BLOCK TURBO CODE TO OFDM FOR WIRELESS LOCAL AREA NETWORK

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Abstract:-

To overcome multipath fading and Inter Symbol Interference(ISI) in convolutional single carrier system equalizers are used. Another approach is to use a multicarrier modulation techniques such as OFDM. To avoid ISI a small interval known as the guard time interval is inserted into OFDM symbols.

The another problem is the reduction of the error rate in transmitting digital data. For that we use error correcting codes in the design of digital transmission system. Turbo codes can be achieved by serial or parallel concatenation of two(or more) codes called the constituent codes. In this paper Block Turbo codes or turbo Product codes are used which is similar to the IEEE 802.11A WLAN standard.

Index Terms:-**Introduction, Block Turbo Code, COFDM transmission & reception, Conclusion**

I. Introduction

Wireless technologies are the veritable explosions in telecommunication industries. Onceexclusively military, satellite and cellular technologies are now commercially driven by ever more demanding consumers, who are ready for seamless communication from their home to their car, to their office, or even for outdoor activities. With this increased demand comes a growing need to transmit information wirelessly, quickly and accurately. To address this need, communications engineer have combined technologies suitable for high rate transmission with forward error correction (FEC) techniques. Orthogonal Frequency Division Multiplexing (OFDM) is the standard being used throughout the world to achieve the high data rates necessary for data intensive applications that must now become routine. A particularly attractive feature of OFDM systems is that they are capable of operating without a classic equalizer, when communicating over depressive transmission media, such as wireless channels, while conveniently accommodation the time- and frequency-domain channel quality fluctuations of the wireless channel. For the quality communications, this must be done without severe degradation in theperformance of the system. FDMA, TDMA and CDMA are the well known multiplexing techniques used in wireless communication systems. While working with the wireless systems using these techniques various problems encountered are (1) multi-path fading (2) timedispersion which lead to inter symbol interference (ISI) (3) lower bit rate capacity (4)requirement of larger transmit power for high bit rate and (5) less spectral efficiency.

Modulation Techniques inOFDM

Modulation is the process of facilitating the transfer of information over a medium.Sound transmission in air has limited range for the amount of power our lungs can generate. To extend the range our voice can reach, we need to transmit it through a medium other than air, such as a phone line or radio

There are two types of modulations: Analog modulation and digital modulation. In analog modulation, an information-bearing analog waveform is impressed on the carrier signal for transmission whereas in digital modulation, an information-bearing discrete-time symbol sequence (digital signal) is converted or impressed onto a continuous-time carrier waveform for transmission. 2G wireless systems are realized using digital modulation schemes.

Digital Modulation Techniques

Three Basic concepts of modulation :

There are three basic types of digital modulation techniques. These are

1. Amplitude-Shift Keying (ASK)
2. Frequency-Shift Keying (FSK)
3. Phase-Shift Keying (PSK)

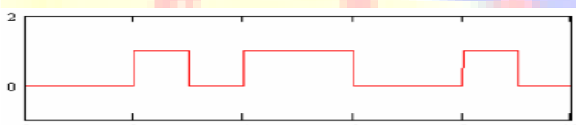
All of these techniques vary a parameter of a sinusoid to represent the information which we wish to send. A general carrier wave may be written: **c(t)=Asin**$(2\pi f t+\theta)$

A sinusoid has three different parameters than can be varied. These are its amplitude, phase and frequency. Modulation is a process of mapping such that it takes your voice (as an example of a signal) converts it into some aspect of a sine wave and then transmits the sine wave, leaving the actual voice behind. The sine wave on the other side is remapped back to a near copy of your sound.

Amplitude Shift keying(ASK):

In ASK, the amplitude of the carrier is changed in response to information and all else is kept fixed. In Binary ASK Bit 1 is transmitted by a carrier of one particular amplitude. To transmit 0, we change the amplitude keeping the frequency constant. On-Off Keying (OOK) is a special form of ASK, where one of the amplitudes is zero as shown in fig 1 and fig 2.

