Block Coding Techniques implementation on DAB System based on OFDM

DISSERTATION-II

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Requirement for the award of the
Degree of

MASTER OF TECHNOLOGY

IN

Electronics and Communication Engineering

By

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Under the guidance of

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Abstract

Radio broadcasting technology has evolved over the last few years due to the ever increasing demands for as high quality sound services with ancillary data transmission in mobile environment. To accomplish this, a completely new digital radio broadcasting technology called the EUREKA-147 Digital Audio Broadcasting (DAB) system which improves the overall broadcasting performance by delivering near CD quality audio and data services in mobile receivers along with efficient use of the available radio frequency spectrum was developed by the members of European Broadcasting Union (EBU), the European Telecommunications Standards Institute (ETSI) and International Telecommunications Union (ITU-R).

A new digital radio technology for broadcasting radio stations that provides high-quality audio and data services to both fixed and mobile receivers was developed within the EUREKA-147 as Digital Audio Broadcasting (DAB). To combat the effect of multipath fading and ISI the system uses COFDM technology and it also makes it spectrally more efficient compared with existing AM/FM systems.

The issues related to system performance using concatenated coding technique, including the outer block codes, the inner convolutional code, or may be some alternatives of it.

CERTIFICATE

This is to certify that Vibhanshu Gupta bearing Registration no. 11009890 has completed objective formulation of thesis titled, "**Block Coding Techniques implementation on DAB System based on OFDM**" under my guidance and supervision. To the best of my knowledge, the present work is the result of his original investigation and study. No part of the thesis has been submitted for any other degree at any university.

The thesis is fit for submission and the partial fulfillment of the conditions for the award of Master of Technology (ECE).

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Date :

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I have taken many efforts in completing this paper. However, it would not have been possible without the kind support and help of many individuals. I would like to extend my sincere thanks to all of them.

With regards I would like to gratitude my guide "Mr. Avinash Rajoriya" who helped me in selecting my research topic "**Block Coding Techniques implementation on DAB System based on OFDM**" for my Dissertation-II and will be helping me out in completing my thesis report in future. Due to his proper guidance and eternal knowledge I was able to complete my literature review comfortably.

I must say thanks to all my friends who helped me out in completion of my literature review. I must say sorry for the errors if I have committed in my report.

At last I would like to thank my parents and to the almighty GOD for all their blessings.

Vibhanshu Gupta

Reg. no. – 11009890

DECLARATION

I hereby declare that the Dissertation-II work entitled "**Block Coding Techniques implementation on DAB System based on OFDM**" is an authenticate record of my own work carried out as a requirement of Dissertation-II work for the award of degree of Master of Technology in Electronics and Communication Department (ECE) from Lovely Professional University, Phagwara, Punjab under the guidance of **Mr. Avinash Rajoriya** during the month from January to April in the year 2015.

(Vibhanshu Gupta)

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Table of Contents

List of Figures

List of Tables

CHAPTER 1

INTRODUCTION TO SUBJECT

1.1 Introduction

In the whole world, Radio broadcasting is one of the most widespread electronic mass media comprising of hundreds of programme providers, thousands of high frequency transmitters and billions of radio receivers services. The market was widely spread by the Amplitude Modulation (AM) services in 1920s, when the broadcasting began. After that Frequency Modulation (FM) was invented which reigns the world till date. We live in the world of digital communication systems with the invention of FM services as it has many advantages over analog communication systems like storage capacity, reliability, quality of service, miniaturization and many more.

Today scientists of different countries are dreaming of to replace the existing AM and FM audio broadcasting services. For this dream they have come with a new technology Digital Audio Broadcasting (DAB) who has the capability to overcome the AM and FM technology services in many parts of the World in near future. This came down under in 1990s with the project named EUREKA-147 DAB project. DAB is suited to the places where multipath reception by the mobile receivers and inter symbol interference (ISI) is present and provides very high tolerance against multipath reception. For high frequency efficiency it allows the use of single frequency networks (SFNs). Implementation of digital broadcasting services using the DAB system has been already started in several countries in Europe and overseas, broadcasting organizations, network providers and receiver manufacturers. There are many new concepts have been adopted for digital radio broadcasting like Coded Orthogonal Frequency Division Multiplexing (COFDM), Perceptual audio coding (MPEG-2), provision for the multiplex of several programs and data transmission protocols.

In the early 19^{th} century radio services was started when AM radio came into existence, which was overcomed by FM radio which is used till now and for future scientist are working to implement a new digital radio system DAB which is a very innovative idea designed by the scientist using several coding techniques to work on and it is universal multimedia broadcast system which is very much expected that it will replace AM and FM radio services in future. This technique is very much efficient in power and spectrum of sound and data broadcasting. For this a project was started named EUREKA-147 DAB standard. It can be operated in any frequency band in VHF and UHF range for the terrestrial, satellite, hybrid (satellite and terrestrial), and cable broadcast networks. DAB receiver provides an unimpaired sound quality even when working in severe multipath conditions, such as in dense urban areas[5].

Not only high quality digital radio services (mono, two channel or multichannel stereophonic), it can provide ancillary data transmission (e.g. travel and traffic information, still and moving pictures, etc.). As compared to AM and FM systems, it requires lower transmitting power as well as it covers larger coverage area. DAB system has been demonstrated many times publicly during its development. In Europe and many places, it has been gone under many extensive computer simulations and field tests. In many European countries and all over the world it is now used as a regular services as FM. To pursue the introduction of DAB services in a concerted manner world-wide the European DAB Forum (Euro Dab) was established in 1995 and later in 1997 it became the World DAB Forum in 1997. Due to the developments in the project of EUREKA-147, the European Telecommunications Standards Institute (ETSI) approved the DAB standard or DAB specification in the form of EN 300401, which defines the characteristics of the DAB transmission signal, including audio coding, data services, signal and service multiplexing, channel coding and modulation.

Within a bandwidth of 1.536 MHz, DAB multiplexes several audio programs (of 24 KHz/48 KHz PCM), where the number of programs per ensemble is flexible and depends on individual program bandwidth requirements.

1.2 DAB – A Brief History

In early 1980s, the first sound broadcasting systems providing CD-like audio quality was developed using Satellite technology. The system was not suitable for mobile reception and employed very low data compression. The range of 10-12 GHz was used as frequency and that’s the reason DAB was not possible to provide service access to large number of listeners. An international research project was necessary to develop terrestrial digital sound broadcasting that would do the job (a new digital solution). An agreement was signed to cooperate in the development of a new standard by few organizations from France, United Kingdom and The Netherlands and Germany in 1986 and like this EUREKA-147 project was born.

EUREKA-147 project was later joined by the members of European Broadcasting Union (EBU), who were the part of work on the satellite delivery of digital sound broadcasting to mobiles within the frequency range between 1 and 3 GHz. Later the standardization process was started by the International Telecommunications Union (ITU-R) and European Telecommunications Standard Institutes (ETSI)[6].

There were some goals which designed for the DAB in the very beginning with the sole aim of audio quality for reception in mobiles, which are as follows:

- Transmitting power should be low.
- Digital audio services should be of high quality.
- Tuning of receivers is easy.
- Options for terrestrial, cable and satellite delivery.
- Even at high speed the mobile reception in vehicles are very well.
- Usage of frequency spectrum should be efficient.
- Comparing with AM and FM systems, have larger coverage area.
- For ancillary data transmission capacity.

To achieve the above mentioned goals, the transmission methods were proposed and they are as follows:

- One frequency-hopping system.
- One narrow-band system.
- One multicarrier OFDM system.
- Single carrier spread spectrum system.

Based on thorough simulation and field tests, the EUREKA-147 consortium alone started choosing the most appropriate transmission methods. The performance showed that the frequency hopping was too demanding as compared to network organization and the proposed narrow-band performed bad as compared to broadband solutions. The Coded Orthogonal Frequency Division Multiplexing (COFDM) system was chosen at the last finally as the spread-spectrum was not developed as hardware.

For EUREKA-147 project the problem was to select the audio coding standard. At that time for data compression of audio and video coding the Moving Pictures Expert Group (MPEG) was already been standardized. The EUREKA-147 proposed some solution with several other options from other countries, which were sent to the MPEG audio group for evaluation. The MPEG standardized the methods which were proposed by the EUREKA consortium as the performance offered by them was clearly superior. MPEG standardized them as MPEG Audio Layers-I, II and III. The decision took a lot of time to finalize the standard to be used for DAB. At the end Layer-II was chosen which is also known as MUSICAM.

The bandwidth consideration was also the most important specification to be specified. From a service area and network planning point of view, a very good performance in a multipath environment was shown by one transmitter having bandwidth 7 MHz of a TV channel which was too much inflexible. So reduction in the transmission bandwidth was very much necessary. For a DAB standard the appropriate bandwidth was fixed at 1.5 MHz[8]. A 7 MHz TV channel can be divided into four DAB blocks, where 5-7 programs were ensemble in each.

The very first DAB standard was developed in 1993 and then in 1995 the ETSI adopted as as the European standard for digital radio. Except USA and Japan, the EUREKA-147 DAB standard as digital radio is accepted worldwide. The strong opposition against the system requiring new spectrum and ensemble of several programs into one transmitter of the EUREKA-147 DAB system was rejected by the National Association of broadcasting, in USA. The USA uses the system approach named ‘In-Band-On- Channel (IBOC)’. The own developed national solution is shown is the solution named as Terrestrial Integrated Digital Broadcasting (ISDB-T).

1.3 What is DAB?

The delivery radio services from the source to the receiver id done by a new digital radio system named as Digital Audio Broadcasting (ISDB-T). DAB was functioned to work to deliver very high quality digital audio programs and data services to fixed, mobile and portable receivers which can use simple whip antennas. The EUREKA-147 project was responsible to develop it in 1990s. It gives very high robustness against multipath reception and that is very well suited for mobile reception. For high frequency efficiency it allows the use of SFNs.

The use of the COFDM technology in DAB transmission frame makes it resistant to multipath fading and ISI. By shadowing (i.e. the blocking or screening of the signals by tall buildings and hills which lie in the in the direction of the transmitter) and by passive echoes (after reflecting from tall buildings and hills the signal arrives at the receiver of delayed “multipath” signals) can be very affecting in the FM reception. While just only using the whip antenna DAB can tolerate all these types of interferences.

The signal fading is defined as the fading of the radio frequency signal amplitude at the receiver input which varies over time. Two types of fading occur slow fading and fast fading. The phenomenon of propagation of the signals the results in radio signal reaching the receiver antenna by two or more paths called as multipath, in wireless telecommunications. Ionospheric reflection and refraction are the two causes of multipath. Reflection occurs from the water bodies and terrestrial objects such as mountains, buildings, etc. The constructive and destructive interference and phase shifting of the signals are the two effects that affect the multipath which causes Rayleigh fading. The multipath effect is illustrated in the given below Fig. 1.1.

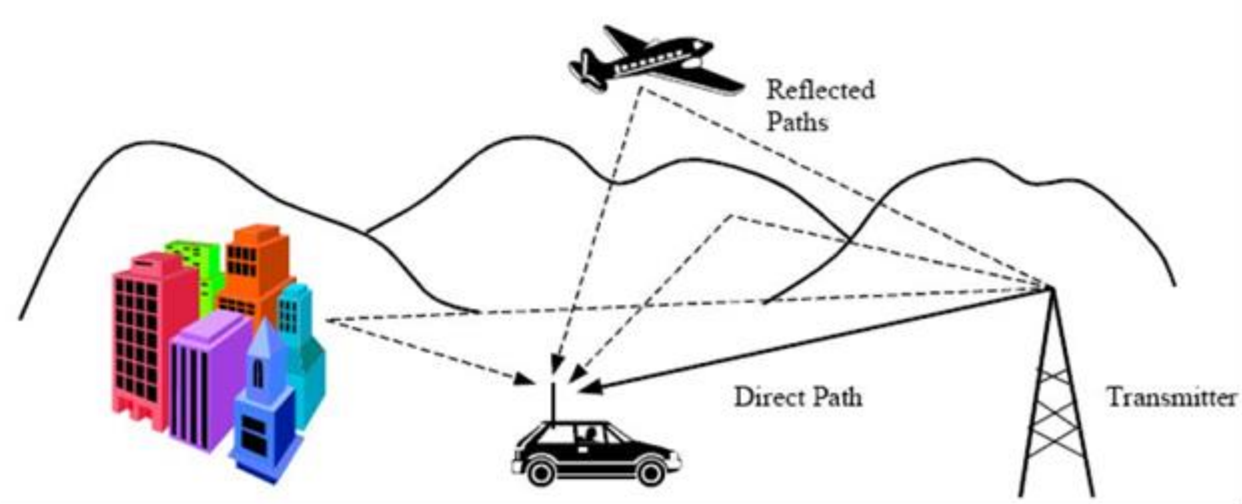

Fig. 1.1: Effect of Multipath on mobile receiver

The multipath effect which are shown above in the Figure 1.1 like scattering, reflection, Doppler spread, diffraction which is now not present in the DAB as it employs advance digital communication techniques like OFDM multicarrier modulation, rate-compatible punctured convolutional codes (RCPC) and time and frequency interleaving.

