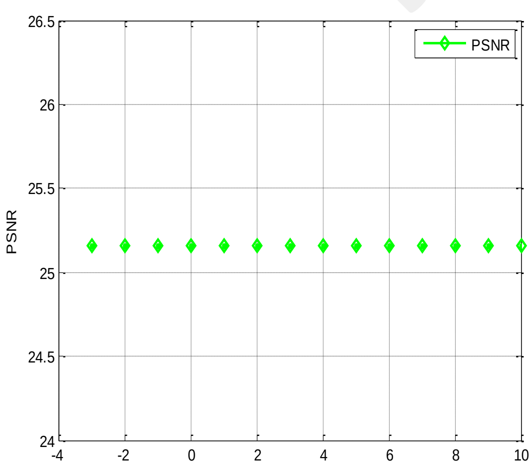

Figure 11: PSNR curve for Wavelet source coding with OFDM

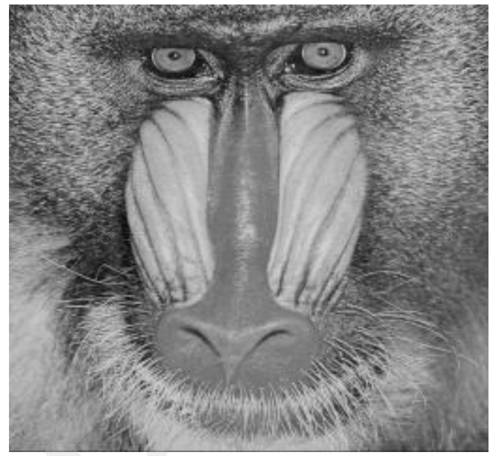

Figure 12: input image for FFT

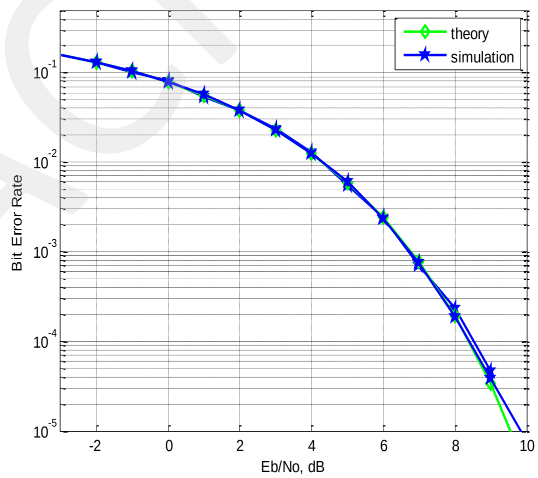

Figure 13: curve of Gaussian noise BER for wavelet source coding with OFDM

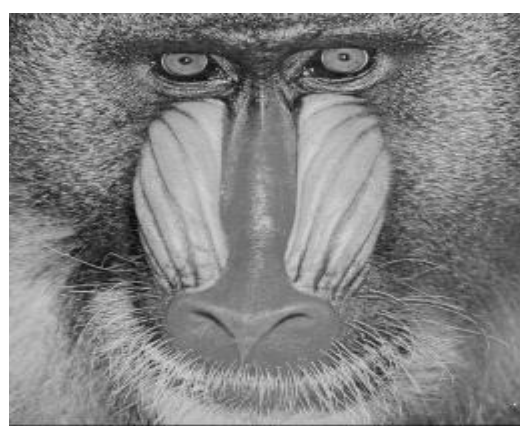

Figure 14: output image for FFT

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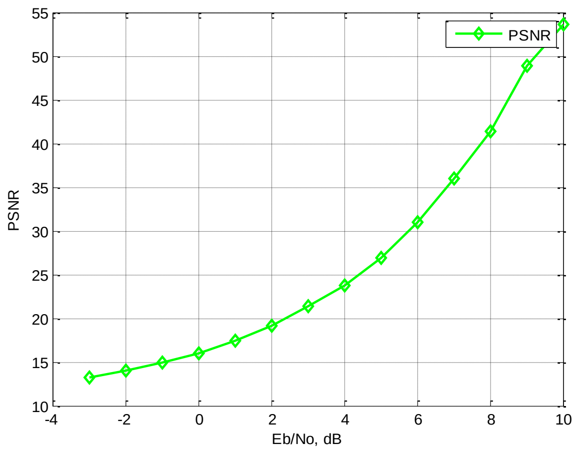