Binary ASK(t)=s(t) sin $(2\pi f t)$



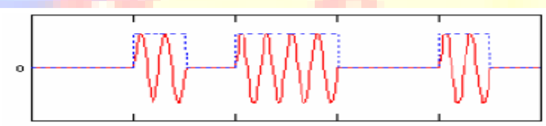


Fig 1 Baseband information sequence 0010110010Fig 2 Binary ASK (OOK) signal

Frequency Shift Keying(FSK) :

In FSK, we change the frequency in response to information, In Binary FSK one particular frequency for a 1 and another frequency for a 0 is used as shown in fig 3 for the same bit sequence as above. In the example below, frequency f1 for bit 1 is higher than f2 used for the 0 bit.

Binary FSK(t)= $sin(2\pi f1\ t)$ **for Bit 1**

= $sin(2\pi f2\ t)$ **for Bit 0**

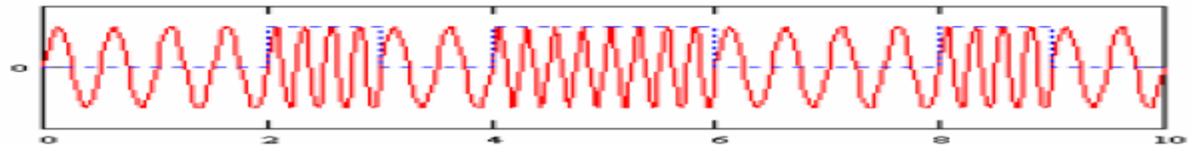


Fig 3 Binary FSK signal

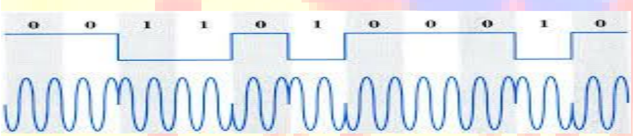
Phase Shift Keying(PSK) :

In PSK, we change the phase of the sinusoidal carrier to indicate information. Phase inthis context is the starting angle at which the sinusoid starts. For Binary PSK It has one fixed phase usually 0° when the data is 1. To transmit 0, we shift the phase of the sinusoid by 180°. Phase shift represents the change in the state of the information in this case. ASK techniques are most susceptible to the effects of non-linear devices which compress and distort signal amplitude. $Q(x)$ will give the probability that a single sample taken from a random process with zero-mean and unit-variance Gaussian probability density function will be greater or equal to x. It is a scaled form of the complementary Gaussian error function:

$$Q(x) = \frac{1}{\sqrt{2\pi}} \int_x^{\infty} e^{-t^2/2} dt = \frac{1}{2}\, \mathrm{erfc}\left(\frac{x}{\sqrt{2}}\right),\ x \geq 0.$$

Binary PSK(t)=$sin(2\pi ft)$ **for Bit 1**

= $sin(2\pi ft + \pi)$ **for Bit 0**



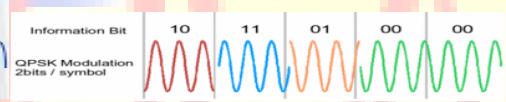


Fig 4 Binary PSK Carrier **Fig 5 QPSK signal**

Quadrature Phase Shift KeyingQPSK :

QPSK (4-ary PSK) involves changing the phase of the transmitted waveform. Each finitephase change represents unique digital data. A phase-modulated waveform can be generated by using the digital data to change the phase of a signal while its frequency and amplitude stay constant. A QPSK modulated carrier undergoes four distinct changes in phase that are represented as symbols and can take on the values of π/4, 3π/4, 5π/4, and 7π/4. Each symbolrepresents two binary bits of data. For a binary sequence m(t) = 0 0 0 1 1 0 1 1, if the sinusoid s(t) is of amplitude of A, then the resulting QPSK signal will be as shown in the figure 2.6. Phase of the sinusoid is shifted by 90°, 180°, 270°, 360° for data 00, 01, 10, 11 respectively. QPSK systems can be implemented in a number of ways. An illustration of the major components of the transmitter and receiver structure are shown below.