1.4 Advantages of DAB

The most significant advancement in radio broadcasting technology after the introduction of FM and AM is the EUREKA-147 DAB system. It provides various combinations of opportunities and benefits to the both listeners and the broadcasters. These various benefits and opportunities include the following points:

- Digital audio services provide high quality.
- The same receiver can receive both the data and music services.
- The DAB system had a universal and good standardized system outlook and wide range of various receiving equipment which includes stationary, movable and portable radio receivers.
- The radio frequency spectrum which is available can be used efficiently.
- The bit rates are between 8 and 384 Kbits/s which provides flexibility.
- A flexible multiplex configuration is combination of various incoming transmitted DAB services.

- Transmitting power should be low.
- As SFNs DAB transmitter networks can be designed.
- Transmitting power should be low.
- Comparing to AM and FM systems have larger coverage area.

CHAPTER 2

REVIEW OF LITERATURE

In the early 19^{th} century radio services was started when AM radio came into existence, which was overcomed by FM radio which is used till now and for future scientist are working to implement a new digital radio system DAB which is a very innovative idea designed by the scientist using several coding techniques to work on and it is universal multimedia broadcast system which is very much expected that it will replace AM and FM radio services in future. This technique is very much efficient in power and spectrum of sound and data broadcasting. For this a project was started named EUREKA-147 DAB standard. It can be operated in any frequency band in VHF and UHF range for the terrestrial, satellite, hybrid (satellite and terrestrial), and cable broadcast networks. DAB receiver provides an unimpaired sound quality even when working in severe multipath conditions, such as in dense urban areas.

Not only high quality digital radio services (mono, two channel or multichannel stereophonic), it can provide ancillary data transmission (e.g. travel and traffic information, still and moving pictures, etc.). As compared to AM and FM systems, it requires lower transmitting power as well as it covers larger coverage area. DAB system has been demonstrated many times publicly during its development. In Europe and many places, it has been gone under many extensive computer simulations and field tests. In many European countries and all over the world it is now used as a regular services as FM. To pursue the introduction of DAB services in a concerted manner world-wide the European DAB Forum (Euro Dab) was established in 1995 and later in 1997 it became the World DAB Forum in 1997. Due to the developments in the project of EUREKA-147, the European Telecommunications Standards Institute (ETSI) approved the DAB standard or DAB specification in the form of EN 300401, which defines the characteristics of the DAB transmission signal, including audio coding, data services, signal and service multiplexing, channel coding and modulation.

Within a bandwidth of 1.536 MHz, DAB multiplexes several audio programs (of 24 KHz/48 KHz PCM), where the number of programs per ensemble is flexible and depends on individual program bandwidth requirements.

2.1 Principle of DAB System

The conceptual block diagram illustrates the working principle of DAB system as shown in Figure 2.1. The details of important DAB system blocks are given in figure. The number of functional blocks is combined to form the overall DAB transmission system that processes the incoming input signal to produce the complete DAB transmitted signal.

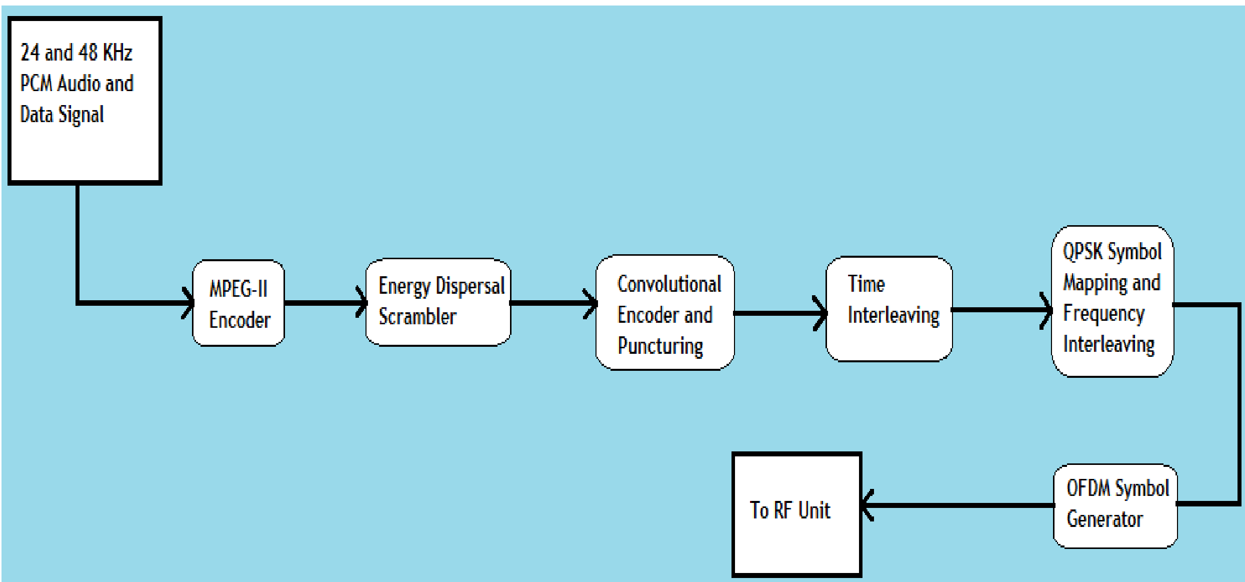

Fig. 2.1: Complete DAB Transmitter Block Diagram

The audio and data services are the analog signals and are MPEG layer-2 encoded and then scrambled (bits) are the inputs of the system. The individual inputs of the energy dispersal scramblers shall be scrambled by a modulo-2 addition with pseudo-random binary sequence (PRBS), just before the convolutional coding just in order to confirm appropriate energy dispersal in the transmitted signal as shown in the above block diagram. The output of the feedback shift register is defined as PRBS and it shall be using a polynomial of degree 9, which is defined as follows:

$$\boxed{P(X) = X^9 + X^5 + 1}$$

The scrambled bit sequence is now given to the block coding techniques also known as forward error correction (FEC) which usually employs puncturing sequence to puncture it (reducing some bits using puncturing sequence), so using punctured convolutional codes whose code-rates should in the range of 0.25-0.88. In the Main Service Multiplexer, the input is the encoded bit-stream (encoded by FEC) which is then passed to time –interleaving block to make the bits time interleaved and then multiplexed with other programs which forms the Main Service Channel (MSC). The complete DAB frame is then formed by mixing or combining the output of the multiplexer and the service information in the Fast Information Channel (FIC).

Then with frequency interleaving of subcarriers Differential QPSK (DQPSK) modulation takes place and then it is passed to the Inverse Fast Fourier Transform which is basically a OFDM signal generator and that forms the final DAB transmission signal.

2.2 Technical Overview

For robust reception by mobile, portable and fixed receivers, using simple antennas which is intended mainly by the EUREKA-147 DAB system is very reliable, multi-programme, digital radio broadcasting system. The complete specification for the DAB transmitted signal is specified by ETSI, EN 300401.

The three main elements of the EUREKA-147 DAB system are as follows:

2.2.1 Source Coding

Source coding basically employs MUSICAM (Masking Pattern Universal Sub-band Integrated Coding and Multiplexing) audio coding which uses the principle of psycho acoustical masking as specified for MPEG-2 Audio Layer-II encoding. According to the psycho acoustical model, MUSICAM codes only audio signal components that ear will listen, and discards any audio component that the ear will not be able to listen, this is the basic principle of MUSICAM.

While preserving the subjective quality of the digital audio signal, this technique allows a bit rate reduction from 768 Kbits/s down to about 100 Kbits/s per mono channel. This allows the DAB to provide high quality of sound to the listeners and to use the bandwidth more efficiently.

The resulting DAB audio frame corresponds to 24 ms duration of audio in accordance with the ISO/IEC 11172-3 Layer II format standard, for a sampling frequency of 48 KHz. The resulting audio frame corresponds to 48 ms duration of audio in accordance with the ISO/IEC 13818-3 Layer II LSF format standard, for a sampling frequency of 24 KHz.

2.2.2 Channel Coding, Multiplexing and Transmission Frame

In the choice of the proper error protection for different applications and for different physical transmission channels, the DAB system allows great flexibility. It is possible to use codes of different redundancy in the transmitted data stream in order to provide ruggedness against transmission distortions, without the need for different decoders using rate-compatible punctured convolutional codes.

Equal Error Protection (EEP) and Unequal Error Protection (UEP), matched to bit an error sensitivity characteristic which is allowed by channel coding and that is based on punctured convolutional forward error correction. The UEP can be used for data but it is primarily designed for audio. The EEP can be used for both the audio as well as for the data. The basic idea of RCPC channel coding is that to generate the mother code first and then by omitting the certain redundancy bits the daughter codes will be generated.

In channel coding, the constraint length of the convolutional code is 7. The provided generator polynomials in the octal forms are 133,171,145 and 133 respectively. The encoder made of the shift register is shown below in the figure 2.2.

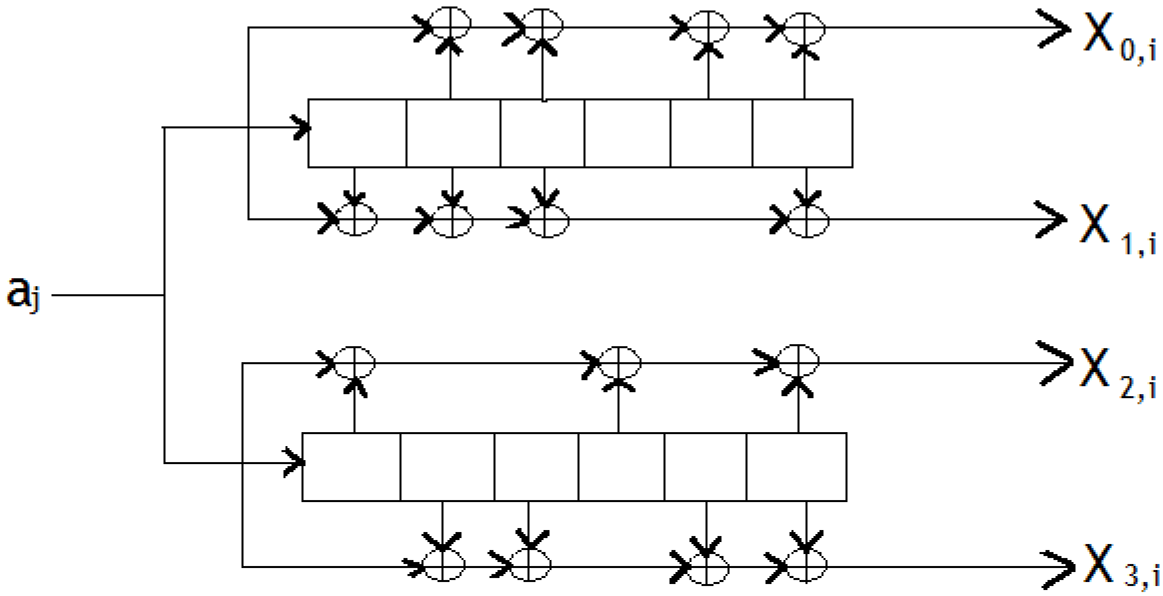

Fig. 2.2: Channel encoder for the DAB mother code

For each data bits a_i the encoder will produce four coded bits $X_{0,i}$, $X_{1,i}$, $X_{2,i}$, $X_{3,i}$, that means mother code has a code rate R=1/4. The mother code is defined by:

$X_{0,i} = a_i \oplus a_{i-2} \oplus a_{i-3} \oplus a_{i-5} \oplus a_{i-6}$

$X_{1,i} = a_i \oplus a_{i-1} \oplus a_{i-2} \oplus a_{i-3} \oplus a_{i-6}$

$X_{2,i} = a_i \oplus a_{i-1} \oplus a_{i-4} \oplus a_{i-6}$

$X_{3,i} = a_i \oplus a_{i-2} \oplus a_{i-3} \oplus a_{i-5} \oplus a_{i-6}$

for i = 0, 1, 2, …., i+5.

By applying appropriate puncturing vectors given in the DAB standard a code rate of ½ and 1/3 can be achieved. UEP of data stream possibility can be provided by RCPC codes. By using RCPC codes it is possible to save capacity and add as much redundancy necessary. UEP is particularly useful for MPEG-1 and MPEG-2 Audio Layer- II data. They are organized in frames of 24 ms. all these data rates have protection profiles that are grouped into five protection levels PL1 to PL5. PL5 is the least robust but PL1 is the most robust one protection level. Except PL5 all protection levels are designed for mobile reception.

The individual program are initially encoded, error protected by applying FEC and then time interleaved. By using Multiplexing Process, a single data stream for transmission is combined by these outputs. For multiplexing, the programs ensemble with a bandwidth of 1.536 MHz.

