DIGITAL AUDIO BROADCAST: MODULATION, TRANSMISSION & PERFORMANCE ANALYSIS

A THESIS SUBMITTED IN PARTIAL FULFILLMENT
OF THE REQUIREMENTS FOR THE DEGREE OF

Master of Technology
In
Telematics and Signal Processing

By:

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Roll No. 208-EC-117

Department of Electronics and Communication Engineering

National Institute of Technology

Rourkela, Orissa, India

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Under the guidance of

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2010

DEPARTMENT OF ELECTRONICS AND
COMMUNICATION ENGINEERING
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CERTIFICATE

This is to certify that the Thesis Report entitled "**DIGITAL AUDIO BROADCAST: MODULATION, TRANSMISSION & PERFORMANCE ANALYSIS**", submitted by **Mr. ARUN AGARWAL bearing roll no. 208-EC-117** in partial fulfillment of the requirements for the award of **Master of Technology in Electronics and Communication Engineering** with specialization in "**Telematics and Signal Processing**" during session 2008-2010 at National Institute of Technology, Rourkela is an authentic work carried out by him under my supervision and guidance.

To the best of my knowledge, the matter embodied in the thesis has not been submitted to any other university/institute for the award of any Degree or Diploma.

Place: Rourkela

Date: 25TH May, 2010

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Arun Agarwal

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ACRONYMS

AWGN	additive White Gaussian Noise
AMPS	Advanced Mobile Phone Service
ADC	Analog-to-Digital Conversion
AIC	Auxiliary Information Channel
ASIC	Application Specific Integrated Circuit
CDMA	Code-Division Multiple Access
CA	Conditional Access
CIF	Common Interleaved Frame
CRC	Cyclic Redundancy Check
CU	Capacity Unit
CW	Control Word
DAC	Digital-to-Analog Conversion
D-QPSK	Differential QPSK
DAB	Digital Audio Broadcasting
DDC	Digital Down Conversion
DUC	Digital Up Conversion
DVB	Digital Video Broadcasting
EDGE	Enhanced Data rates for GSM Evolution
EBU	European Broadcasting Union
ETS	European Telecommunication Standard
ETSI	European Telecommunications Standards Institute
EEP	Equal Error Protection
FCC	Federal Communications commission
FPGA	Field Programmable Gate Array
FIB	Fast Information Block
FIC	Fast Information Channel
FIDC	Fast Information Data Channel

FIG	Fast Information Group
GPRS	General Packet Radio Services
GPS	Global Positioning System
GSM	Global System for Mobile communication
IP	Internet Protocol
ISI	Inter Symbol Interference
ICI	Inter Carrier Interference
IDE	Integrated Development Environment
MIPS	Million Instructions Per Second
MPEG	Moving Pictures Expert Group
MSC	Main Service Channel
OFDM	Orthogonal Frequency Division Multiplex
PAD	Programme Associated Data
PCM	Pulse Coded Modulation
PRBS	Pseudo-Random Binary Sequence
SNR	Signal To Noise Ratio
QPSK	Quadrature Phase Shift Keying
TDMA	Time Division Multiple Access
UMTS	Universal Mobile Telecommunications System
UHF	Ultra High Frequency
VHF	Very High Frequency
WAP	Wireless Application Protocol
WCDMA	Wideband Code-Division Multiple Access
X-PAD	Extended Programme Associated Data

ABSTRACT

Radio broadcasting technology has evolved rapidly over the last few years due to ever increasing demands for as high quality sound services with ancillary data transmission in mobile environment. In order to accomplish this, Members of European Broadcasting Union (EBU), the European Telecommunications Standards Institute (ETSI) and International Telecommunications Union (ITU-R) developed 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.

Digital Audio Broadcasting (DAB) system developed within the Eureka 147 Project is a new digital radio technology for broadcasting radio stations that provides high-quality audio and data services to both fixed and mobile receivers. The system uses COFDM technology that combats the effect of multipath fading & ISI and makes it spectrally more efficient compared with existing AM/FM systems.

This project presents the performance analysis of Eureka-147 DAB system. DAB transmission mode-II is implemented first and then extended successfully to other modes. A frame-based processing is used in this study. Performance studies for AWGN, Rayleigh and Rician channels have been conducted. For all studies BER has been used as performance criteria.

