Performance of PCCC Turbo Coded OFDM

Malathe Salah AL-Deen
University of Baghdad / Engineering college

Abstract

Orthogonal frequency division multiplexing (OFDM) has become a popular modulation method in high speed wireless communication. This paper gives the results of a simulation study on the performance of coded Orthogonal Frequency Division Multiplexing (OFDM) system using turbo code (parallel concatenated convolutional code PCCC) with BPSK,QPSK and QAM modulation over Rayleigh fading channel and AWGN channel. The simulation results show that improvement on the performance has been achieved by applying turbo coding to uncoded OFDM system. the simulated programs is carried out using MATLAB programming languages

الخلاصة

اصبح التقسيم المتعدد المتعامد للتردد طريقة شائعة للبث في مجال الاتصالات اللاسلكية عالية السرعة . هذا البحث اعطى نتائج دراسة المحاكات على اداء نظام التقسيم المتعدد المتعامد للتردد المرمز (CODED OFDM) باستخدام شفرات التيربو نوع (parallel concatenated convolutional code PCCC TURBO CODE) مع التضمين المرمز BPSK , QPSK وQAM فوق قناة Rayleigh fading وAWGN تبين نتائج المحاكات تحقق تحسن في الاداء من خلال تطبيق استخدام شفرات التيربو لنظام OFDM نسبتا لنظام OFDM غير مشفر. استخدمت لهذه المحاكات لغة برمجة ماتلاب MATLAB.

Introduction:

Orthogonal frequency division multiplexing (OFDM). Is an important broadband wireless communication scheme. Originally developed in the late 1950s and 1960s , is being used or considered in various wireless communication systems.(Peter Fertl,2009, Gregory E. Bottomley ,2006) In a wireless communication the signals that are send from a sender to a receiver can follow multiple paths with each its own characteristics (attenuation, delay, etc.). this is called multi- path propagation. This multi- path propagation of a wireless channel often introduces Inter Symbol Interference (ISI).(Jeroen Theeuwes,2004) Orthogonal frequency division multiplexing (OFDM) is a special case of multicarrier transmission, where a single datastream is transmitted over a number of lower rate subcarriers. OFDM is most preferred for high speed communication in multipath environment due to its immunity to ISI. OFDM avoids ISI problem by sending many low speed transmissions simultaneously. OFDM is presently used in a number of commercial wired and wireless applications. One of the wired applications is digital subscriber line (DSL). For wireless, OFDM is used for several digital video and digital audio broadcast applications(Alok Joshi, Davinder S. Saini,2010). Coded OFDM is a modified version of conventional OFDM where OFDM is combined with channel coding techniques, resulting in to higher data transmission rate or lower BER in wireless fading medium.(Alok Joshi, Davinder S. Saini,2010)In this paper the performance of COFDM is analyzed for using parallel concatenated convolutional code PCCC. The result is compared with uncoded OFDM system.

1- TURBO CODES:

Turbo codes are parallel concatenated codes (PCC's) whose performance in the additive white Gaussian noise (AWGN) channel is astonishingly good (Petri Komulainen

, Kari Pehkonen, 1998). Turbo Codes are a class of error correcting codes introduced in 1993 by Berrou that come closer to approaching Shannon's limit than any other class of error correcting codes. The Shannon's Channel Coding Theorem explains that exists a channel coding scheme which allows to achieve arbitrarily reliable

transmission as long as the code rate does not exceed the channel capacity(Alexandre Graell I Amat, 2004).

PCCC resulted from the combination of three ideas that were known to all in the coding community. The transforming of commonly used non-systematic convolutional codes into systematic convolutional codes, the utilization of soft input soft output decoding. Instead of using hard decisions, the decoder uses the probabilities of the received data to generate soft output which also contain information about the degree of certainty of the output bits, Encoders and decoders working on permuted versions of the same information. This is achieved by using an interleaver (M.K.GUPTA,VISHWAS SHARMA , 2009).

1.1 Turbo Encoding

The encoder for a turbo code is parallel concatenated convolutional code (M.K.GUPTA , VISHWAS SHARMA , 2009, C. Berrou, A. Glavieux, and P. Thitimajshima,1993, G. D. Forney, 1973, L.Bahl, J.Cocke, F.Jelinek, and J.Raviv, 1974, B.Balaji Naik, 2008). the block diagram of the encoder is shown in fig -1. The binary input data sequence is represented by $dk = (d1 , \ldots\ldots dN)$. The input sequence is passed into the input of a convolutional encoder. $ENC1$ and a coded bit stream, $x^{p}_{k_1}$ is generated. The data sequence is then interleaved. That is, the bits are loaded into a matrix and read out in a way so as to spread the positions of the input bits. The bits are often out in a pseudo-random manner. The interleaved date sequence is passed to a second convolutional encoder $ENC2$, and a second coded bit stream, $x^{p}_{k_2}$ is generated. The code sequence that is passed to the modulator for transmission is a multiplexed (and possibly punctured) stream consisting of systematic code bits x^{s}_{k} and parity bits from both the first encoder $x^{p}_{k_1}$ and the second encoder $x^{p}_{k_2}$.

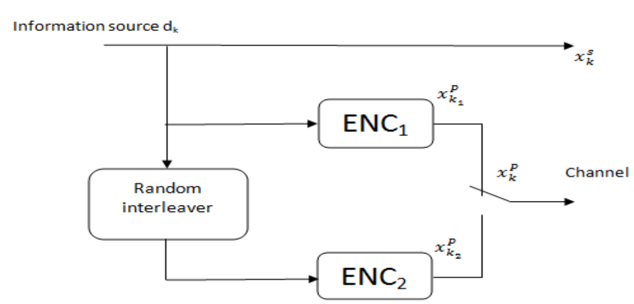

Fig-1. Structure of a turbo encoder (M.K.GUPTA ,VISHWAS SHARMA, 2009)

1.2 Turbo decoding

For implementing the turbo decoder, the very first consideration is to select a SISO(soft input soft output) algorithm, which can give efficient performance. A block diagram of a turbo decoder is shown in fig -2. (M.K.GUPTA ,VISHWAS SHARMA , 2009, Hsin-YihLi, 2006, Akash Kumar Gupta, sanjeet Kumar, 2010).