The Figure 2.3 illustrates the structure of DAB signal frame in efficient signal synchronization is given below:

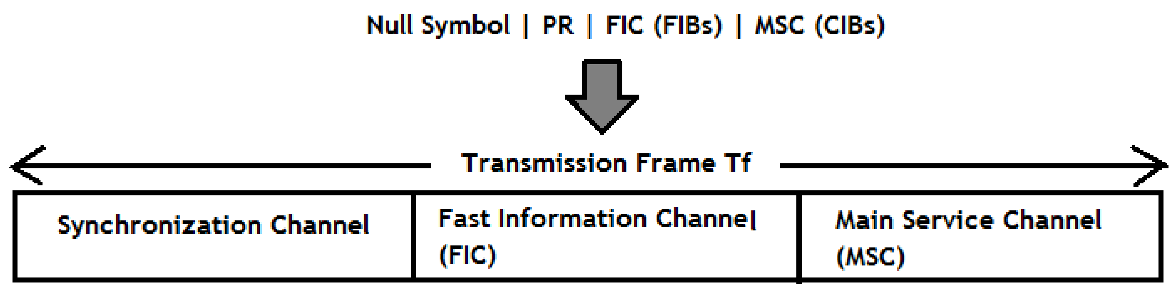

Fig. 2.3: DAB Transmission Signal Frame structure

Above figure illustrates that the first symbol is the synchronization channel which consists of Null symbol and phase reference symbol. The second symbol is the FIC channel and the last symbol is the MSC channel. The useful payload of the DAB frame is formed by the MSC.

In above structure of the frame: PR = Phase Reference Symbol

FIC = Fast Information Channel

FIB = Fast Information Bit = 256 bits

MSC = Main Service Channel

CIF = Common Interleaved Frame = 864 CU = 55296 bits

CU = Control Unit = 64 bits

The period T_F of each DAB transmission frame is either the same as the MPEG-1 and MPEG-2 Audio Layer II frame length of 24ms or an integer multiple of it. The transmission frame length is 24ms. The synchronization channel is constituted by the first two OFDM symbols[6]. The FIC carries the next 3 OFDM symbols which carry the information regarding multiplex configuration and transmitted programs. The MSC carries the next 72 OFDM symbols. The transmission frame structure is shown below:

1) *Synchronization Channel*: The two symbols i.e.; Null Symbol (during which no information is transmitted) and Time Frequency Phase Reference (TFPR) symbol together constitutes synchronization channel, which has predetermined modulation. For basic demodulator functions it is used internally within the transmission system, such as transmission frame synchronization, automatic frequency control, channel state estimation, and transmitter identification.

2) *Fast Information Channel:* It is used to signal the multiplex configuration of the DAB transmission and service information. To decode any of the sub-channels instantly, it has fixed symbols which are known to the receivers. The FIBs together make up the FIC. The 256 bits equals to the 1 FIB. The data of the FIC is non-time-interleaved channel with fixed equal error protection (code rate 1/3).

3) *Main Service Channel:* To carry audio and data service components this channel is used. The MSC is a time interleaved data channel divided into a number of sub-channels which are individually convolution coded, with equal and unequal error protection. One or more service components are carried by each sub-channel. The CIFs together make up the MSC. The multiplexed configuration is defined as the organization of sub-channels and service components. The MSC of the DAB system has a gross capacity of 2.304 Mbps.

2.2.3 COFDM Modulation

In the European EUREKA-147 standard the main advantage of the DAB system was its ability to deliver high quality audio (near CD quality) services to mobile receivers under different channel conditions. This is due to the effect of a transmission technology called Coded Orthogonal Frequency Division Multiplexing (COFDM). The system was designed in such a way that can exploit both time and frequency diversity by combining multi-carrier modulation technique OFDM (Orthogonal Frequency Division Multiplexing) with convolutional channel coding, which is done by COFDM modulation. To achieve this prior to transmission data symbols are interleaved in the time and frequency domains.

2.3 OFDM Theory

The mobile channel is always affected by the multipath fading environment, due to which at the receiver, the received signal contains not only the direct line-of-sight (LOS) wave but also a large number of reflected waves that arrives at different times. The delaying of signals are due to the three process that takes place in the multipath fading environment, Reflection, Diffraction and Scattering from trees, buildings, mountains, vehicles, any sharp edges, particles in the environment and water bodies. There is loss of information or data which is termed as Inter-symbol Interference (ISI) when these reflected delayed waves interfere with the direct wave and there is degradation of network performance.

To carry more than one signal over a telephone line Frequency Division Multiplexing (FDM) was used for a long time. In FDM the channel bandwidth is divided into different sub-channels and then multiple relatively low rate signals are transmitted by carrying each signal on a separate carrier frequency. Guard bands are necessary to confirm that the signals of one sub-channel did not overlap with the signal from an adjacent one and it is an loss of spectrum and therefore bandwidth.

OFDM is expanded as Orthogonal Frequency Division Multiplexing which was proposed to overcome the problem of the multipath fading environment and bandwidth efficiency. Modulation and Multiplexing togetherly combines to form OFDM. OFDM is derived on a scheme to reduce the effect of multipath fading and transmit the data parallely and make the use of complex equalizers unnecessary, and the fact that the high bit stream data is transmitted over large number sub-carriers (obtained by dividing the available bandwidth) each of a different frequency and these carriers are orthogonal to each other.

Frequency Selective Fading channel is converted into N flat fading channels by OFDM, where N is the number of sub-carriers. The orthogonality between the two signals are carried by keeping

keeping the carrier spacing multiple of $1/T_s$ by using fourier transform methods, where T_s is the symbol duration.

The below given equation is for giving OFDM symbol in baseband,

$$x(t) = \sum_{k=0}^{N-1} (dk\, e^{\frac{j2kt}{T}}) \qquad \text{If } 0 < t < T$$

The block diagram of OFDM transmitter and receiver in given below in Fig 6:

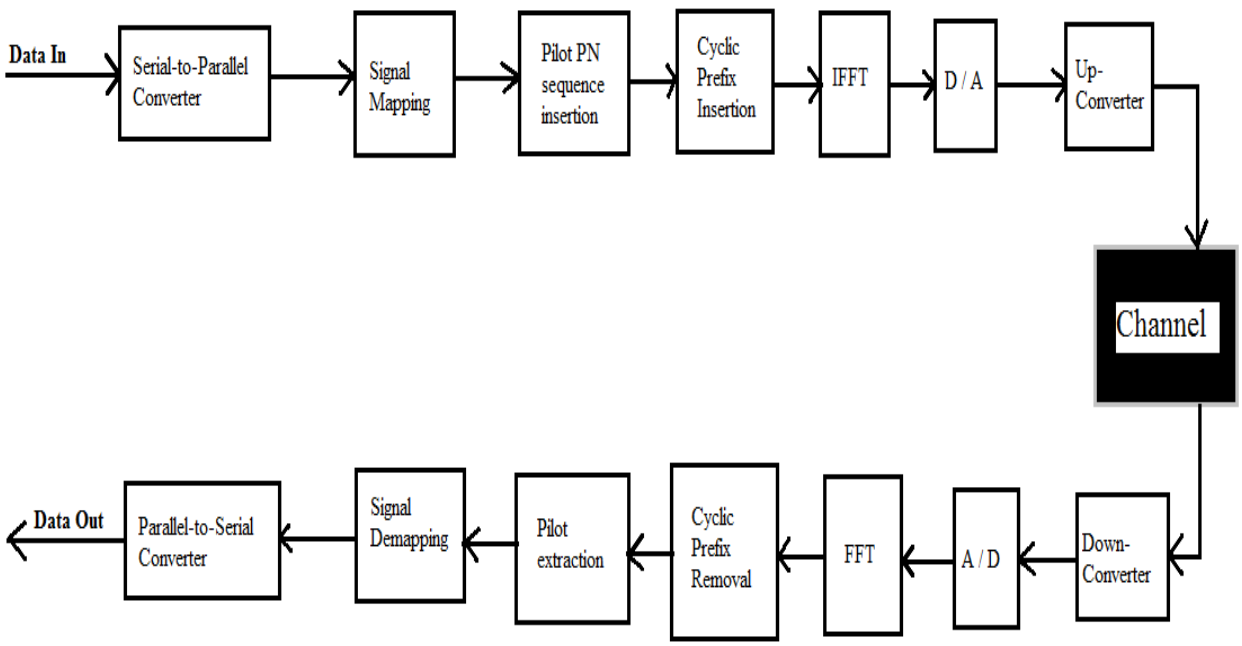

Fig. 2.4: Basic Block Diagram of OFDM system

2.3.1 Interpretation of IFFT and FFT

The DFT can be calculated very efficiently using Fast Fourier Transform. It can be implemented at fairly low cost in integrated circuits. IFFT and FFT are the heart of OFDM technology. The implementation cost has drastically reduced in OFDM with the advancement in VLSI and DSP technology. The complexity of performing an FFT is dependent on the size of the FFT. The N-point DFT can be calculated by using this formula:

$$x(k) = \sum_{n=0}^{N-1} x(n).e^{\frac{-j2\pi nk}{N}}$$

where k=0,1,2,…..N

It requires N^2 complex multiplications and N*(N-1) complex additions whereas use of FFT algorithm reduces the number of computations to the order of $N/2*\log_2(N)$ complex multiplications and $N*\log_2(N)$ additions. The number of sub-carriers are usually kept as the power of 2 as FFT algorithm works efficiently when N is a power of 2.

IFFT/FFT operation ensures that sub-carriers do not interfere with each other. To obtain the time domain samples of the multicarrier signal the IFFT is used at the transmitter. The data sent on individual sub-carriers is retrieved by the FFT which is its most important use. Therefore OFDM had a very simple implementation capability.

2.3.2 Guard Time and Cyclic Prefix

In OFDM technology, to add guard interval between OFDM symbols is done to overcome the problem of multipath fading environment where inter symbol interference (ISI) occurs. The guard interval is formed by a cyclic continuation of the signal characteristics so the information in the guard interval is actually present in the OFDM symbol. The system becomes robust against multipath delay spread by implying guard interval between two OFDM symbols[2]. The concept behind making the guard channel is to actually taking the last portion of the OFDM symbol and then placing it at the start of the symbol as shown in the below diagram:

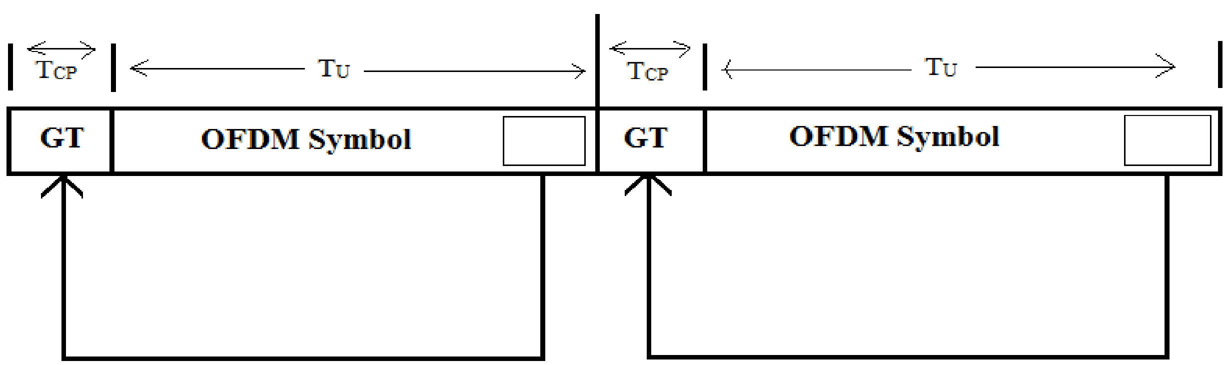

Fig. 2.5: Guard Time and Cyclic Prefix

where T_U = the OFDM symbol time without guard interval

T_{CP} = the duration of the copied information in the guard interval using cyclic prefix.

The last portion of the symbol is appended to the front and transmitted during the guard time. Therefore the total symbol time during transmission, $T_S = T_U + T_{CP}$

There will be ISI in the result if the guard time needs to be greater than maximum delay spread. If guard time interval is used in the DAB than the need of pulse shaping filter is eliminated, and it reduces the sensitivity to time synchronization problems. The orthogonality between the sub-carriers is maintained by the cyclic prefix i.e; in multipath environment converting linear convolution to the circular convolution. The use of cyclic prefix also avoids the Inter Channel Interference (ICI).

2.3.3 Error Control Coding

To improve the signal communication performance by increasing the robustness against channel impairments like noise, interference, fading, etc. is done by error control coding or channel coding by transforming the signal. In this process the original data is appended by the redundant bits. A function of signal-to-noise ratio at the receiver input and the information rate for a particular signaling scheme is the probability of error.

Three broad classifications if error control coding is as follows:

1) *Automatic Repeat Request (ARQ)* → It is only error detection codes for full duplex. In ARQ method, when an error had occurred and it is detected of a received signal, the receiver sends a request to the transmitter for retransmission. The time delays will occur due to request and repeat signals making ARQ inappropriate for real time systems. Therefore there is an disadvantage of these codes is the time delay.

2) *Forward Error Correction (FEC)* → These are the simplex codes. They are having inbuilt capability of error detection as well as error correction. As compared to ARQ here we have advantage of time saving as there is no re-transmission of data. It is widely used in the real

time system applications such as DAB, DAB-T, WiMAX, etc. Examples: Linear Block Codes, Cyclic Codes, Convolutional Codes.

3) *Hybrid ARQ (ARQ + FEC)* → these are the full duplex codes. Here also error detection as well as error correction takes place. It is the combination of both above 2 classifications FEC and ARQ. Compared to simple ARQ it performs better in poor signal conditions.