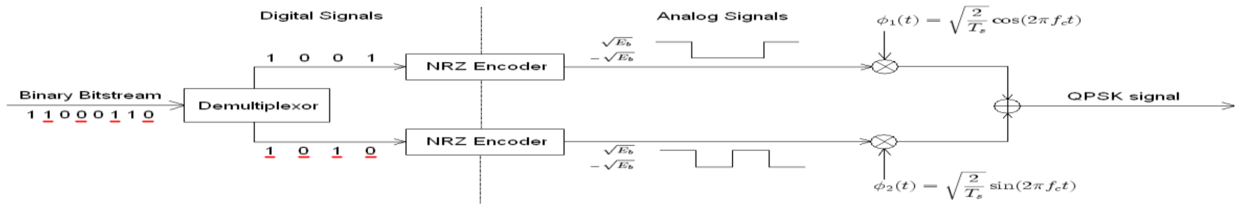


Fig 6 Transmitter QPSK

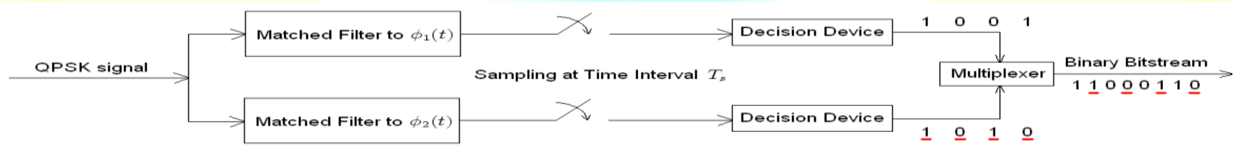


Fig 7 Receiver QPSK

Bit rate and symbol rate :

To understand and compare different PSK modulation format efficiencies, it is important to first understand the difference between bit rate and symbol rate. The signal bandwidth for the communications channel needed depends on the symbol rate, not on the bit rate. Bit rate is the frequency of a system bit stream. Take, for example, a radio with an 8 bit sampler, sampling at 10 kHz for voice. The bit rate, the basic bit stream rate in the radio, wouldbe eight bits multiplied by 10K samples per second, or 80 Kbits per second. (For the moment we will ignore the extra bits required for synchronization, error correction, etc.). The symbol rate is the bit rate divided by the number of bits that can be transmitted with each symbol.

$$\text{Symbol Rate} = \frac{Bit\ rate}{No.of\ bits\ transmitted\ with\ each\ symbol}$$

If one bit is transmitted per symbol, as with BPSK, then the symbol rate would be thesame as the bit rate of 80 Kbits per second. If two bits are transmitted per symbol, as in QPSK, then the symbol rate would be half of the bit rate or 40 Kbits per second. Symbol rate is sometimes called baud rate. Note that baud rate is not the same as bit rate. These terms are often confused. If more bits can be sent with each symbol, then the same amount of data can be sent in a narrower spectrum. This is why modulation formats that are more complex and use a higher number of states can send the same information over a narrower piece of the RF spectrum. The bit error rate (BER) of BPSK in AWGN can be calculated as:

$$P_b = Q\left(\sqrt{\frac{2E_b}{N_0}}\right).$$

Where $No/2$ = noise power spectral density (W/Hz)

Eb = PsTb is the energy contained in a bit duration.

Where Ps = power of sinusoid of amplitude A

= ½ A2

However, in order to achieve the same bit-error probability as BPSK, QPSK uses twice the power (since two bits are transmitted simultaneously).The symbol error rate is given by:

$$P_s = 1 - (1 - P_b)^2$$
$$= 2Q\left(\sqrt{\frac{E_s}{N_0}}\right) - \left[Q\left(\sqrt{\frac{E_s}{N_0}}\right)\right]^2.$$

If the signal-to-noise ratio is high (as is necessary for practical QPSK systems) the probability of symbol error may be approximated:

$$P_s \approx 2Q\left(\sqrt{\frac{E_s}{N_0}}\right)$$

Expressions for the symbol error-rate of rectangular QAM are not hard to derive but yield rather unpleasant expressions. They are most easily expressed in a *per carrier* sense:

$$P_s\ c = 2\left(1 - \frac{1}{\sqrt{M}}\right) Q\left[\sqrt{(3/(M-1))}\,\frac{Es}{No}\right]$$

$$P_s = 1 - (1 - P_b)^2$$

The bit-error rate will depend on the exact assignment of bits to symbols, but for a Graycodedassignment with equal bits per carrier:

$$\text{Pbc} = \frac{4}{k}\left[1 - \frac{1}{\sqrt{M}}\right] Q\left[\sqrt{(3k/M-1)}\,\frac{Es}{No}\right]$$

$$P_s = 1 - (1 - P_b)^2$$

M = Number of symbols in modulation constellation

Eb = Energy-per-bit

Es = Energy-per-symbol with *k* bits per symbol *bkE*

N0 = Noise power spectral density (W/Hz)