The turbo decoder consists of two component decoder DEC_1 to decode sequences from ENC_1, and DEC_2 to decode sequences from ENC_2.

Turbo code is composed of two decoders, linked by interleavers same to that of the encoder. Each of these decoders in (SISO)MAX-log- a Maximum A Posteriori (MAP) algorithm and Each decoder receives three input, the systematic bits y_k^s , the parity bit $y_{k_1}^p$ and $y_{k_2}^p$ transmitted from the corresponding encoder, and the information from the other component decoder providing reliability values for the decoded bits. This information is referred to as a-priori information $L(u_k)$. the component decoder exploits both the input from the channel reliability $L_c y_{ks}$ and the a-priori information $L(u_k)$ from the previous component decoder, and provides extrinsic information $L_e(u_k)$ known as the soft outputs for the decoded bits. Extrinsic information gives for each bit not only the hard decision but also the associated probability that it has been correctly decoded.

The soft output from the component decoder are represented in terms of Log-Likelihood Ratios (LLR); the sign bit gives the hard decision and the remaining bit provide the probability of a correct decision. The LLR $L(u_k)$ of a decoded bit u_k is given by:

$$L(u_k) = \ln\left\{\frac{P(u_k=+1)}{P(u_k=-1)}\right\} \qquad (1)$$

Where $P(u_k = +1)$ denotes the probability of decoded bit $u_k = +1$, and likewise for $P(u_k = -1)$.

For decoding of turbo codes, the decoder operates iteratively. In the first iteration, the first component decoder DEC1 takes channel output values and generates a soft output for the decoded bit. Notice that only channel output values are taken as input, since a-prior information from the other decoder is not yet available. The soft output from DEC1 is interleaved and inputted as a-priori information for the second decoder DEC2, along with the channel outputs. The soft output for the decoded bit is calculated, ending the first iteration. The second iteration begins, again DEC1 decodes the channel output, but this time with the a priori information de-interleaved from the soft output provided by DEC2 from the first iteration. This a-priori information allows DEC1 to obtain more accurate soft output, which is then provided to DEC2 as a-priori information. This process is repeated, and the BER of the decoded bits tends to fall.

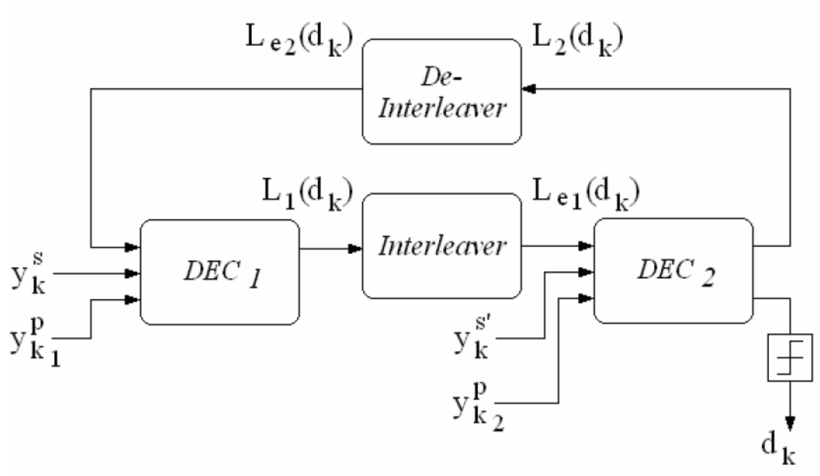

Fig -2. Turbo decoder structure (Hsin-YihLi, 2006)

1.3 Max-Log-MAP Algorithm

The Max-Log-MAP Algorithm was proposed by Koch and Baier and Erfanian. It simplified the MAP algorithm by transferring the recursions into the log arithmetic domain and using an approximation, reducing the complexity.

The MAP algorithm gives for each decoded bit u_k the probability that this bit was +1 or -1,given the received symbol sequence $\underline{y}$.(Hsin-YihLi, 2006)

the a-posteriori LLR$L\left(u_k \middle| \underline{y}\right) = \ln\left(\frac{P\left(u_k=+1\middle|\underline{y}\right)}{P\left(u_k=-1\middle|\underline{y}\right)}\right)$ (2)

let S_k denote a state at time stage k, the previous state S_{k-1}= S' and present state S_k = S are known in a trellis, then the input bit u_k that caused the transition between these two states will be known. The fact that the transitions between the previous state S_{k-1} and present state S_k in a trellis are mutually exclusive, and with Bayes' rule, eq.(2) rewrite as

$$L\left(u_k \middle| \underline{y}\right) = \ln\left(\frac{\sum_{u_k=+1} P(S_{k-1}=s' \Lambda S_k=S\Lambda \underline{y})}{\sum_{u_k=-1} P(S_{k-1}=s' \Lambda S_k=S\Lambda \underline{y})}\right) \quad (3)$$

Let denote the probability $P(S_{k-1} = s' \Lambda S_k = S\Lambda \underline{y})$ as $P(s' \Lambda S \Lambda \underline{y})$, break the received sequence $\underline{y}$ into three sections: 1) the received codeword associated with the present transition $\underline{y}_k$; 2) the received sequence prior to the present transition $\underline{y}_{j>k}$; 3) the received sequence after the present transition $\underline{y}_{j<k}$,the probability $P(s' \Lambda S \Lambda \underline{y})$ can written as

$$P\left(s' \Lambda S \Lambda \underline{y}\right) = P(s' \Lambda S \wedge \underline{y}_{j<k} \wedge \underline{y}_k \wedge \underline{y}_{j>k}) \quad (4)$$

Applying Bayes' rule $P(a\Lambda\, b) = P(a/b).P(b)$ and assuming the channel is memoryless, the future received sequence $\underline{y}_{j>k}$ will depend only on the present state S_k and not on the previous state S_{k-1} or the present and previous received channel sequences $\underline{y}$ and $\underline{y}_{j<k}$, we can obtain

$$P\left(s'\Lambda\, S\, \Lambda \underline{y}\right) = \alpha_{k-1}(S') \,.\gamma_k(S',S).\beta_k(S) \qquad (5)$$

from fig-3. We can see the meaning of the probability terms $\alpha_{k-1}(S')$, $\gamma_k(S',S)$and $\beta_k(S',S)$.