The OFDM symbols are generated after punctured convolutional coding. This is why it is named "Coded" in COFDM. If the incoming bits are random in nature than error correction process works well. In wireless channel, to help FEC to work properly and combat the effect of deep fades that may occur that's why time interleaving and frequency interleaving is applied to the coded bits.

The coding gain is given by the following general equation:

$$\boxed{\text{Gain[dB]} = (E_b/N_o)_{uncoded}[\text{dB}] - (E_b/N_o)_{coded}\,[\text{dB}]}$$

2.4 DAB Transmitted Signal

The DAB transmission signal is built up according to the frame structure consisting of synchronization channel, the FIC and the MSC which is presented in Fig 4. The transmitted frame duration is denoted by T_F. A sequence of OFDM symbols together makes up each transmission frame. The number of OFDM symbols in a transmission frame depends on the transmission modes which will be explained in next section. The synchronization channel reserves the first two OFDM symbols in each transmission frame.

The first OFDM symbols of the transmission frame should be a Null symbol of duration T_{NULL} and the remaining part of the frame to be made of OFDM symbols of duration T_S which is defined by the DAB standard. Each of these OFDM symbols had sets of equally spaced carriers, with carrier spacing $1/T_U$.

The following formula defines the main DAB transmitted signal s(t):

$$s(t) = Re\left\{e^{j2\pi fct}\sum_{m=-\infty}^{\infty}\sum_{l=0}^{L}\sum_{k=-\frac{k}{2}}^{\frac{k}{2}} Z\, m, l, k \;\times\; \mathrm{g}k, \mathrm{l}(t - mTf - Tnull - (L-1)Ts)\right\}$$

with

$$\mathrm{g}k, l(t) = \left\{e^{\frac{2j\pi k(t-\Delta)}{Tu}} Rect\left(\frac{t}{Ts}\right)\right\} \qquad\qquad for\ l = 1,2, \dots \dots.., L$$

But $g_{k,l} = 0$ for l = 0 and $T_S = T_U + \Delta$

where

L → the number of OFDM symbols per transmission frame (the Null symbol being excluded)

K → the number of transmitted carriers

T_F → the transmission frame duration

T_{Null} → the Null symbol duration

T_S → the duration of the OFDM symbol of indices l=1, 2, 3,……, L

T_U →the inverse of carrier spacing

Δ → the duration of the time interval called guard interval

$Z_{m,l,k}$ → the complex D-QPSK symbol associated with carrier k of OFDM symbol 1 during transmission frame m. For k=0, $Z_{m,l,k}$=0, so that the central carrier is not transmitted.

F_c → the central frequency of the signal.

2.5 DAB Modes and System Parameters

The EUREKA-147 DAB system has four transmission modes of operation named as mode-I, mode-II, mode-III, and mode-IV, each having its particular set of parameters. The network configuration and operating frequencies are the depending parameters to choose the transmission modes. Due to this the DAB system operate over a wide range of frequencies from 30 MHz to 3GHz[8].

As it was discussed above that FICs that are made up of FIBs and MSCs that are made up of CIFs together makes up the DAB transmission frame. The details of the number of FIB’s and CIFs for each transmission modes in shown in Table 2.1.

Table 2.1 DAB Transmission Frame Composition

Mode-I	96ms	12	4
Mode-II	24ms	3	1
Mode-III	24ms	4	1
Mode-IV	48ms	6	2

Transmission Mode – I → It is designed for a large coverage area. It is suited for single frequency networks (SFNs) operating at frequencies below 300 MHz (VHF Band - III).

Transmission Mode – II → It is designed principally for Terrestrial DAB for small to medium coverage areas at frequencies below 1.5 GHz (UHF L-Band).

Transmission Mode – III → It is available for satellite broadcasting below 3 GHz (UHF L-Band).

Transmission Mode – IV → It is used for seamless coverage of large areas by means of SFNs operating in the L-Band. The parameters of Mode-IV lie between those of Mode-I and Mode-II.

The details of DAB system parameters for all the four transmission modes are shown in Table 2.2. The values of time related parameters are given as multiples of the elementary period T=1/2048000 seconds. All the four DAB modes have same signal bandwidth of 1.536 MHz.

Table 2.2: System parameters of the four DAB transmission modes

K	1536	384	192	768
L	76	76	153	76
T_F	196608 T 96ms	49152 T 24ms	49152 T 24ms	98304 T 48ms
T_{NULL}	2656 T 1297ms	664 T 324µs	345 T 168µs	1328 T 648µs

T_S	2048 T 1ms	638 T 312µ	319 T 156µs	1276 T 623µs
T_U	504 T 1ms	512 T 250µs	256 T 125µs	1024 T 500µs
Δ	504 T 246µs	126 T 62µs	63 T 31µs	252 T 123µs
Maximum Radio Frequency	375 MHz	1.5 GHz	3 GHz	750 MHZ
Carrier Spacing	1 KHz	4 KHz	8KHz	2KHz
FFT Length	2048	512	256	1024

2.6 Base Paper Read

Performance prediction of OFDM based DAB system using Block coding techniques

Radio services was started when AM radio came into existence, which was overcomed by FM radio which is used till now and for future scientist are working to implement a new digital radio system DAB which is a very innovative idea designed by the scientist using several coding techniques to work on and it is universal multimedia broadcast system which is very much expected that it will replace AM and FM radio services in future. This technique is very much efficient in power and spectrum of sound and data broadcasting. For this a project was started named EUREKA-147 DAB standard. It can be operated in any frequency band in VHF and UHF range for the terrestrial, satellite, hybrid (satellite and terrestrial), and cable broadcast networks. DAB receiver provides an unimpaired sound quality even when working in severe multipath conditions, such as in dense urban areas[1].

The audio and data services are the analog signals and are MPEG layer-2 encoded and then scrambled (bits) are the inputs of the system. The individual inputs of the energy dispersal scramblers shall be scrambled by a modulo-2 addition with pseudo-random binary sequence (PRBS), just before the convolutional coding just in order to confirm appropriate energy dispersal in the transmitted signal as shown in the above block diagram. The output of the feedback shift register is defined as PRBS and it shall be using a polynomial of degree 9. The scrambled bit sequence is now given to the block coding techniques also known as forward error correction (FEC) which usually employs puncturing sequence to puncture it (reducing some bits using puncturing sequence), so using punctured convolutional codes whose code-rates should in the range of 0.25-0.88. In the Main Service Multiplexer, the input is the encoded bit-stream (encoded by FEC) which is then passed to time –interleaving block to make the bits time interleaved and then multiplexed with other programs which forms the Main Service Channel (MSC). The complete DAB frame is then formed by mixing or combining the output of the multiplexer and the service information in the Fast Information Channel (FIC). Then with frequency interleaving of subcarriers Differential QPSK (DQPSK) modulation takes place and then it is passed to the Inverse Fast Fourier Transform which is basically a OFDM signal generator and that forms the final DAB transmission signal[1].

In this paper, the Bit Error Rate has been considered as the performance index in all analysis.

CHAPTER 3

PRESENT WORK

3.1 Problem Formulation

The detailed technique for modeling DAB transmission and reception system which is described in detail in previous chapter is done in this chapter. All the standard parameters have been selected are of Transmission Mode – II as this mode has been selected for modeling. All the simulation work has been done in the baseband transmission and frame based processing is done.

Software used for the simulation of DAB system is SIMULINK, which also provides the MATLAB environment to perform the computationally intensive task to work faster than any other language.

3.2 DAB Simulation Model

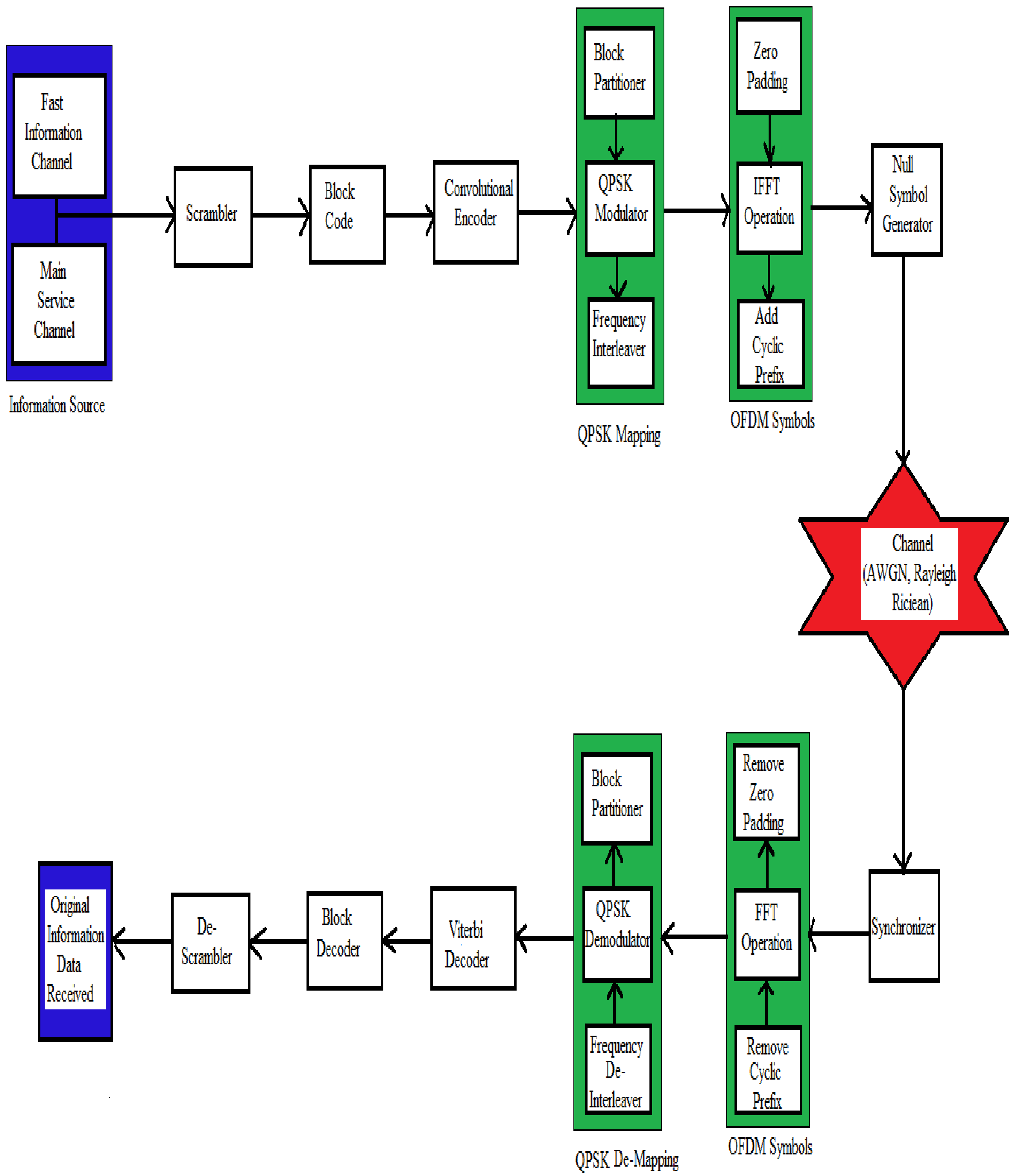

Fig. 3.1: Implemented DAB Block Diagram

3.2.1 Information Source

In the DAB system model in the transmitter section this is the first block which generates the randomly binary data bit sequence for FIC and MSC. As discussed in above section, 76 OFDM symbols are present in each frame of Transmission Mode – II where first OFDM symbols is reserved for phase reference symbol, next 3 OFDM symbols are for FIC channel and the last 72 OFDM symbols are MSC channel. Each FIC has 3 FIBs and only 1 CIF for MSC data for transmission frame of 24 ms. There are 2 bits per carrier in QPSK symbol mapping and no. of subcarriers are 384. Therefore no. of bits per OFDM symbol are 768(384*2)[5].

FIC_length= no. of OFDM symbol*bits/OFDM symbol = 3*768 => 2304 bits

MSC_length= no. of OFDM symbol*bits/OFDM symbol = 72*768 => 55296 bits

Tx_bits = FIC_length + MSC_length = 2304+55296 => 57600 bits

In SIMULINK we use "bernoulli binary generator" to generate random data bits for both FIC and MSC with an equal probability of 0 and 1. Then these both blocks are concatenated using "matrix concatenator" to make the output as Tx_bits.

3.2.2 Scrambler

The Tx_bits are scrambled by a modulo-2 addition with a pseudo random binary sequence, before convolutional encoder. The output of the feedback shift register is defined as PRBS and it shall be using a polynomial of degree 9, which is defined as follows:

$$P(X) = X^9 + X^5 + 1$$

which makes the scramble polynomial as [1 0 0 0 1 0 0 0 1].

This is done by taking the block of "Scrambler" and setting parameters of base and scramble polynomial.

3.2.3 Block Coding

The scrambled bit is now transmitted to the Forward Error Correction (FEC) block coding techniques prior to convolutional encoder. The principle of block coding techniques is to split the message bits into blocks, each of k bits and referred to as data words. Then we add certain bits at the end of data words to encode it which is called as redundant bits, r.

$$n = k + r$$

Here n represents the codewords after adding data and redundant bits.