Pb = Probability of bit-error

Pbc = Probability of bit-error per carrier

Ps = Probability of symbol-error

Psc = Probability of symbol-error per carrier

II. Block Turbo Code

Turbo codes were first presented at the International Conference on Communications in1993 by C. Berrou. Until then, it was widely believed that to achieve near Shannon's bound performance, one would need to implement a decoder with infinite complexity or close. Turbo codes can be achieved by serial or parallel concatenation of two (or more) codes called the constituent codes. The constituent codes can be either block codes or convolutional codes. Currently, most of the work on turbo codes have essentially focused on Convolutional Turbo Code (CTC)s and Block Turbo Code (BTC)s have been partially neglected. Yet, the BTC solution is more attractive for a wide range of applications. In 1994 we proposed a new softinput/soft-output decoder [26] for all linear block codes and it has been showed that BTC had performances comparable to those of CTC using suboptimal weighting algorithms. BTC offers a good compromise between performance and complexity and is very attractive for implementation.

1. BTC resulted from the combination of three ideas that were known to all in the coding community

2. The utilization of block codes instead of commonly used non-systematic or systematic convolutional codes.

3. 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.

4. Encoders and decoders working on permuted versions of the same information. This is achieved by using an interleaver.

Concatenated codes :

The power of Forward Error Correction codes can be enhanced by using the concatenated codes, which are shown in Figure. Concatenated codes were first introduced by Elias in 1954[28]. The principle is to feed the output of one encoder (called the outer encoder) to the input of another encoder, and so on, as required. The final encoder before the channel is known as the inner encoder. The resulting composite code is clearly much more complex than any of the individual codes. However it can readily be decoded: we simply apply each of the component decoders in turn, from the inner to the outer.

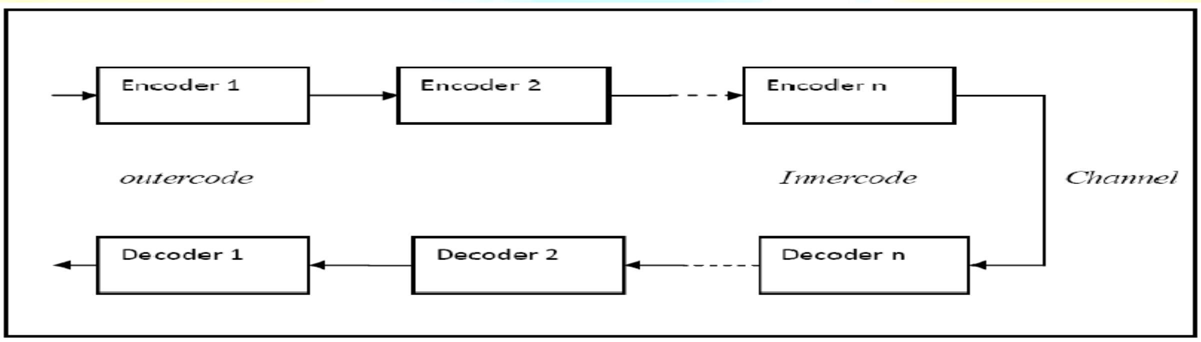


Fig 8 Principle of Concatenated codes

This simple scheme suffers a drawbacks which is called error propagation. If a decodingerror occurs in a codeword, it usually results in a number of data errors. When these are passed on to the next decoder they may overwhelm the ability of that code to correct the errors. The performance of the outer decoder might be improved if these errors were distributed between a number of separate code-words. This can be achieved using an interleaver/de-interleaver. This interleaver (sometimes known as a rectangular or block interleaver) consists of a twodimensionalarray, into which the data is read along its rows. Once the array is full, the data is read out by columns, thus permuting the order of the data.