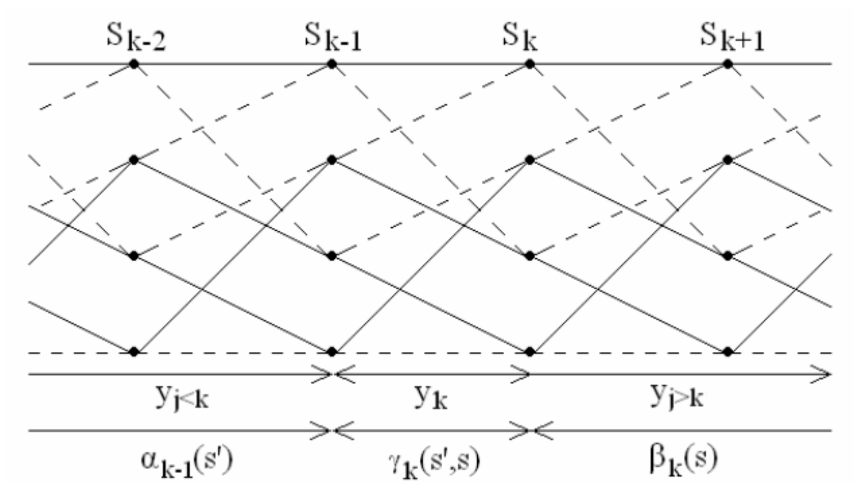

Fig-3. MAP decoder trellis (Hsin-YihLi, 2006)

Rewrite the conditional LLR of u_k , given the received sequence $\underline{y}$

$$L\left(u_k\middle|\underline{y}\right) = \ln\left(\frac{\sum_{u_k=+1} \quad P(S_{k-1}=s'\Lambda S_k=S\Lambda \underline{y})}{\sum_{u_k=-1} \quad P(S_{k-1}=s'\Lambda S_k=S\Lambda \underline{y})}\right)$$

$$= \ln\left(\frac{\sum_{u_k=+1} \quad \alpha_{k-1}(S').\gamma_k(S',S).\beta_k(S)}{\sum_{u_k=-1} \quad \alpha_{k-1}(S').\gamma_k(S',S).\beta_k(S)}\right) \qquad (6)$$

The forward recursion: (Hsin-YihLi, 2006)

$$\alpha_k(S) = \sum_{all\ s'} \quad \alpha_{k-1}(S').\gamma_k(S',S) \qquad (7)$$

The backward recursion :(Hsin-YihLi, 2006)

$$\beta_{k-1}(S') = \sum_{all\ s'} \beta_k(S).\gamma_k(S',S) \qquad (8)$$

branch transition probabilities: (Hsin-YihLi, 2006)

$$\gamma_k(S',S) = P\left(\underline{y}_k \middle| \underline{x}_k\right).P(u_k) \qquad (9)$$

where $\underline{x}_k$ and $\underline{y}_k$ are the transmitted and received codeword

The Max-Log- MAP algorithm further simplified this using the approximation:

$$\ln(e^{\delta_1} + e^{\delta_2} + \cdots + e^{\delta_n}) \approx \max_{i\in\{1\ldots n\}} \delta_i \qquad (10)$$

The Log arithmetic domain of the equation are derived, and We define $A_K(S)$,$B_K(S)$ and $\Gamma_K(S',S)$ as followed:(Hsin-YihLi, 2006)

$$A_K(S) = \ln\left(\alpha_k(S)\right)$$

$$= \ln\left(\sum_{all\ s'} \alpha_{k-1}(S').\gamma_k(S',S)\right)$$

$$= \ln\left(\sum_{al\ s'} exp[A_{K-1}(S') + \Gamma_K(S',S)]\right)$$

$$\approx \max_{S'}\left(A_{K-1}(S') + \Gamma_K(S',S)\right) \qquad (11)$$

$$B_K(S) = \ln(\beta_k(S))$$

$$B_{K-1}(S') = \ln(\beta_{k-1}(S'))$$

$$\approx \max_{S'}\left(B_K(S) + \Gamma_K(S',S)\right) \qquad (12)$$

$$\Gamma_K(S',S) = \ln(\gamma_k(S',S))$$

$$= \ln P\left(\underline{y}_k \middle| \underline{x}_k\right) + \ln P(u_k) \qquad (13)$$

Using the Log-likelihoods, the a priori probability $P(u_k)$ can be expressed as

$$P(u_k = +1) = \frac{e^{L(u_k)}}{1+e^{L(u_k)}} = \frac{1}{1+e^{-L(u_k)}}$$

$$P(u_k = -1) = \frac{e^{-L(u_k)}}{1+e^{-L(u_k)}} \qquad (14)$$

The probability with general equation :

$$P(u_k = \pm 1) = \frac{e^{\pm L(u_k)}}{1+e^{\pm L(u_k)}} = \left(\frac{e^{-\frac{L(u_k)}{2}}}{1+e^{-L(u_k)}} \right). e^{\frac{u_k L(u_k)}{2}}$$

$$= C_z . e^{u_k L(u_k)/2} \qquad (15)$$

The conditional received sequence probability $P\left(\underline{y}_k \middle| \underline{x}_k\right)$ is given , assuming a Gaussian channel with BPSK modulation, as

$$P\left(\underline{y}_k \middle| \underline{x}_k\right) = \prod_{l=1}^{n} P(y_{kl}|x_{kl})$$

$$= \prod_{l=1}^{n} \frac{1}{\sqrt{2\pi\sigma}} \ exp\left(-\frac{E_b R}{2\sigma^2}(y_{kl} - a x_{kl})^2\right)$$

$$= \frac{1}{\left(\sqrt{2\pi\sigma}\right)^n} \ exp\left(-\frac{E_b}{2\sigma^2} \sum_{l=1}^{n} \left({y_{kl}}^2 + a^2 {x_{kl}}^2 - 2a x_{kl} y_{kl}\right)\right)$$

$$= \ C_y . C_x . exp\left(\frac{E_b}{2\sigma^2} 2a \sum_{l=1}^{n} y_{kl} x_{kl}\right) \qquad (16)$$