The simulation is been done with Linear Block Codes (LBC), Hamming Codes, Cyclic Codes and the improvement is shown by implementing Bose-Chaudhary Hacquenghem (BCH) Codes. These codes are been implemented by taking

3.2.4 Convolutional Encoder

The block encoded data stream of bits is now input to convolutional encoder. In channel coding, the constraint length of the convolutional code is 7. The provided generator polynomials in the octal forms are 133,171,145 and 133 respectively. The mother code has a code rate R=1/4 which specifies each bit is covered by 4 bits.

The “Convolution Encoder” is taken from the communication toolbox where trellis structure is made using poly2trellis function and the puncturing vector can also be written is there is any.

Using both the Block codes and Convolution codes we used a concept of concatenated coding where Block codes and Convolutional codes are outer code and inner encoder respectively. This technique also improves the BER performance of the channels.

At the decoder side we have Block decoder and Viterbi decoder as the outer decode and inner decoder.

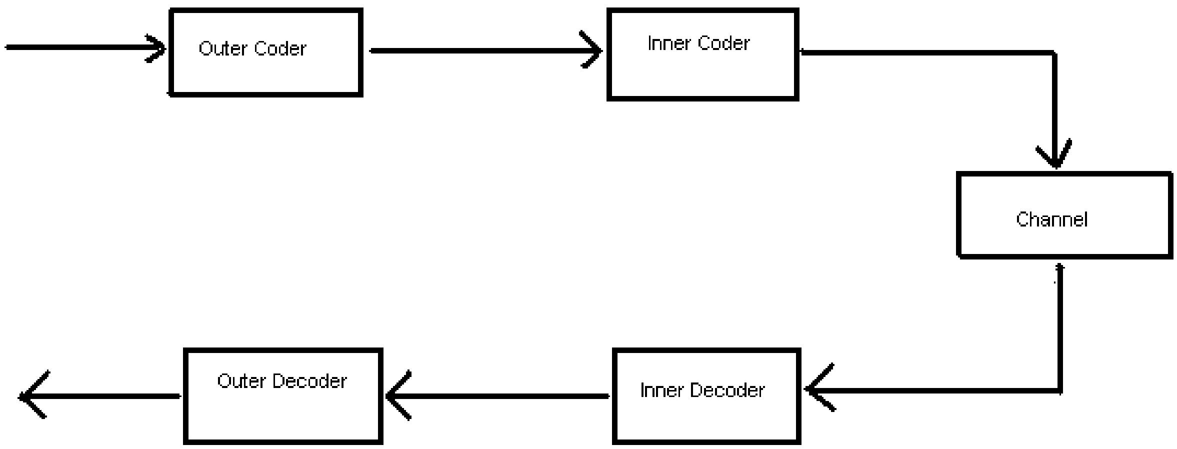

Fig. 3.2: Block Diagram of Concatenated Coding

3.2.5 Block Partitioner

This block is used to split the convolutional code bits at its input into blocks of data which are OFDM symbols. The 57600 bits of convolutional encoder is splited into 75 consecutive blocks of 768 bits. These blocks are OFDM symbols.

3.2.6 QPSK Modulator

The mapping of serial bit stream in each data block into digital constellation according to QPSK modulation scheme is done by using QPSK Modulator. In DAB standard, it is defined in a equation as:

$$q_{l,n} = \frac{1}{\sqrt{2}}\left[\left(1 - 2b_{l,n} + j\left(b_{l,n+k}\right)\right)\right]$$

where n = 1,2,…………..k and l=2,3,4,………..76 and k is the no. of carrier used.

One OFDM symbol of time period T_s, each data block of 768 bits is mapped to the 384 complex coefficients where the real parts of the 384 QPSK symbols will be mapped by the first 384 bits and the imaginary parts will be mapped by the last 384 bits. The 75 blocks of 384 bits is the complex output of this block.

Without adding noise the simulated QPSK constellation mapping is shown by:

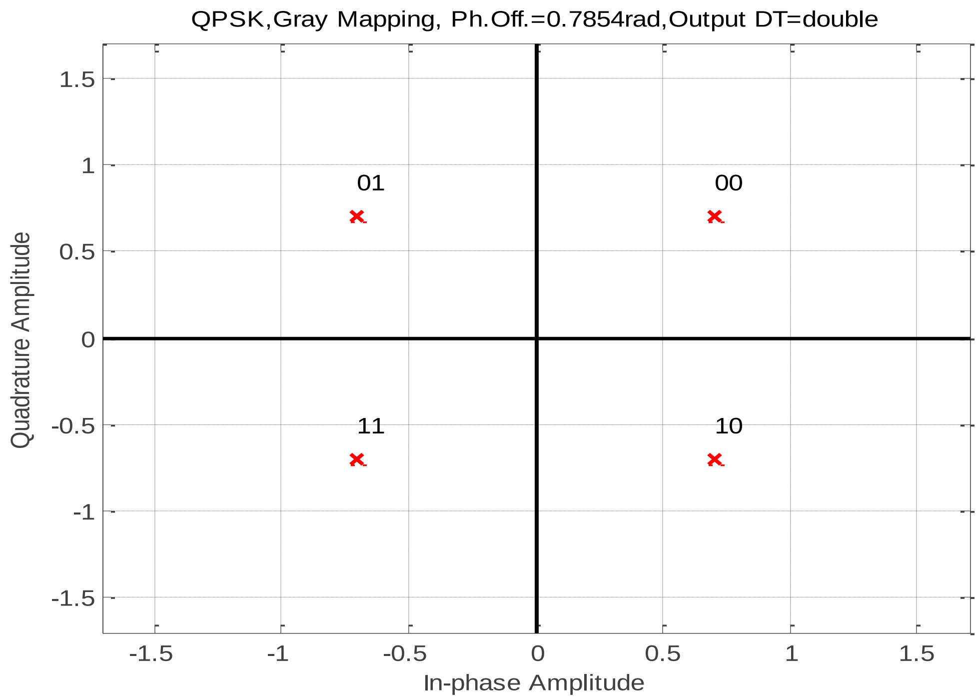

Fig. 3.3: QPSK Block Diagram

Addition of AWGN noise the QPSK constellation plot is shown as:

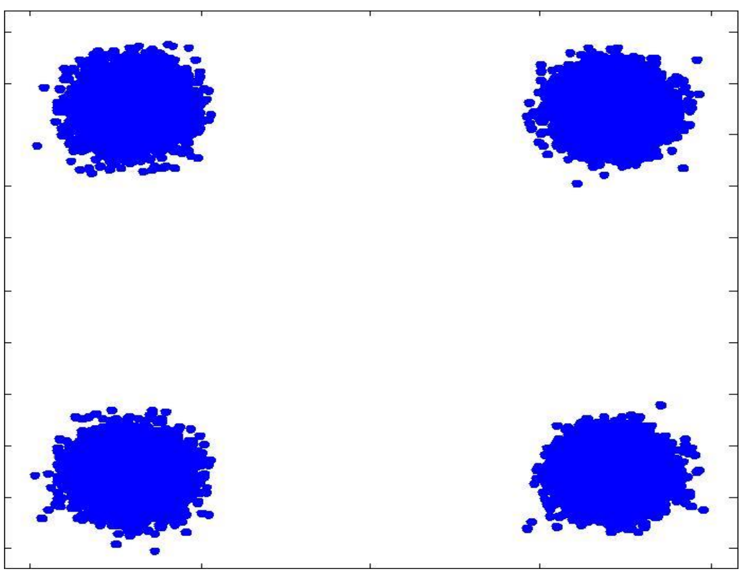

Fig. 3.4 QPSK Symbol Constellation with AWGN noise

This modulator is found in Communication toolbox in Modulation library named as "QPSK Modulator Baseband". In its properties change input type to bits and gray coding.

3.2.7 Frequency Interleaving

The effects of selective fading are removed by frequency interleave block. In the wireless channel, it offsets any deep fades that occur by spreading the data bits over the sub-carrier channel. It defines the correspondence between the index n of the QPSK symbols q(l, n) and the carrier index $k\left(-\frac{k}{2} \leq k < 0 \; and \; 0 < k \leq \frac{k}{2}\right)$. According to the following relation the QPSK symbol must be reordered:

$$y(l,k) = q(l,n)$$

for l = 2, 3, 4, ……, L and with k = F(n)

Let *Π* (i) be a permutation in the set of integers i = 0, 1, 2, ……, 511 obtained from the following relation :

$$\pi(i) = 13\,\pi(i-1) + 127\ (mod\ 512) \qquad and \qquad \pi(0) = 0$$

for i = 1, 2, …….511

$$d_n = \Pi\ (i) \quad \text{(excluding 256)}$$

Between QPSK symbols and carrier index the frequency interleaving rule is defined as,

$$k = F(n) = d_n - 256$$

The table of interleaving rule is defined below:

Table 3.1 Frequency interleaving rule for TM-II

i	Π(i)	d_n	n	k
0	0			
1	127	127	0	-129
2	242	242	1	-14
3	201	201	2	-55
4	180	180	3	-76
5	419	419	4	163
6	454			
7	397	397	5	141
8	168	168	6	-88
9	263	263	7	7
10	474			
11	145	145	8	-111
12	476			

13	171	171	9	-85
14	302	302	10	46
15	469			
16	80	80	11	-176
17	143	143	12	-113
18	450			
. . .	. . .	. . .	. . .	. . .
508	140	140	380	-116
509	411	411	381	155
510	350	350	382	94
511	69	69	383	-187

No Interleaving

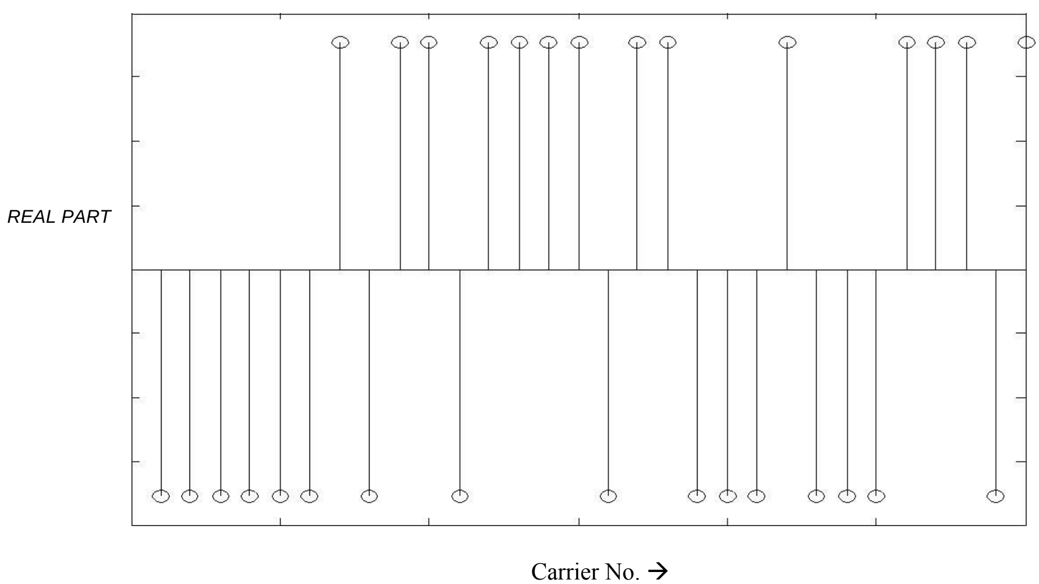

Fig. 3.5 without Frequency Interleaving

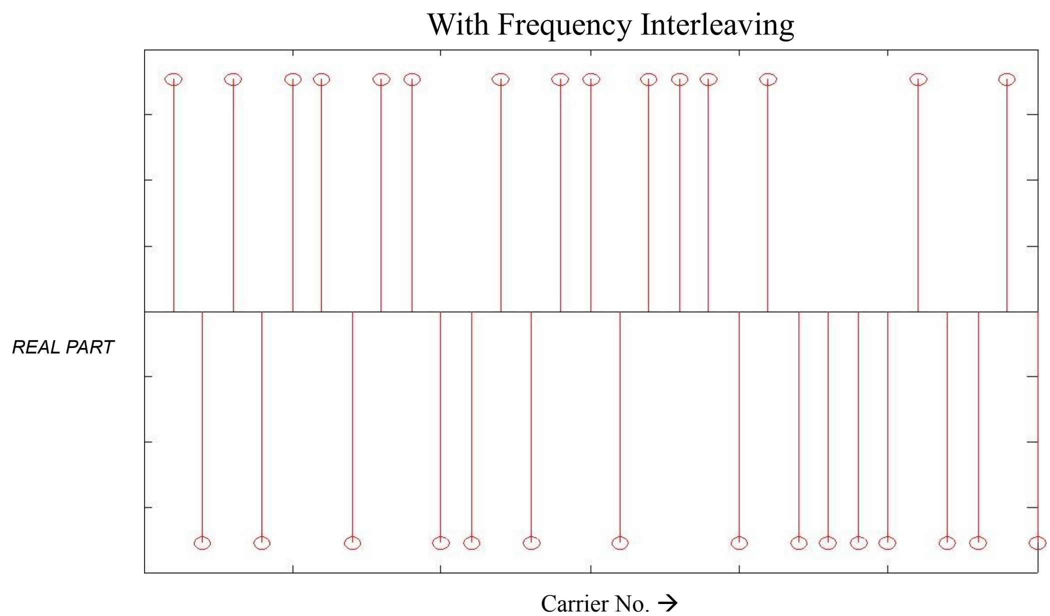

Fig. 3.6 with Frequency Interleaving

3.2.8 OFDM Symbol Generator

The heart of the DAB system is this block. IN DAB this OFDM technology makes it robust against multipath fading environment delivering high quality of audio services. OFDM symbol generation will be explained in this section.

a) Zero Padding → If the number of sub-carriers is power of two the IFFT/FFT algorithm works efficiently. The output from the previous block has the length of 384. The FFT length of Transmission mode-II is 512. So to make it power of two zero padding is necessary (i.e; from 384 to 512). This sub-block adds 128 zeros to each symbol block so as to work with 512 FFT_length which is shown below.