Figure 15: PSNR curve for FFT source coding with OFDM

In the bar graphs shown below notations are as

1. Reed Solomon
2. LDPC
3. DCT
4. DWT
5. FFT

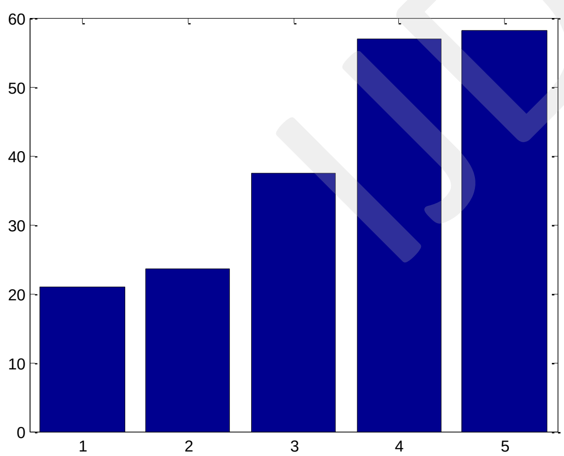

Figure 16: BAR chat for PSNR

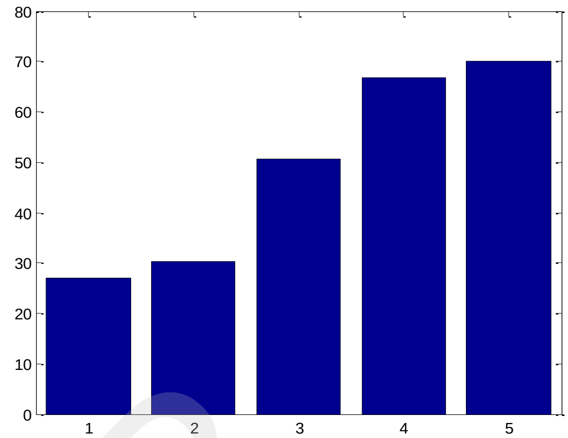

Figure 17: bar chart for WPSNR

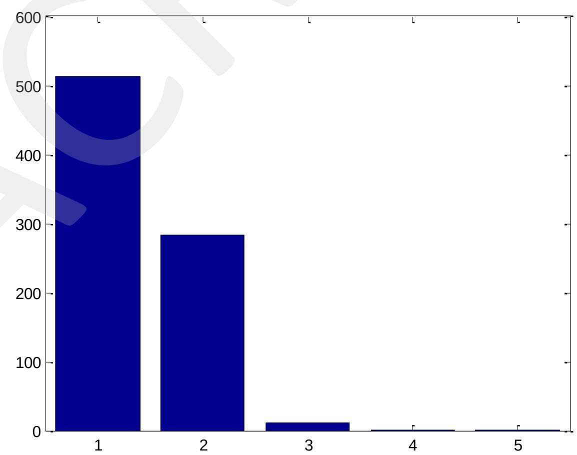

Figure 18: Bar chart for MSE

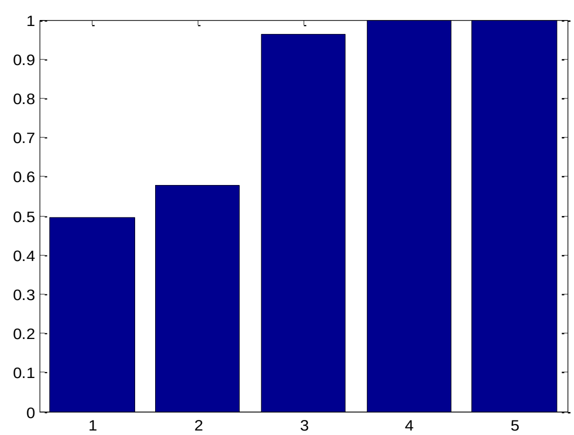

Figure 19: Bar chart for SSIM

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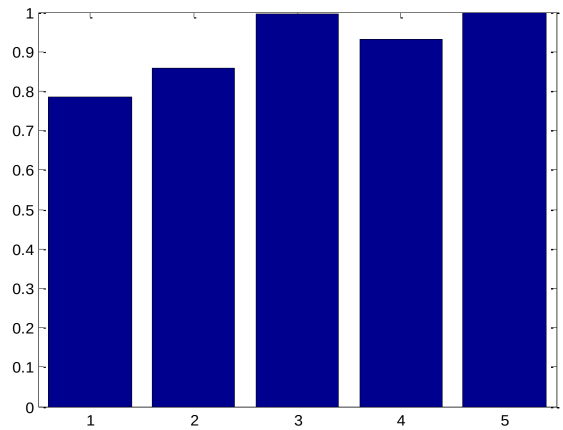

Figure 20: Bar chart for corr2:

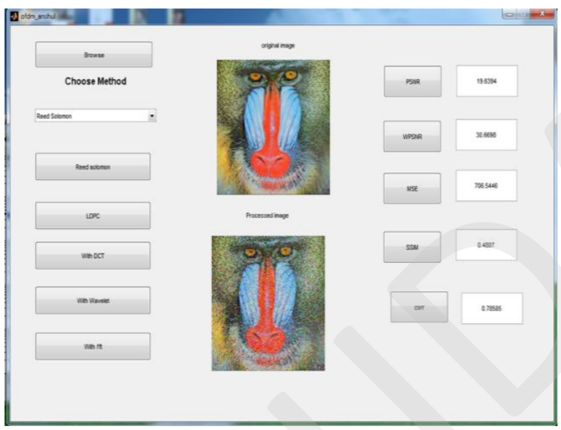

Figure 21: GUI of the recommended model

III. CONCLUSION

OFDM is a very attractive technique for multicarrier transmission and has become one of the standard choices for high speed data transmission over a communication channel. It has various advantages; but also has one major drawback: it has a very high PAPR.

In this project, the different properties of an OFDM System are analyzed along with DCT and DWT. The simulation results shows that the DWT gives better result than the DCT. We found a constant higher PSNR in DWT. It shows the greater efficiency.

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