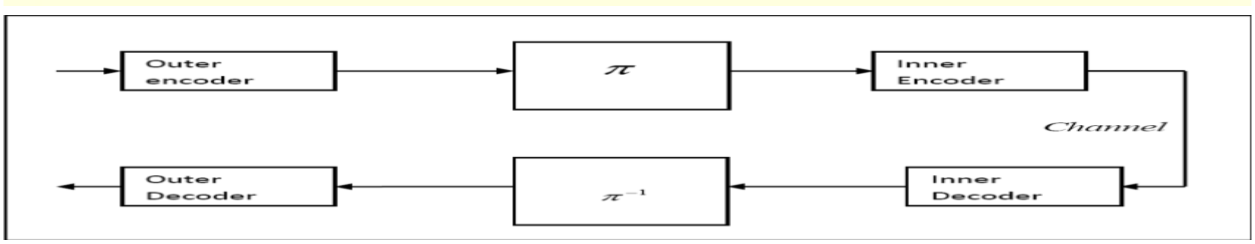
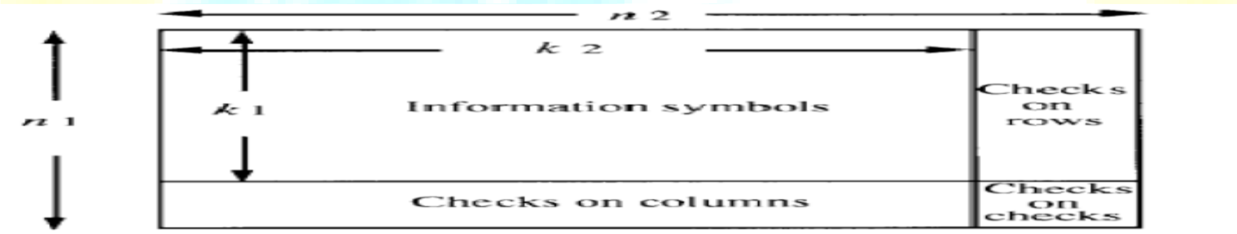


Fig 9 Concatenated encoder and decoder with interleaver

Block Turbo Code :

Block turbo codes (or product codes) are serially concatenated codes [27] which wereintroduced by Elias in 1954 [28]. The concept of product codes is very simple and relatively efficient for building very long block codes by using two or more short block codes. Let us consider two systematic linear block codes c^1 with parameters (n_1, k_1,δ_1) and $c^2$with parameters (n_2, k_2, δ_2), where n□ , k□ and δ stand for codeword length, number of information bits, and minimum Hamming distance, respectively. The product code is obtained (as shown in Figure 3.3) by

1. placing (k_1*k_2) information bits in an array of k_1 rows and k_2 columns;
2. coding the k_1 rows using code c^2;
3. coding the k_2 columns using code c^1.



Construction of product code $P = c^1 \otimes c^2$

As indicated by Elias [28], these codes can be decoded by sequentially decoding the rows and columns of P in order to reduce decoding complexity. However, to achieve optimum performance, one must use MLD (soft decoding) of the component codes. Thus, we need softinput/soft-output decoders to maintain optimum performance when decoding the rows andcolumns of P. Provided we have a soft-input/softoutput decoder for decoding the rows and columns of P, we can iterate the sequential decoding of P and thus reduce the BER after each iteration as for CTC [16].

Turbo decoding of product codes

The decoding procedure described below is generalized by cascading elementarydecoders illustrated in Fig. Let us consider the decoding of the rows and columns of a product code P described and transmitted on a Gaussian channel using BPSK signaling. On receiving matrix [R] corresponding to a transmitted codeword [E], the first decoder performs the soft decoding of the rows (or columns) of P using as input matrix [R]. Soft-input decoding is performed using the Chase algorithm and the soft output is computed.

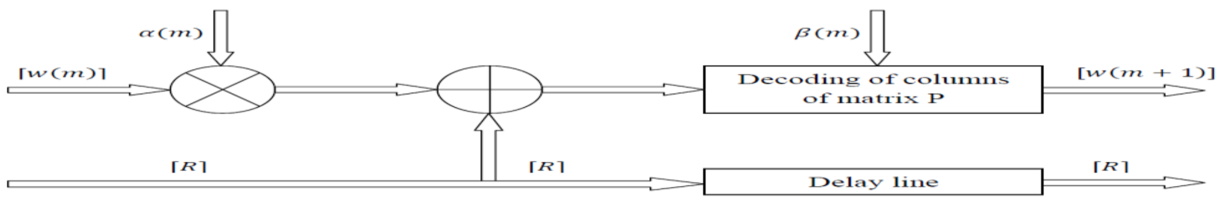


Fig 10 Block diagram of elementary block turbo decoder

By subtracting the soft input from the soft output we obtain the extrinsicinformation[$W(2)$] where index 2 indicates that we are considering the extrinsic information for the second decoding P of which was computed during the first decoding of P. The soft input for the decoding of the columns (or rows) at the second decoding of P is given by**[R(2)] = [R] +** $\boldsymbol{\alpha}(\mathbf{2})[\boldsymbol{\omega}(\mathbf{2})]$