Where x_{kl} and y_{kl} are individual bits within the transmitted and received codeword $\underline{x}_k$ and $\underline{y}_k$ respectively, n is the number of bits in each codeword, E_b is the transmitted energy per bit, R is the code rate and a is the fading amplitude.(Hsin-YihLi, 2006)

Substitute (15) and (16) in eq. (13),we get

$$\Gamma_K(S',S) = \ln C_y . C_x . exp\left(\frac{E_b}{2\sigma^2} 2a \sum_{l=1}^{n} y_{kl} x_{kl}\right) + \ln C_z . e^{u_k L(u_k)/2}$$

$$= C + \frac{L_c}{2} \sum_{l=1}^{n} y_{kl} x_{kl} + \frac{1}{2} u_k L(u_k) \qquad (17)$$

Where L_c is denoted as the channel reliability. The approximation of the LLR of each bit u_k is:

$$L\left(u_k \middle| \underline{y}\right) = \ln\left(\frac{\sum_{u_k=+1} \alpha_{k-1}(S') . \gamma_k(S',S) . \beta_k(S)}{\sum_{u_k=-1} \alpha_{k-1}(S') . \gamma_k(S',S) . \beta_k(S)}\right)$$

$$= \ln\left(\frac{\sum_{u_k=+1} \exp\left(A_{k-1}(S') . \Gamma_k(S',S) . B_k(S)\right)}{\sum_{u_k=-1} \exp\left(A_{k-1}(S') . \Gamma_k(S',S) . B_k(S)\right)}\right) \qquad (18)$$

$$\approx \max_{u_k=+1}\left(A_{k-1}(S') + \Gamma_k(S',S) + B_k(S)\right) - \max_{u_k=-1}\left(A_{k-1}(S') + \Gamma_k(S',S) + B_k(S)\right)$$

The a-posteriori LLR $L\left(u_k \middle| \underline{y}\right)$ is calculated by considering every transition from the trellis stage S_{k-1} to the stage S_k . these transition are split into two groups: those that might have occurred if $u_k = +1$ and those that might have occurred if $u_k = -1$. The transition giving the maximum value of $A_{k-1}(S') + \Gamma_k(S',S) + B_k(S)$ is found both group and the a-posteriori LLR is calculated based on only these two best transitions.(Hsin-YihLi, 2006)

2- System Model

The transmission conditions in wireless communication channels are severe due to multipath fading. Therefore, in order to design a communication system with an acceptable BER for lower level of signal to noise ratio, error correction coding must be used to protect the data from transmission errors. Linear block codes, convolutional codes (Hrudananda Pradhan, 2009 , Nguyen, and Kuchenbecker, 2001), and turbo codes are the most widely used error correction codes. A block diagram of COFDM system is shown in fig -4. The system use Turbo codes based on parallel concatenated convolutional code and have best performance compared to convolutional and Reed Solomon codes due to its soft-in, soft-out (SISO) decoding algorithm.

In OFDM high bit rate data is divided into N low bit rate parallel data streams and then transmitted simultaneously with deferent frequencies. The OFDM time signal is generated by an inverse FFT.

The IFFT takes in N symbols at a time where N is the number of subcarriers in the system. Each of these N input symbols has a symbol period of T seconds. The basis functions for an IFFT are N orthogonal sinusoids. Each input symbol acts like a complex weight for the corresponding sinusoidal basis function. Since the input symbols are complex, the value of the symbol determines both the amplitude and phase of the sinusoid for that subcarrier. The IFFT output is the summation of all N sinusoids. Thus, the IFFT block provides a simple way to modulate data onto N orthogonal subcarriers. The block of N output samples from the IFFT make up a single OFDM symbol.

Cyclic prefix is a crucial feature of OFDM used to combat the Inter- Symbol-Interference (ISI) introduced by the multi- path channel through which the signal is propagated. The basic idea is to replicate part of the OFDM time- domain waveform from the back to the front to create a guard period. The duration of the guard period Tg should be longer than the worst- case delay spread of the target multi- path environment. Parallel data converted into serial form and then transmitted over the Rayleigh fading channel.
On the receiver side of the COFDM system, corrupted signal by the channel is received and it basically does the reverse operation to the transmitter. The received signal is serial to parallel converted, the guard period is removed, the FFT converts the signal hack to the frequency domain. This frequency domain signal is coherently demodulated. Then turbo decoder back the data as the original.