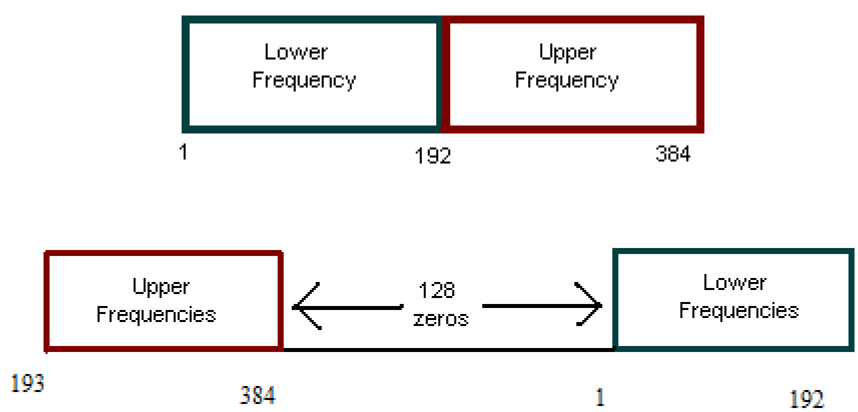

Fig. 3.7 Symbol Block before and after zero padding

b) IFFT operation → IFFT is the heart of OFDM technology. On each block obtained after zero padding it performs 512 point IFFT operation. Each block in frequency domain sample is transformed into time domain samples.

c) Guard time insertion → The responsibility for making OFDM symbols resistant to inter-symbol interference (ISI) is of this block. From each OFDM symbol it picks up copy of last 126 samples equal to guard interval and replace it at the OFDM symbol beginning through which the length of OFDM symbol is equal to OFDM symbol duration Tequivalent to 638 samples.

3.2.9 Null Symbol Generator

The addition of this block completes the final DAB frame structure for transmission as it is the last block of DAB transmitter. As Null symbol duration, Tnull equals to 664 samples, so at the beginning of the frame 664 zeros are appended. No information is transmitted during null symbol period.

3.2.10 Channel

Channel is that physical transmission medium through which the final structure of DAB signal produced by transmitter part is passed for BER performance analysis. The main parameter for mobile channels is the Doppler frequency shift. There is a frequency shift in the incoming signal because of the relative speed between the mobile receiver and the fixed transmitter. The following relation calculates the maximum Doppler frequency shift (in Hz).

$$f_D = \frac{v}{c} f_0 = \frac{1}{1080} \frac{f_0}{MHz} \frac{v}{km/hr} Hz$$

In Simulink the channels are found in communication toolbox named “AWGN Channel”, “Multipath Rayleigh Fading” and “Multipath Ricien Fading”.

3.2.11 Decoder Part

This part is same as that of Encoder part. Instead of padding zero, zeros will be removed and cyclic prefix also will be removed. QPSK demapping will be done in the opposite way as it was in encoder part. Therefore decoder part will work in the opposite manner as it was in encoder part.

3.2.12 Viterbi Decoder

Due to channel impairments the DAB system in the transmitter employed powerful rate compatible punctured convolutional code (RCPC) with constraint length 7 and mother code rate of ¼ for channel coding. According to maximum likelihood criteria which offers best performance for decoding purpose we use Viterbi Decoder which works basically on the hard decision. It is found in the communication toolbox and convolutional library.

3.2.13 Error Rate Calculation

The Bit Error Rate is calculated by using “Error Rate Calculator” and its results are shown at the “Display” block.

It computes the error rate of the received data by comparing it to a delayed version of the transmitted data. The block output is a three-element vector consisting of the error rate, followed by the number of errors detected and the total number of symbols compared. This vector can be sent to either the workspace or an output port. The delays are specified in number of samples, regardless of whether the input is a scalar or a vector. The inputs to the 'Transmitter' and 'Receiver' ports must be scalars or column vectors. The 'Stop simulation' option stops the simulation upon detecting a target number of errors or a maximum number of symbols, whichever comes first.

3.3 Objectives

In this era of compact disc the audio the listeners requires does not accomplished using the analog radio broadcasting such as AM/FM. These technologies are not capable of providing multi-program sound and data services. On the portable radio is the reception quality is badly pretentious by multipath fading (reflections, diffraction and scattering) and shadowing. Interference is also suffered in these systems from the equipment, vehicles and other radio stations.

The VHF frequency band accessible for sound broadcasting throughout world has either saturated or fast imminent saturation. To operate in so called single-frequency networks there is necessity for spectrally more efficient broadcasting technology.

In radio broadcasting, in the future the AM and FM will be replaced by the DAB as it uses the concept of COFDM technology.

The objective of the study is to improve the channel coding. The channel is random in nature so it cannot be improved but some algorithm can be applied so that error in the data can be detected and can be corrected. The Radio technologies AM and FM are providing us data with some error but due to the use of Digital Audio Broadcasting these error detection and correction capability can be improved to an extent and obviously it is better than the other two previous technologies.

The DAB uses the FEC and puncturing of vectors to make detect the data efficiently and this concept is called as COFDM.

CHAPTER 4
RESULT AND DISCUSSION

4.1 MATLAB Introduction

The MATLAB is “The Language of Technical Computing”.

The MATLAB is a high performance language for technical computing integrates computation, visualization, and programming in an easy-to-use environment where problems and solutions are expressed in familiar mathematical notation.

MATAB is a high level language and interactive environment that enables you to perform computationally intensive tasks faster than with traditional programming languages such as C,C++, and Fortran.

The key features of MATLAB are:

- High level language of technical computing
- Development environment for managing code, files, and data
- Interactive tools for interactive iterative exploration, design, and problem solving
- Mathematical functions for linear algebra, statistics, Fourier analysis, Filtering, optimization, and numerical integration.
- 2-D and 3-D graphics functions for visualizing data
- Tools for building custom graphical user interfaces
- Functions for integrating MATLAB based algorithms with external applications and languages.

4.2 Final Results

Results will be based on the BER (Bit Error Rate). A constant number of bits are transmitted and received whenever bit error rate simulation is done. It is very much necessary to know at what rate our bits are corrupted so we need to find out BER.

BER is defined as the number of bits is in error to the number of bits transmitted. So in my research work I have implemented DAB system based on OFDM technology where I have also used block coding techniques such as Linear Block Codes, Hamming Codes and Cyclic Codes. I have decreased BER to some extent using Bose-Chaudhary Hocquenghem (BCH) Codes where there was same BER with Linear Block Codes, Hamming Codes and Cyclic Codes with some amount decrease but with BCH Codes there was much decrease which can be seen in the graphs provided below.

The Time-Scatter Plot for Linear Block Codes, Hamming Codes and Cyclic Codes was almost same so only one of them is shown i.e; Linear Block Codes

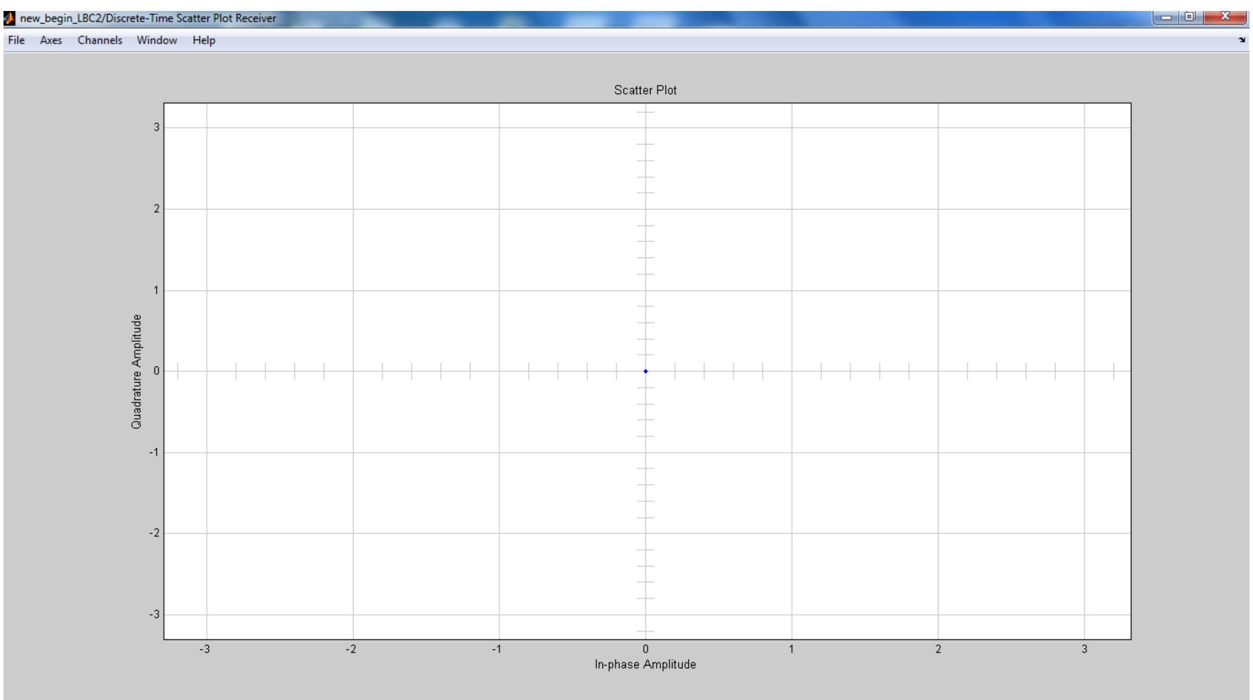

Fig. 4.1 Time-Scatter Plot of Linear Block Codes for Receiver

The above graph is the Time Scatter plot of Linear Block Codes, Hamming Codes and Cyclic Codes of receiver side and the below is for Time Scatter plot of Linear Block Codes, Hamming Codes and Cyclic Codes of transmitter side.

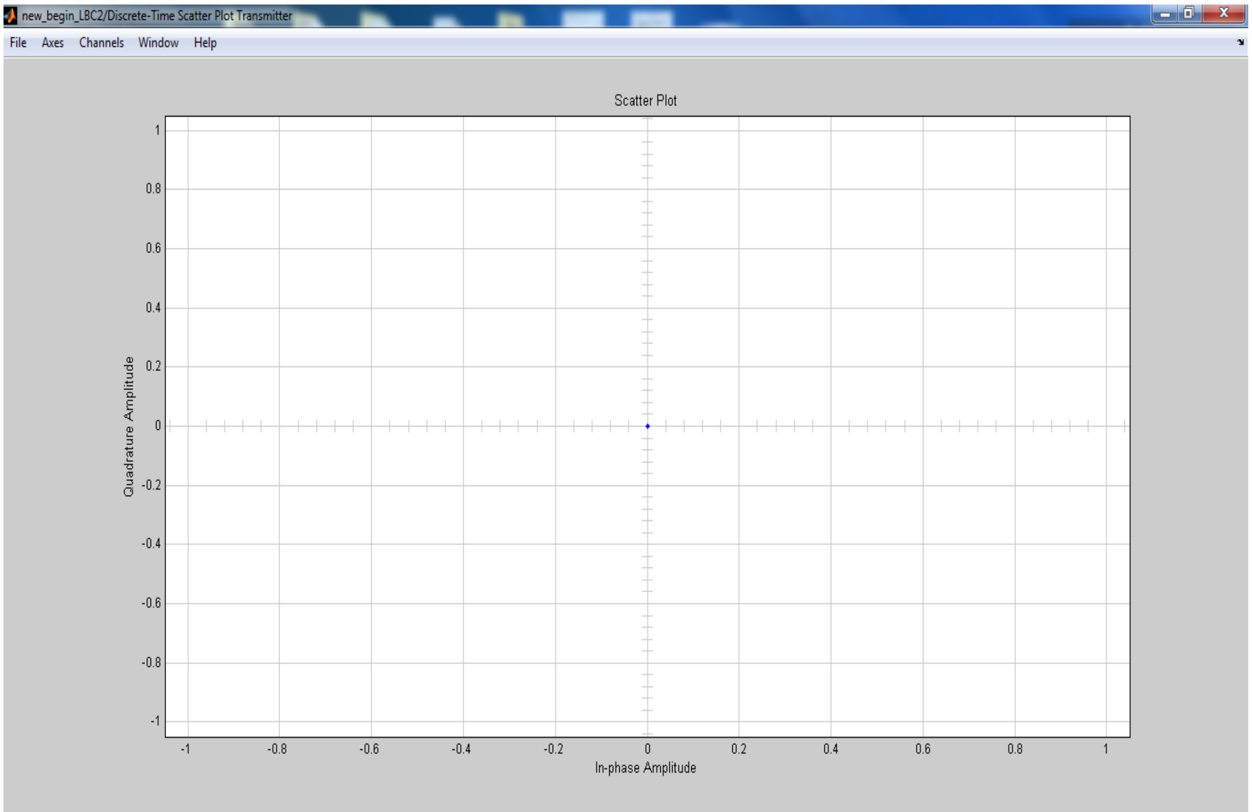

Fig. 4.2 Time-Scatter Plot for Linear Block Codes for Transmitter

The output is somewhat at the same points which look at the same points which were the result of my paper which was implemented on MATLAB and I have implemented the result on SIMULINK which shows my result in this manner. The graph showing the Bit Error Rate (BER) is given below for all the techniques.