where α(2) is a scaling factor which takes into account the fact that the standard deviation ofsamples in matrix [R] and in matrix [ω] are different (as given in [16]). The standard deviation of the extrinsic information is very high in the first decoding steps and decreases as we iterate the decoding. This scaling factor is also used to reduce the effect of the extrinsic information in the soft decoder in the first decoding steps when the BER is relatively high. It takes a small value in the first decoding steps and increases as the BER tends to zero.

Coded Orthogonal Frequency Division Multiplexing system model

Orthogonal Frequency Division Multiplexing (OFDM) also known as discrete multitonemodulation (DMT), is based upon the principle of frequency division multiplexing (FDM), but it utilized as a digital modulation scheme. The bit stream that is to be transmitted is split into several parallel bit streams, typically thousands. The available frequency spectrum is divided into sub-channels and each low rate bit stream is transmitted over one sub channel by modulating subchannel by , modulating a sub-carrier using a standard modulation scheme, for example: PSK, QAM. The sub-carrier frequencies are chosen so that the modulated data streams are orthogonal to each other, meaning that the signals are totally independentand cross talk between the subchannels is eliminated. It is achieved by ensuring that the carriers are placed exactly at the nulls in the modulation spectra of each other. Orthogonal Division Multiplexing (OFDM) has grown to be the most popular communications systems in high speed communications in the last decade. In fact, it has been said by many industry leaders that OFDM technology is the future of wireless communications.

Frequency division multiplexing modulation :

Frequency division multiplexing (FDM) extends the concept of single carrier modulationby using multiple subcarriers within the same single channel. The total data rate to be sent in the channel is divided between the various subcarriers. The data do not have to be divided evenly nor do they have to originate from the same information source. Advantages include using separate modulation/ demodulation customized to a particular type of data, or sending out banks of dissimilar data that can be best sent using multiple, and possibly different, modulationschemes. FDM offers an advantage over single-carrier modulation in terms of narrowband frequency interference since this interference will only affect one of the frequency sub-bands. The other subcarriers will not be affected by the interference. Since each subcarrier has a lowerinformation rate, the data symbol periods in a digital system will be longer, adding someadditional immunity to impulse noise and reflections.

Orthogonal frequency division multiplexing (OFDM):

OFDM is a combination of modulation and multiplexing. Multiplexing generally refers toindependent signals, those produced by different sources. In OFDM [2], the question of multiplexing is applied to independent signals but these independent signals are a sub-set of the one main signal. In OFDM the signal itself is first split into independent channels, modulated by data and then re-multiplexed to create the OFDM carrier. If the FDM system above had been able to use a set of subcarriers that were orthogonal to each other, a higher level of spectral efficiency could have been achieved. The guard bands that were necessary to allow individual demodulation of subcarriers in an FDM system would no longer be necessary. The use of orthogonal subcarriers would allow the subcarriers' spectra to overlap, thus increasing the spectral efficiency.

Coded orthogonal frequency division multiplexing:

Coded OFDM (COFDM) is a term used for a system in which the error control codingand OFDM modulation processes work closely together. COFDM , systems are able to achieve excellent performance on frequency selective channels because of the combined benefits of multicarrier modulation and coding. Due to the effects of noise and multipath fading in the channel, the transmitted signal arrives at the receiver with some errors

In forward error correction coding, a certain number of redundant bits are added to databits in a particular pattern according to the type of the code. In other words, for every *k* data bits,*n* coded bits are transmitted, where *n* >*k*. In the receiver, the *k* data bits can be recovered by performing a decoding operation on the *n* received coded bits. The transmission conditions in wireless communication channels are severe due to multipath fading and the variation of the signal-to-noise power ratio.