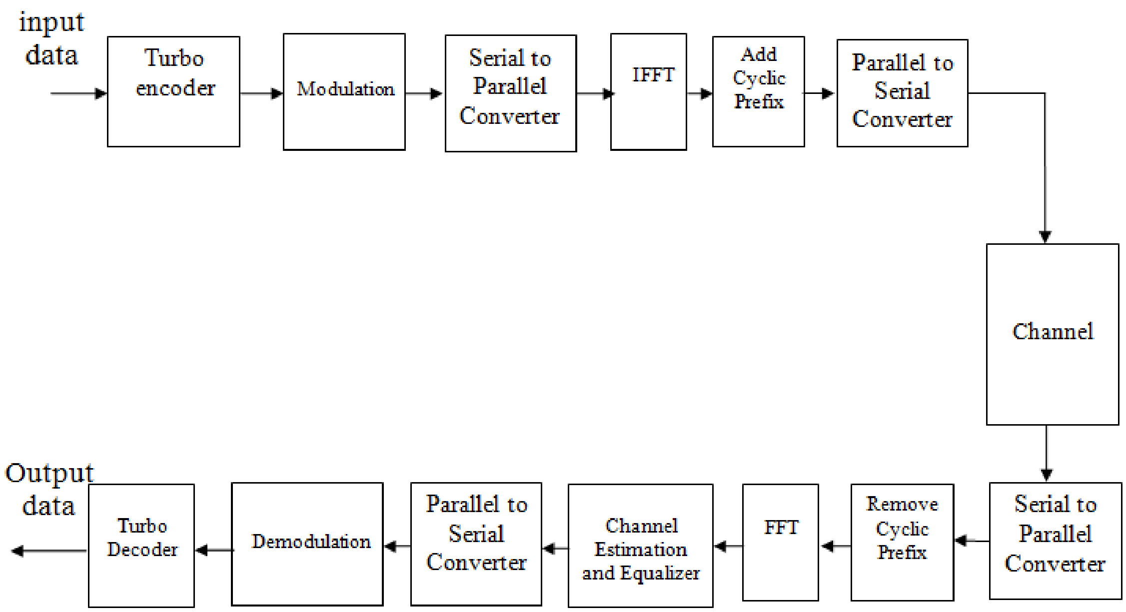

Fig -4.COFDM system diagram

2.1 Rayleigh fading channel

Raleigh fading (E.K Hall, S.G Wilson, 1998), is a statistical model for the effect of a propagation environment on a radio signal such as that used by wireless devices. It assumes that the power of a signal that has passed through such a transmission medium (also called a communications channels) will vary randomly or fade according to a Raleigh distribution the radial component of the sum of two uncorrelated Gaussian random variables. It is reasonable model for tropospheric and ionospheric signal propagation as well as the effect of heavily built up urban environment on radio signals.
Raleigh fading is most applicable when there is no line of sight between the transmitter and receiver (Hrudananda Pradhan, 2009).

2.2 Channel Estimation

Channel estimation can be achieved by transmitting pilot OFDM symbol as a preamble . To design a channel estimator for wireless systems with both low complexity and good channel tracking ability, one must choose a way of how pilot information (data/signals known to the receiver) should be transmitted. These pilots are usually needed as a point of reference for such estimator .

A fading channel requires constant tracking so pilot information has to be transmitted more or less continuously. However, an efficient way of allowing continuously update channel estimate is to transmit pilot symbol instead of data at certain location of the OFDM time frequency lattice .

Assuming P is the transmitted pilot data, the received signal after FFT is:

$$Y(k) = H(k)P(k) + W(k) \qquad (19)$$

where w(k) is the noise components, and since, the pilot data is known at the receiver, then the simplest way to estimate the channel is by dividing the received signal by the known pilot :

$$\tilde{H}(k) = Y(k)/P(k) \qquad (20)$$

where $\tilde{H}(K)$ is the estimate of the channel, and without noise, this gives the correct estimation. When noise is present, there could be an error (Buthaina Mosa Omran ,2007). The channel estimation can be performed by either inserting pilot tones into all of the subcarriers of OFDM symbols with a specific period or inserting pilot tones into each OFDM symbol.

2.3 Equalization

Although the guard time which has longer duration than the delay spread of a multipath channel can eliminate ISI because of the previous symbol, but it is still have some ISI because of the frequency selectivity of the channel. In order to compensate this distortion, a one-tap channel equalizer is needed. At the output of FFT on the receiver side, the sample at each subcarrier is multiplied by the coefficient of the corresponding channel equalizer. (Kamran arshad, 2002).

3- Simulation

3.1 Simulation parameters

The simulation parameters used are in the following table 1.

Table 1. Simulation parameters

parameters	values
Turbo code	PCCC
Digital modulation	BPSK, QPSK, QAM
Turbo code rate	½
SISO decoder	MAX- log-MAP
Interleaver	random interleaver
Iteration	6
Channel	AWGN, Rayleigh fading
Number of data sub carriers	60
Number of pilot sub carriers	4
Number of total sub carriers	64
Sub carrier frequency spacing	(20 MHZ/64)
IFFT,FFT period	3.2µsec
Guard interval time	0.8 µsec
Signal duration	4.0 µsec

3.2 Algorithm of simulation

We measured the performance of the turbo coded OFDM through MATLAB simulation. The simulation follows the procedure listed below:

1. Generate the information bits randomly.
2. Encode the information bits using a turbo encoder .
3. Use BPSK,QPSK or different QAM modulation to convert the binary bits, 0 and 1, into complex signals .
4. Performed serial to parallel conversion.
5. Use IFFT to generate OFDM signals
6. add cyclic prefix to create a guard period.
7. Use parallel to serial convertor to transmit signal serially.
8. Introduce noise to simulate channel errors. We assume that the signals are transmitted over an AWGN (Additive White Gaussian Noise) and Rayleigh channel.
9. At the receiver side, perform reverse operations to decode the received sequence.
10. Count the number of erroneous bits by comparing the decoded bit sequence with the original one.
11. Calculate the BER and plot it.

4- Results

All the simulations are done to achieve BER at 10^{-3}. .For simulation results two channel are AWGN and RAYLEIGH are used. The BER performance of TCOFDM system is compared with uncoded OFDM system. Fig -5. shows the performance of the uncoded OFDM system with AWGN.

When the OFDM signal is perfectly equalized, each subcarrier seems to be modulated independently as if in a single carrier scheme. Thus, the error probability of OFDM is the same with the error probability of a single carrier scheme which uses the same modulation map.

As in a single carrier scheme, BPSK and QPSK have better performance than 16-QAM and 64-QAM. The BER is very low after 10 dB SNR for BPSK and 14 dB SNR for QPSK, whereas the limit for such a low BER is 23 dB when 16-QAM is employed. This behavior is inherited from the single carrier schemes.

Turbo coded OFDM offers a good performance comparable to the of Convolutional coded OFDM and uncoded under Rayleigh channel . fig-6. Shows the BER equal 10^{-3} is approximately obtain at 5.5dB for TCOFDM and 6.5 for Convolution OFDM where the uncoded have 10.5 dB.

In fig-7. It shown that turbo codes with BPSK modulation under Rayleigh fading channel gives performance improvement of approximately 5.5 dB over the QPSK (7.5 dB) and 16-QAM (10.5 dB).fig-8. Shown the BER vs. SNR for uncoded and turbo code OFDM under the Rayleigh and AWGN channels. We achieve the BER at 10^{-3} approximately at 4.5dB for TCOFDM under AWGN and 5.5 dB in Rayleigh channel , the uncoded have 8.5 dB in AWGN and 10.5 dB in Rayleigh channel. In fig-9.the BER performance plot for uncoded and turbo coded OFDM under Rayleigh fading channel with different Doppler frequency which The sampled impulse response for fd=5 ,50 and 200 are plotted in fig-10.,fig-11. and fig -12., we observe that turbo codes give good BER performance compared with uncoded OFDM, the performance is consider very well in operation under fading channel.