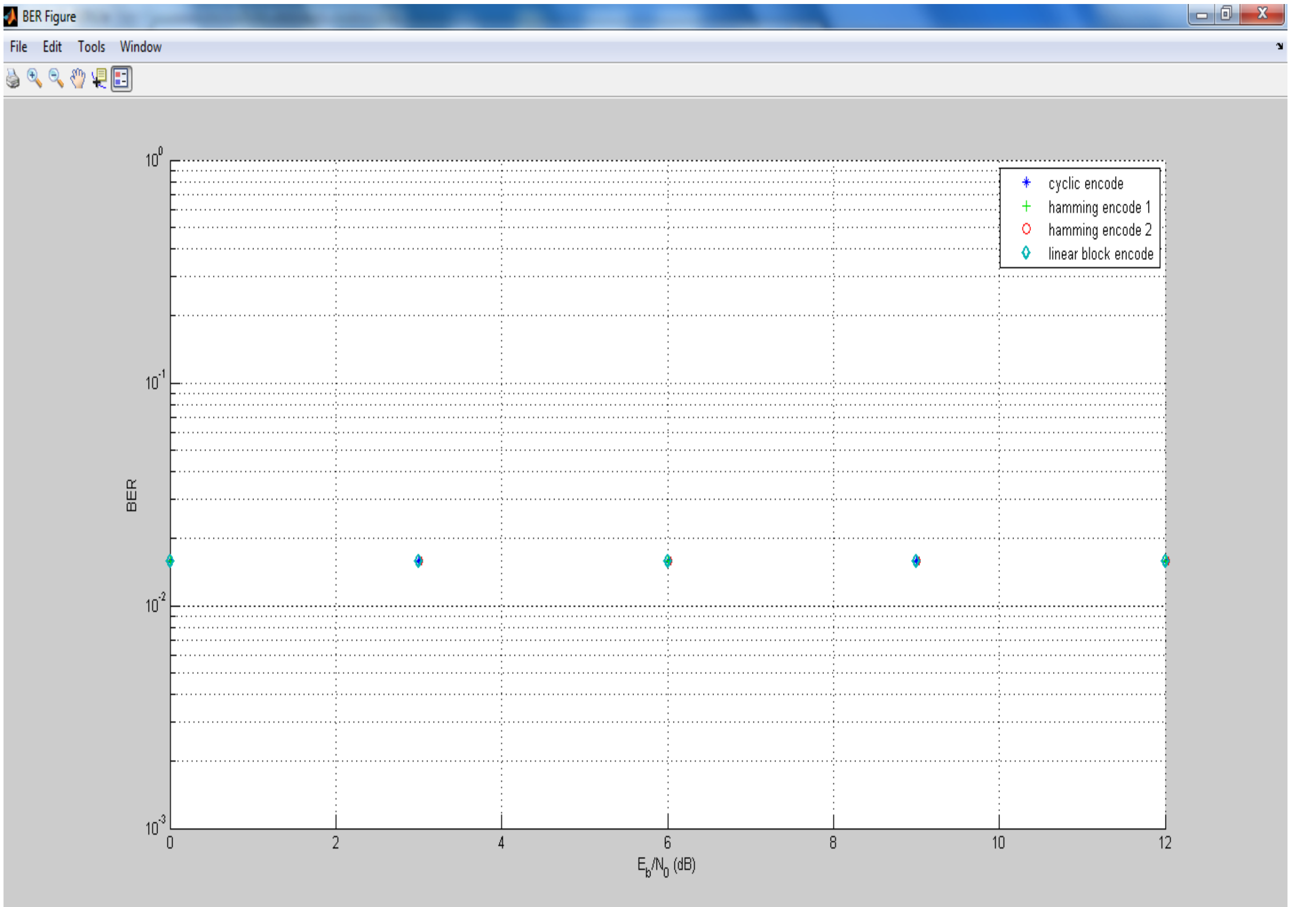

Fig. 4.3 BER plot of Linear, Hamming and Cyclic codes

Now the improvement is shown by me by the result of Bose-Chaudhary Hocquenghem (BCH) Codes on SIMULINK. My results are given below. Firstly Time Scatter plot of BCH Codes of receiver side and then the Time Scatter plot of BCH Codes of transmitter side is shown. After that individual result of BCH codes on BER is shown.

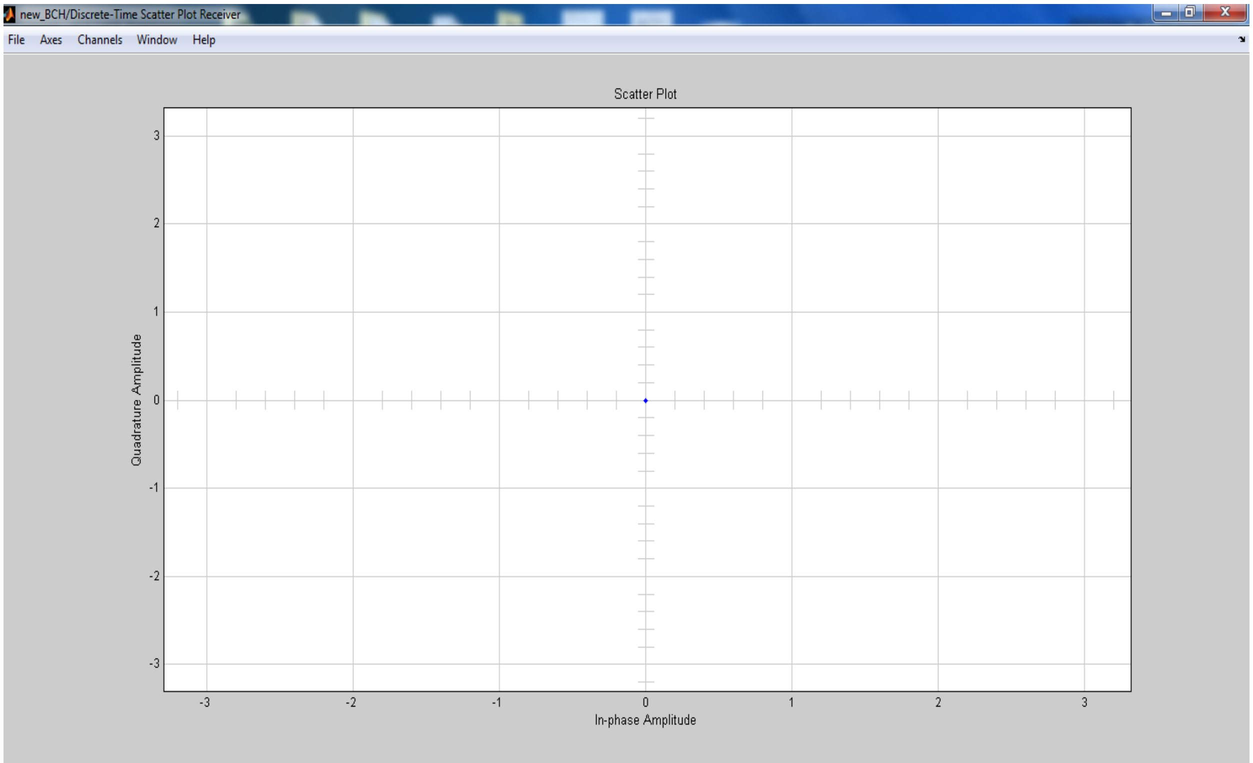

Fig. 4.4 Time-Scatter Plot of BCH Codes for Receiver

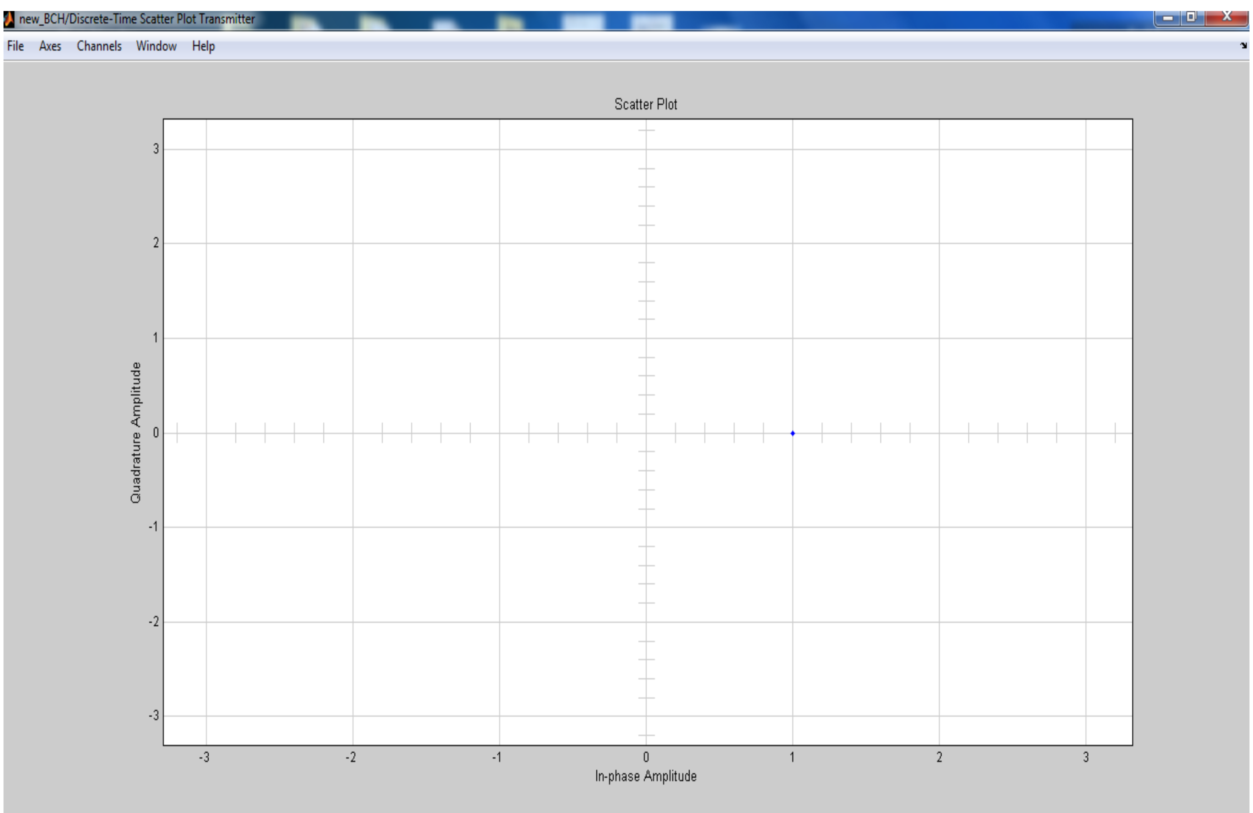

Fig. 4.5 Time-Scatter Plot of BCH Codes for Transmitter

BER result of individual is shown below:

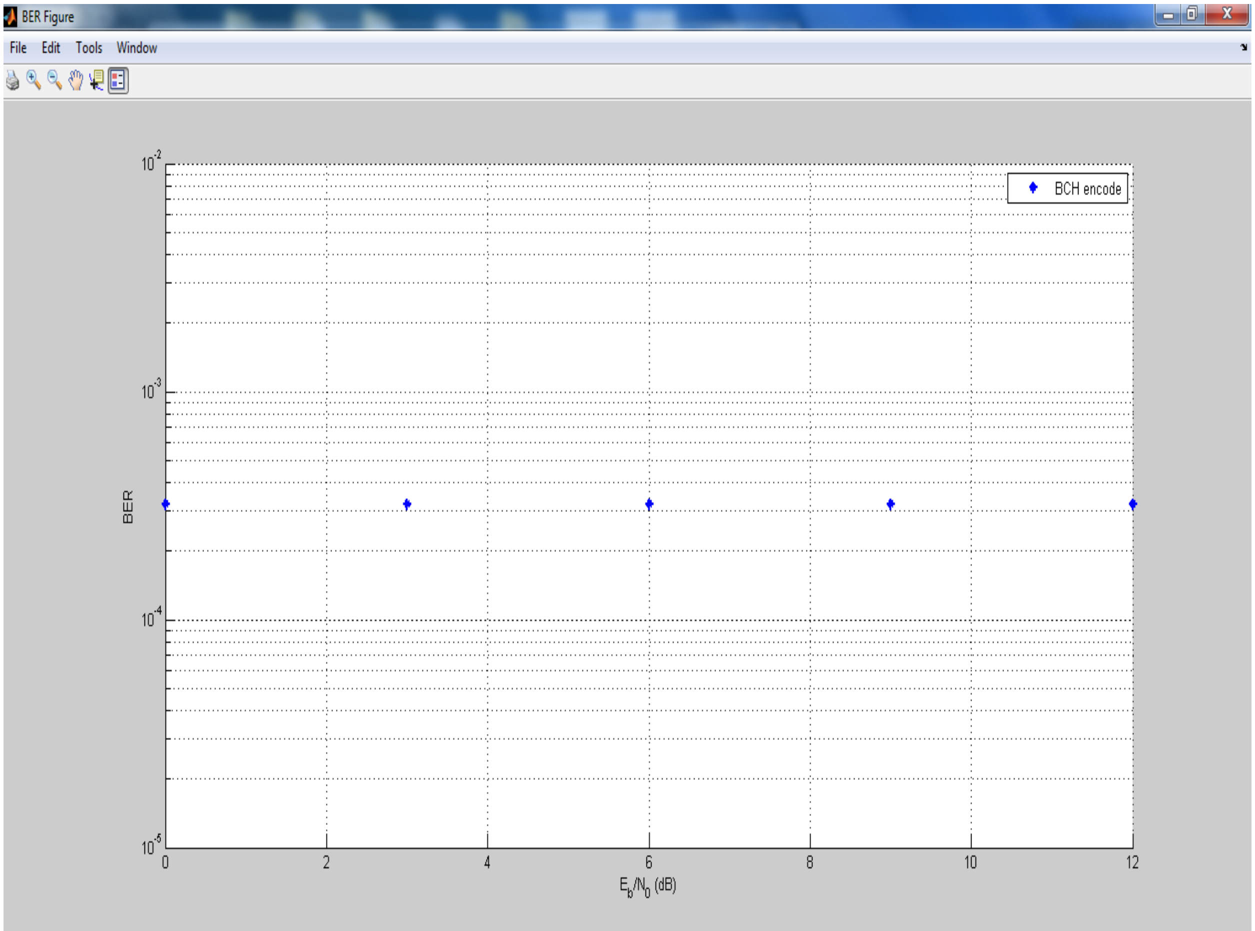

Fig. 4.6 BER result of BCH Codes

The comparison between the BER results of BCH and Linear Block Codes, Hamming Codes and Cyclic Codes are given in the next graph.

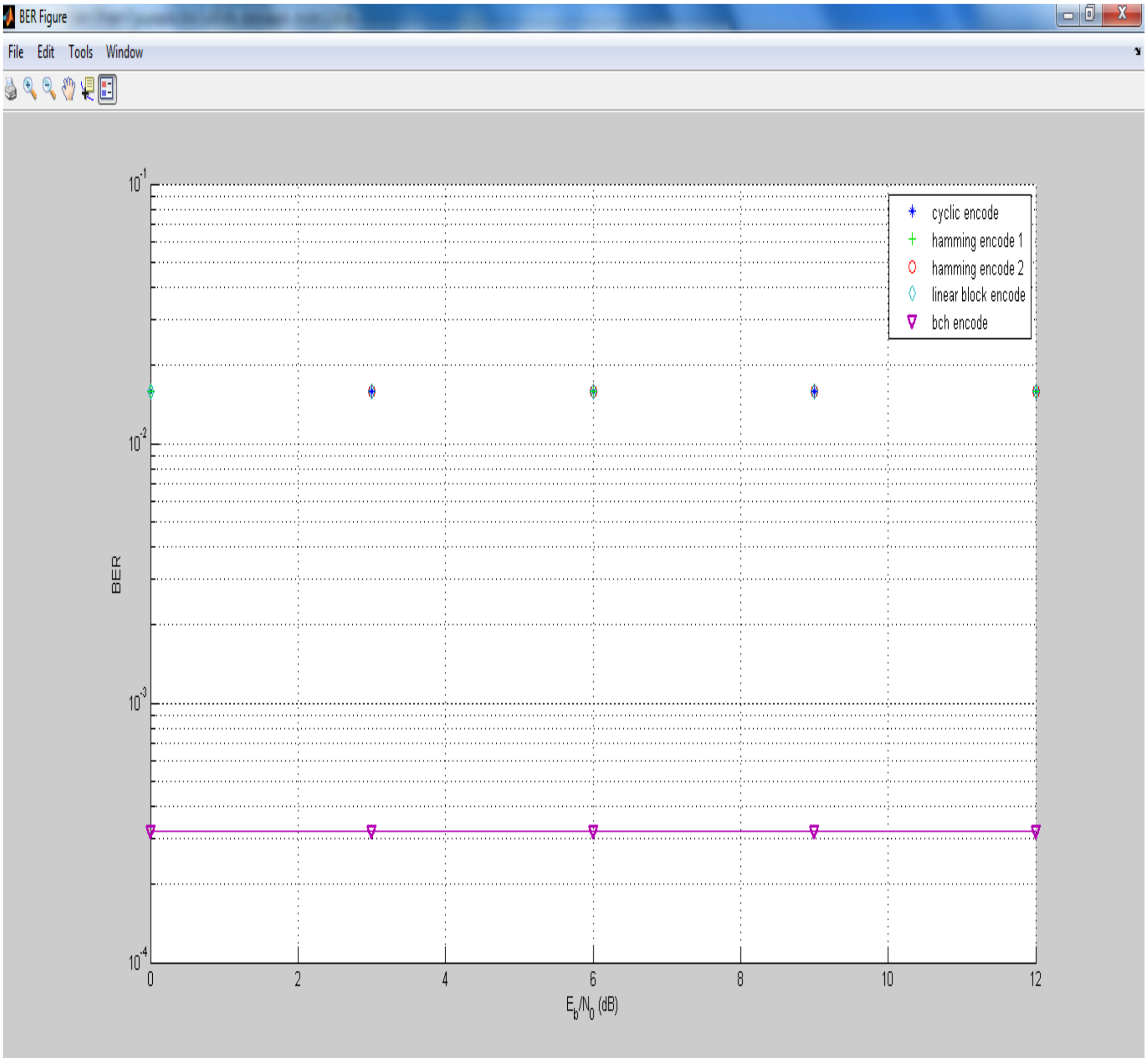

Fig. 4.7 Final BER Comparison between Block Coding Techniques

This graph shows the comparison between the BER between hard decision, soft decision of Linear Block Codes and Convolutional Encoder, Hamming Encoder and without using any convolutional and Block Coding techniques.

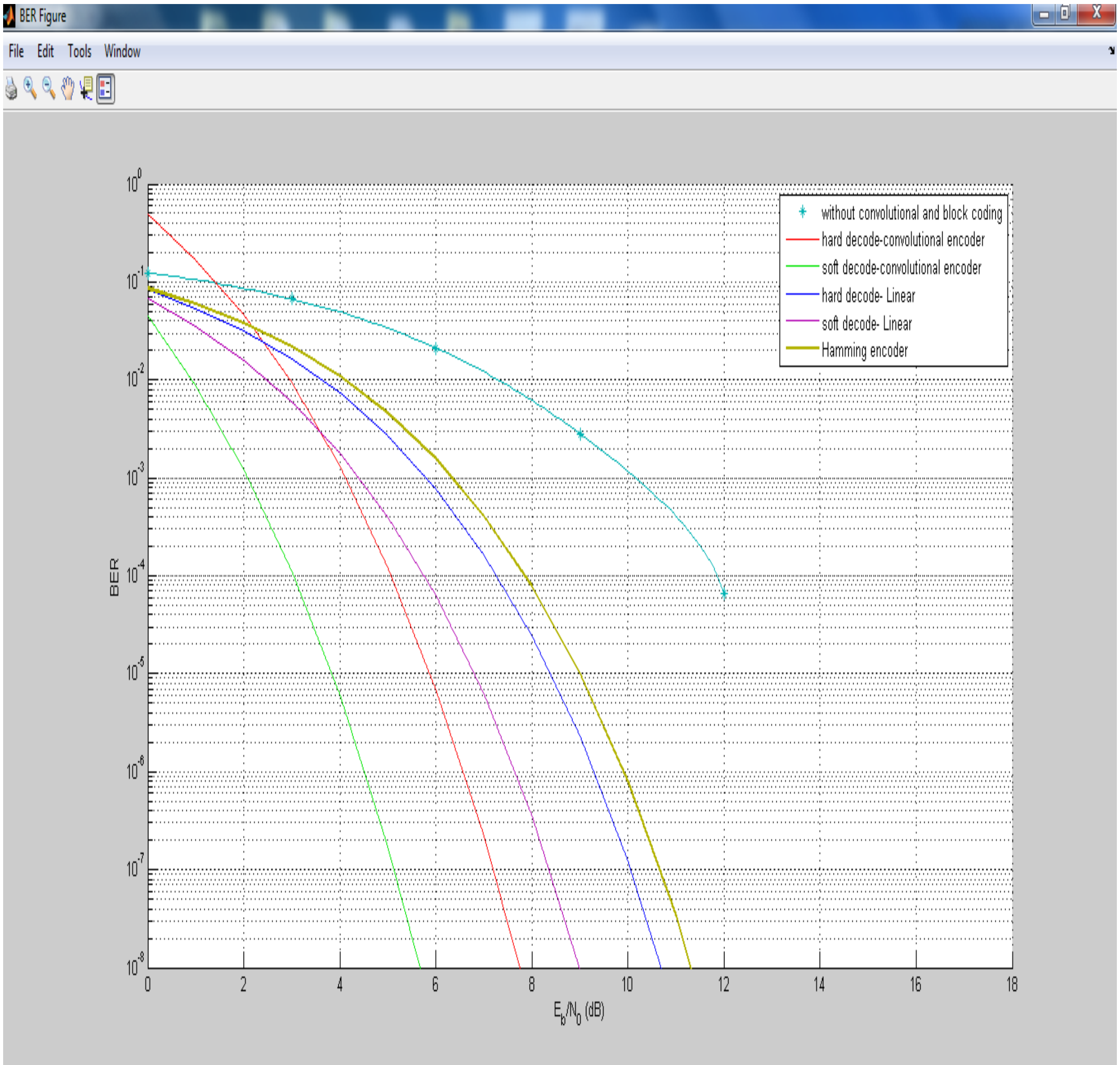

Fig. 4.8 BER results of hard and soft decision encoding and with and without convolutional encoder only

CHAPTER 5

CONCLUSION AND FUTURE SCOPE

5.1 Conclusions

Radio services was started when AM radio came into existence, which was overcome by FM radio which is used till now and for future scientist are working to implement a new digital radio system DAB which is a very innovative idea designed by the scientist using several coding techniques to work on and it is universal multimedia broadcast system which is very much expected that it will replace AM and FM radio services in future. This technique is very much efficient in power and spectrum of sound and data broadcasting. For this a project was started named EUREKA-147 DAB standard. It can be operated in any frequency band in VHF and UHF range for the terrestrial, satellite, hybrid (satellite and terrestrial), and cable broadcast networks. DAB receiver provides an unimpaired sound quality even when working in severe multipath conditions, such as in dense urban areas.

The chapter above discussed the detailed theoretical background of the EUREKA-147 DAB system. It presented the working principle and technical overview of the DAB system and told about each and everything on how to work on. It has given the detailed description related to the topic and provided the MATLAB environment on how to work on. It has also given the detailed description about the COFDM which is supposed to be the heart of the DAB system. It is the part where the main mental work to be tested which implements the Forward Error Correction codes using convolution codes using puncturing patterns.

So this was the detailed description which was given this report and I very keen to work on this topic and want to get the result as soon as possible.

5.2 Future Scope

The research work presented in this report can be extended in numerous ways. One most adequate area according to me could be the use of adaptive symbol and frame synchronization using phase reference symbols. Another could be the extension of these concepts can be done in DAB in Femtocell.

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APPENDIX

List of Abbreviations

1. AM → Amplitude Modulation
2. FM → Frequency Modulation
3. DAB → Digital Audio Broadcasting
4. ISI → Inter symbol Interference
5. SFN → Single Frequency Networks
6. MPEG → Moving Pictures Expert Group
7. COFDM → Coded Orthogonal Frequency Division Multiplexing
8. ETSI → Electronics Telecommunications Standard Institute
9. IBOC → In-Band-On-Channel
10. ISDB-T → Terrestrial Integrated Services Digital Broadcasting
11. PRBS → Pseudo Random Binary Sequence
12. FEC → Forward Error Correction
13. MSC → Main Service Channel
14. FIC → Fast Information Channel
15. IFFT → Inverse Fast Fourier Transform
16. FFT → Fast Fourier Transform
17. QPSK → Quadrature Phase Shift Keying
18. DQPSK → Differential QPSK
19. MUSICAM →Masking Pattern Universal Subband Integrated Coding And Multiplexing

20. LBC → Linear Block Codes

21. BCH → Bose-Chaudhary Hocquenghem.