III.COFDM transmission & reception

The basic principle of OFDM, as mentioned in the above, is to divide a high-rate datastream into N lower rate streams and to transmit them at the same time over a number of subcarriers. Since the symbol duration is increased, the relative amount of dispersion in time caused by multipath delay spread is decreased. Intersymbol interference (ISI) is another problem, which can almost be eliminated by introducing a guard time in every OFDM symbol. In order to avoid the ICI, an OFDM symbol is cyclically extended by adding a guard time. A general block diagram of the transmitter and the receiver for the COFDM scheme is shown in Figure . Here is the transmitted sequence and is the estimated sequence of transmitted signal.

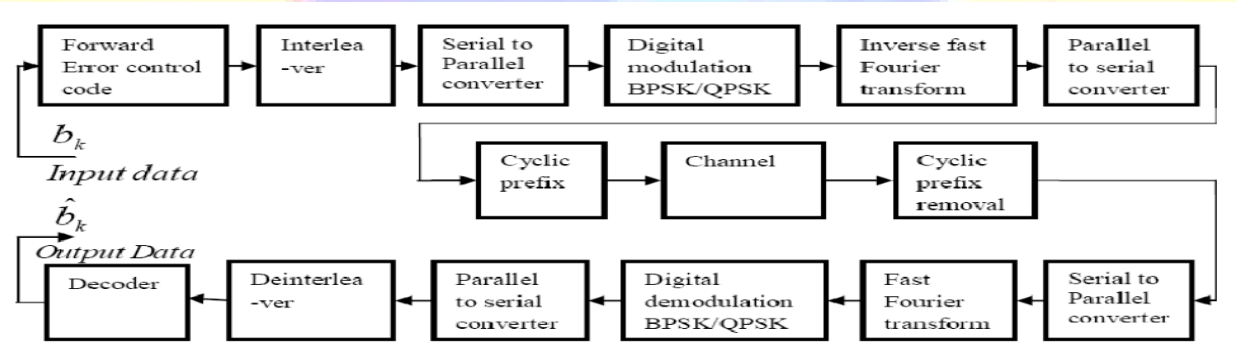


Fig 11 COFDM system diagram

FFT and IFFT :

OFDM systems are implemented using a combination of fast Fourier Transform (FFT)and inverse fast Fourier Transform (IFFT) blocks that are mathematically equivalent versions of the DFT and IDFT, respectively, but more efficient to implement. An OFDM system treats the source symbols at the transmitter as though they are in the frequency-domain. These symbols are used as the inputs to an IFFT block that brings the signal into the time domain. 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. These sinusoids each have a different frequency and the lowest frequency is DC. 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. 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. The length of the OFDM symbol is NT where T is the IFFT input symbol period mentioned above.

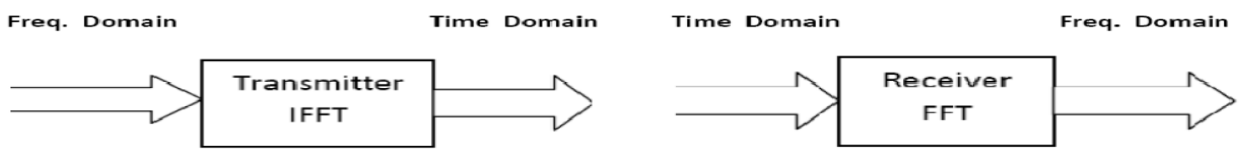


Fig 12 Block diagram of FFT and IFFT system

Additive White Gaussian noise (AWGN)

In the study of communication systems, the classical (ideal) additive white Gaussiannoise (AWGN) channel, with statistically independent Gaussian noise samples corrupting data samples free of intersymbol interference (ISI), is the usual staring point for understanding basic performance relationships. An AWGN channel adds white Gaussian noise ti the signal that passes through it.

$$r(t) = s(t) + n(t)$$

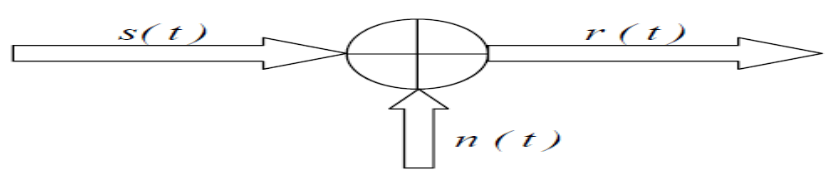


Fig 13 Received signal corrupted by AWGN

Where *n(t)* is a sample function of the AWGN process with probability density function (pdf) and power spectral density as follows

$$\theta\,nm(f) = \tfrac{1}{2}No\left[{}^{\omega}/_{Hz}\right]$$

WhereNo is a constant and called the noise power density.