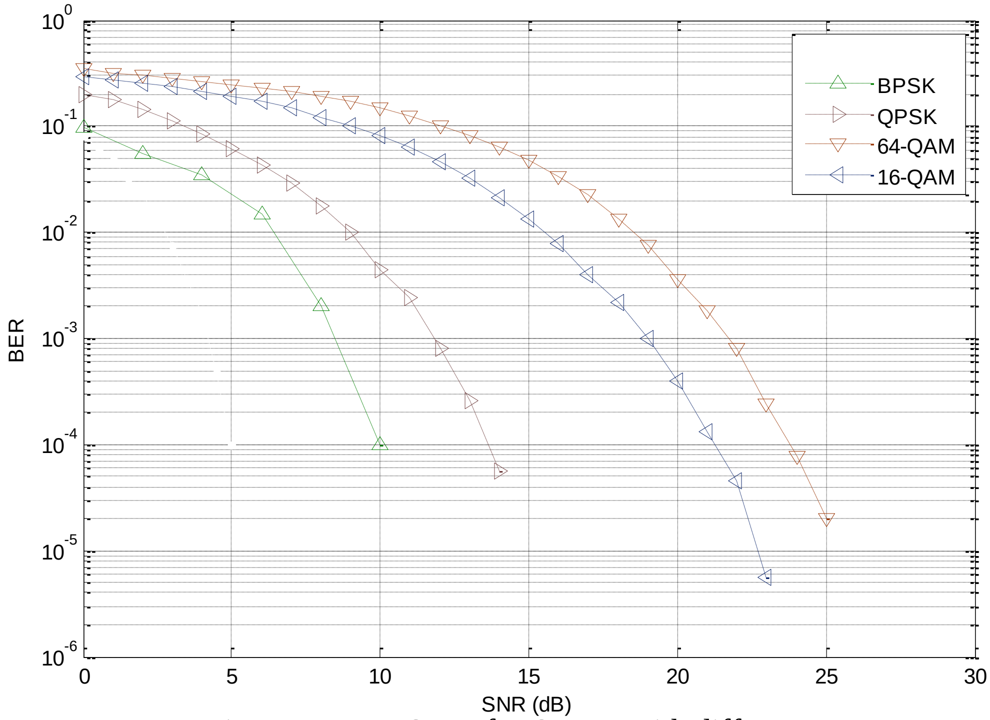

Fig-5. BER vs. SNR for OFDM with different modulation under AWGN channel

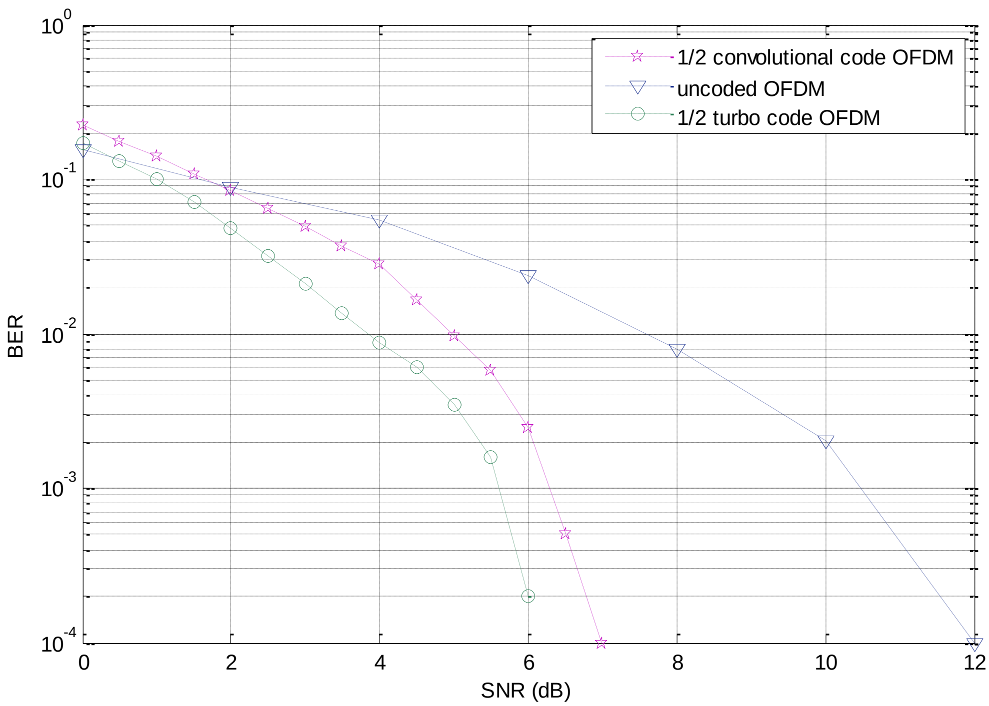

Fig-6. Comparison between turbo coded OFDM with convolutional coded and uncoded OFDM under Rayleigh fading channel.