Fading :

Fading is about the phenomenon of loss of signal in telecommunications. Fading orfading channels refers to mathematical models for the distortion that a carrier modulated telecommunicationsignal experiences over certain propagation media. Short team fading alsoknown as multipath induced fading is due to multipath propagation. Fading results from thesuperposition of transmitted signals that have experienced differences in attenuation, delay andphase shift while travelling from the source to the receiver. It may be caused by attenuation of asingle signal.

The most common types of fading are known as “slow fading” and “fast fading” as they apply to a mobile radio environment. Fading refers to the time variation of the received signal power caused by changes in the transmission medium or path. Fading channel models are often used to model electromagnetic transmission of information over wireless media such as with cellular phones and in broadcast communication. There are two types of fading based on multipath time delay spread:

1. **Flat fading :**The bandwidth of the signal is less than the coherence Bandwidth of the channel or the delay spread is less than the symbol period.
2. **Frequency selective fading :**The bandwidth of the signal is greater than the coherence bandwidth of the channel or the delay spread is greater than the symbol period.

There are two types of fading based on Doppler spread:

1. **Fast fading :**There exist a high Doppler spread and the coherence time is less than the symbol time and the channel variations are faster than baseband signal variation.
2. **Slow fading :**It has a low Doppler spread. The coherence time is greater than the symbol period and the channel variations are slower than the baseband signal variation.

Matlab simulation model :

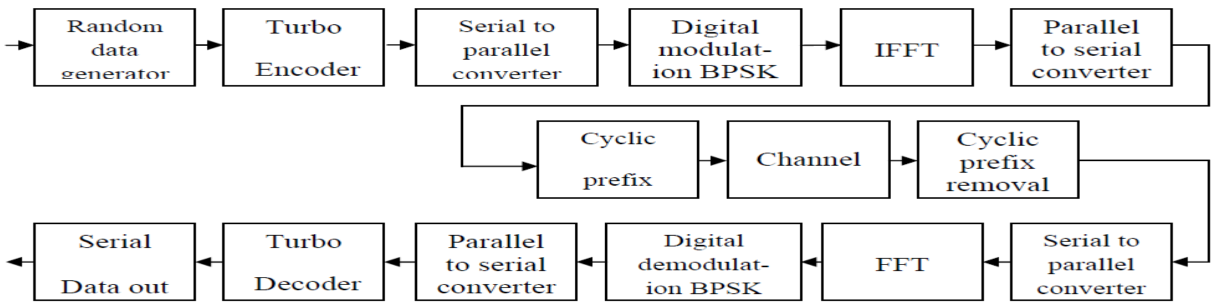


Fig 14 Turbo coded OFDM model used for simulation

IV.Conclusion

In this thesis BTC-OFDM uses the diversity of both t i e and frequency domains toachieve higher performance and more robustness against frequency and time selective fading. The benefits of coding gain of Turbo codes over Convolutional codes, provide an improvement in performance and efficiency of the proposed scheme over IEEE802.11a. Therefore, BTGOFDM has superior performance and efficiency over EEE802.11a standard. On the other hand, Turbo codes can eliminate the residual inter symbol interference (ISI) and inter channel interference (ICI) and therefore reduce the length of the required Cyclic prefix in an OFDM system. This decreases the overhead associated with the Cyclic Prefix. The use of Turbo codes in OFDM system for high data rate transmission in wireless LANs, results in a considerable improvement in terms of bit error rate performance and bandwidth efficiency.

Acknowledgement

This project is by far the most significant accomplishment is my life and it would be impossible without people who supported me and believed in me. I would like to thank all my friends for all the thoughtful and mind stimulating discussion we had. Last but not least I would like to express my gratitude to my parents, whose love and encouragement have supported me though out my education.

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