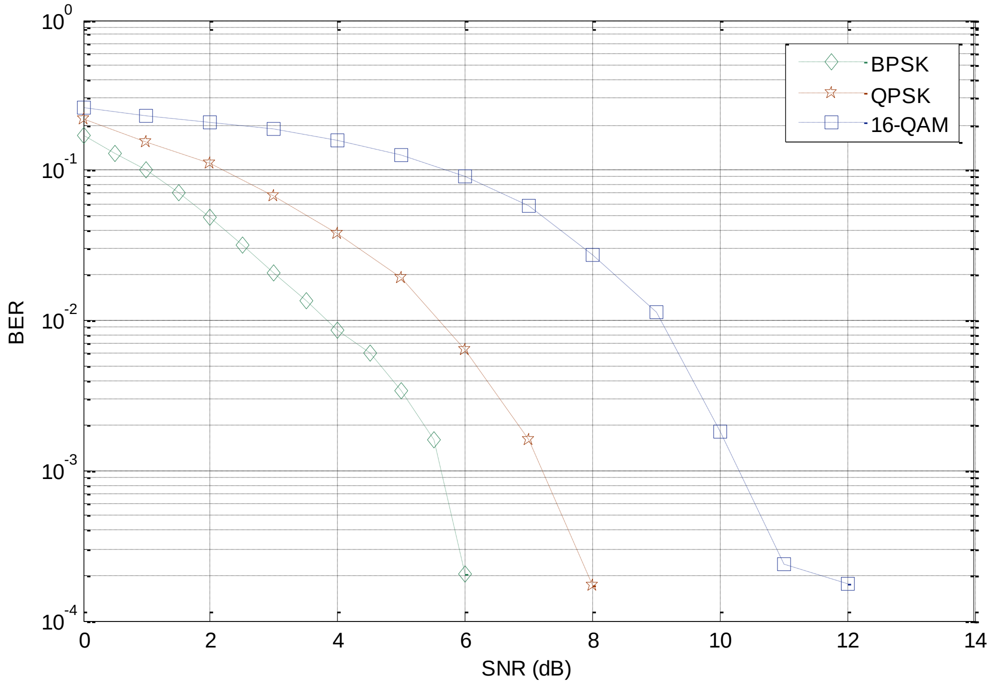

Fig-7. BER vs. SNR for turbo coded OFDM with different modulation under Rayleigh fading channel.

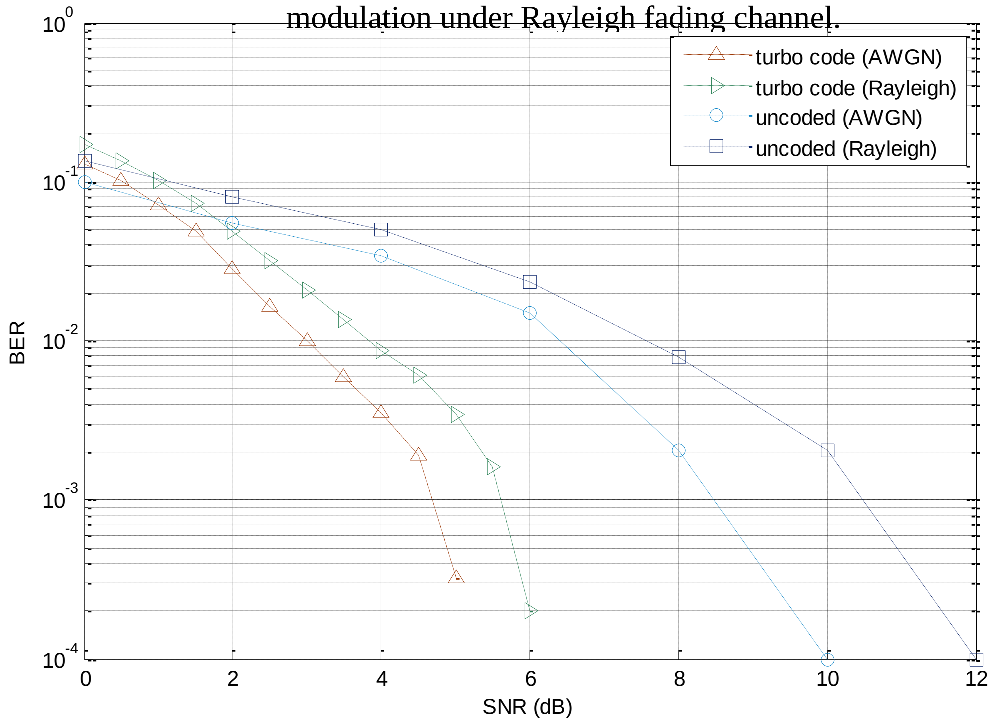

Fig-8. BER vs. SNR for ucnoded and turbo coded OFDM under Rayleigh fading channel and AWGN channel.

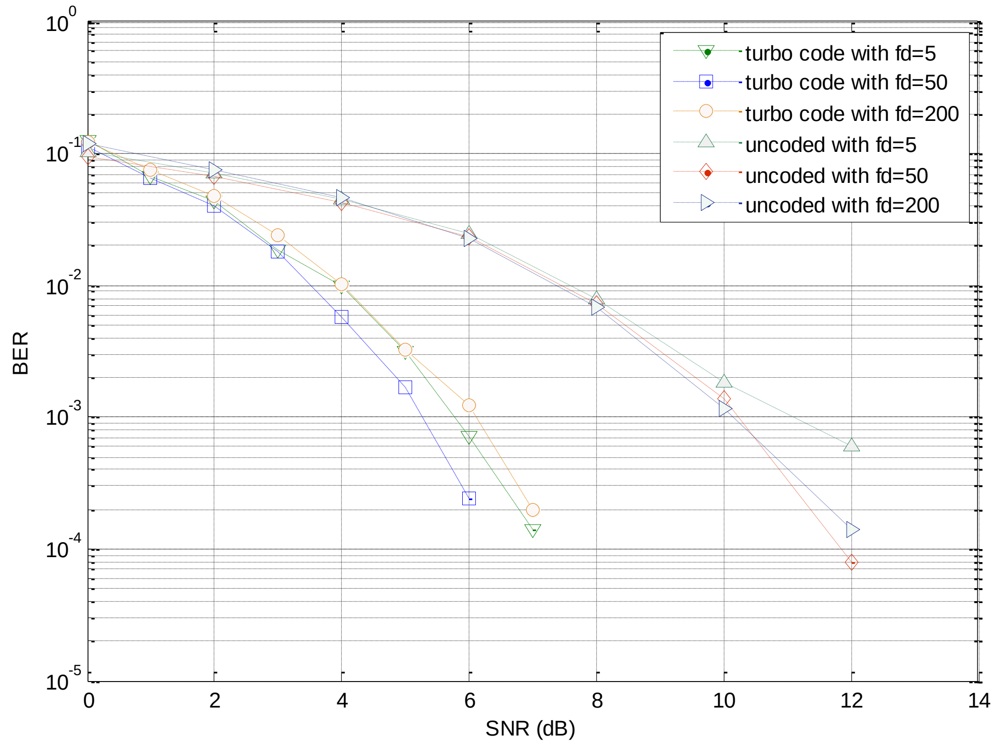

Fig-9. BER vs. SNR for ucnoded and turbo coded OFDM under Rayleigh fading channel with different Doppler frequency

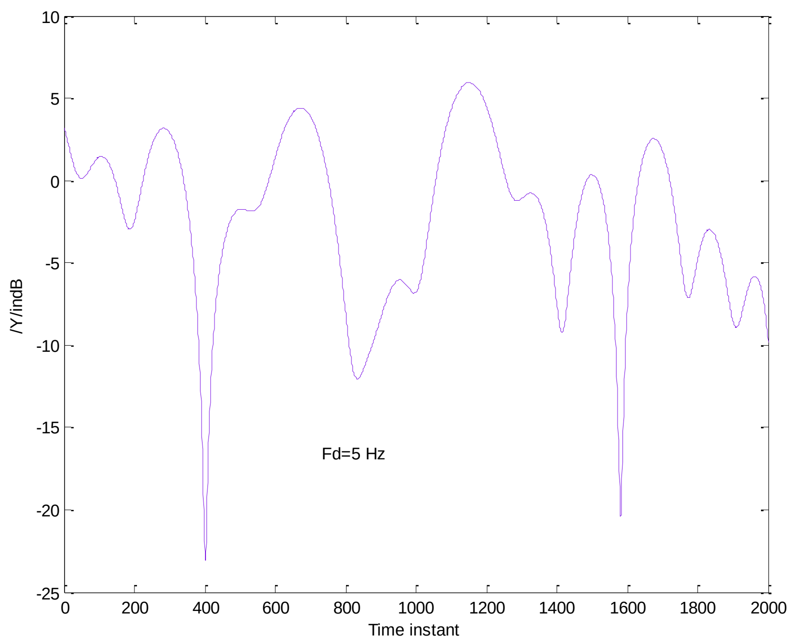

Fig-10.sampled impulse response of Rayleigh fading channel with fd=5

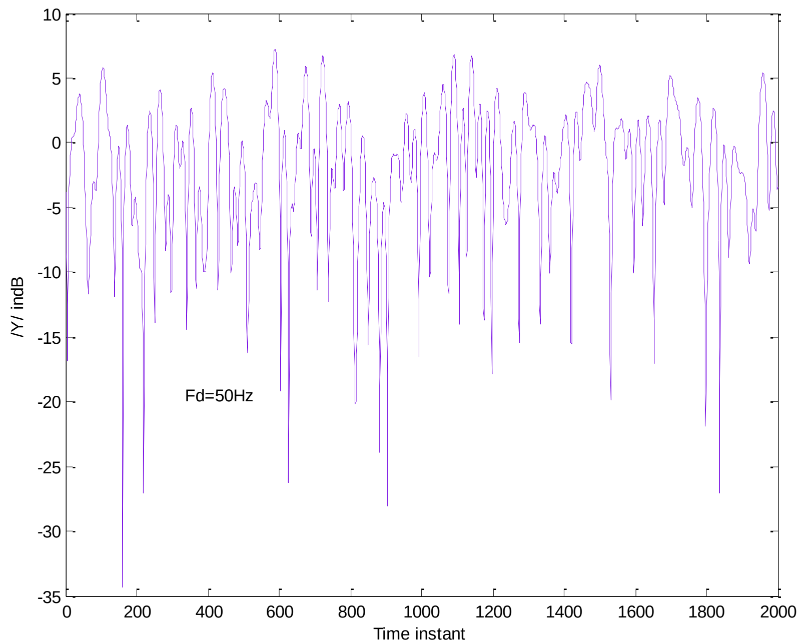

Fig-11.sampled impulse response of Rayleigh fading channel with fd=50

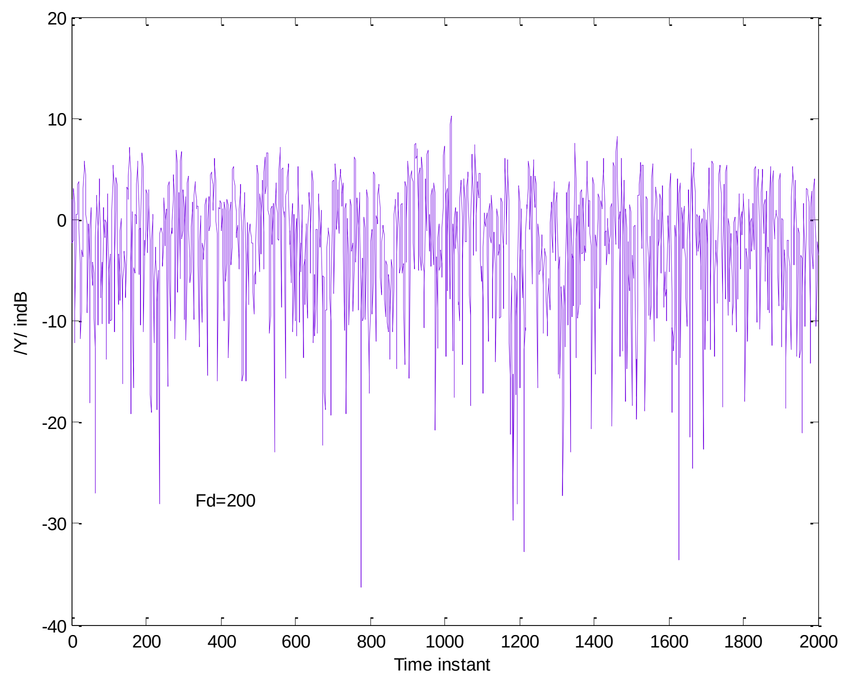

Fig-12.sampled impulse response of Rayleigh fading channel with fd=200

5- Conclusion

In this work, the performance of the PCCC turbo code OFDM system is evaluated by simulation in different channels AWGN and Rayleigh fading channel. It can show that we are able to improve the performance of uncoded OFDM by convolutional coding scheme, further improvement on the performance has been achieved by applying turbo coding to uncoded OFDM system. Turbo codes with 6 decoding iteration have been evaluated, the turbo code with 6 iteration are sufficient to provide good BER performance at SNR equal 10^{-3} , that are suitable for speed and data application .

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