This project also discusses issues related to system performance using concatenated coding technique, including the outer Block code, the inner convolutional code, outer BCH code and the inner convolutional code.

Chapter 1
INTRODUCTION

1.1 Introduction

Radio broadcasting is one of the most widespread electronic mass media comprising of hundreds of programme providers, thousands of HF transmitters and billions of radio receivers worldwide. Since the broadcasting began in the early 1920s, the market was widely covered by the AM services. Today with the invent of FM we live in a world of digital communication systems and services because of its advantages over analog systems like storage capacity, reliability, quality of service, miniaturization and many more.

The new digital radio system Digital Audio Broadcasting (DAB) has the capability to replace the existing AM and FM audio broadcast services in many parts of the World in near future. This was developed in the 1990s by the Eureka 147 DAB project. DAB is very well suited for mobile receivers and provides very high tolerance against multipath reception and inter symbol interference (ISI). It allows use of single frequency networks (SFNs) for high frequency efficiency. In several countries in Europe and overseas, broadcasting organizations, network providers and receiver manufacturers are already implementing digital broadcasting services using the DAB system. Perceptual audio coding (MPEG-2), Coded Orthogonal Frequency Division Multiplexing (COFDM), provision for the multiplex of several programmes and data transmission protocols, are the new concepts of digital radio broadcasting [1] [2].

1.2 Thesis Objectives

The work reported in this thesis evaluates the performance of DAB. Performance studies have been carried out for DAB employing Coded Orthogonal Frequency Division Multiplexing (COFDM). Performance of COFDM DAB in multipath fading channels with Rayleigh and Rician fading have been analyzed. Bit error rate (BER) has been considered as the performance index in all analysis. The analysis has been carried out with simulation studies under MATLAB environment.

1.3 Motivation

The analog radio broadcasting such as AM/FM does not provide the listeners the audio quality they desire in this era of compact disc. Moreover these technologies are not capable of providing multi-programme sound and data services. The reception quality of these analog systems on portable radio is badly affected by Multipath fading (reflections from aircraft, vehicles, buildings, etc.) and shadowing [3]. These systems also suffer from interference from equipment, vehicles and other radio stations.

Additionally the VHF frequency band available for sound broadcasting throughout the world has either saturated or fast approaching saturation. There is a need for spectrally more efficient broadcasting technology that can operate in so called single-frequency networks (SFN) [3].

DAB is expected to be the future for radio broadcasting replacing AM/FM as it uses concept of COFDM technology.

1.4 Thesis outline

Following the introduction, the rest of the thesis is organized as follows:

Chapter 2: This chapter describes the theoretical background and history of DAB system along with its brief technical overview. Details of DAB modes and system parameters are provided. OFDM basic theoretical concepts are also covered.

Chapter 3: This chapter describes the complete details of MATLAB model used to simulate o the physical part of the DAB system in transmission mode-II.

Chapter 4: This chapter reports the results of the DAB system simulated along with some analysis.

Chapter 5: This chapter will conclude on the results from all the simulations. Discussions and analysis are included in this section. There is, also, a discussion on the suggestion for future work.

Chapter 2
LITERATURE REVIEW

2.1 Introduction

The new digital radio system DAB is a very innovative and universal multimedia broadcast system which is expected to replace analog amplitude and frequency modulation radio. It is a rugged, yet highly power and spectrum efficient sound and data broadcasting technique. The Eureka 147 DAB standard is designed to operate in any frequency band in the VHF and UHF range for the terrestrial, satellite, hybrid (satellite and terrestrial) and cable broadcast networks [1] [4]. Even when working in severe multipath conditions, such as in dense urban areas, DAB receiver provides an unimpaired sound quality [3].

Besides high-quality digital audio services (mono, two-channel or multichannel stereophonic), DAB can provide ancillary data transmission (e.g. travel and traffic information, still and moving pictures, etc.) [2]. It has large coverage area than current AM and FM systems and requires low transmitting power. During its development DAB system has been publicly demonstrated many times. It has been subject to extensive computer simulations and field tests in Europe and elsewhere. It is now in regular service in many European countries and throughout the world. In 1995, the European DAB Forum (Euro Dab) was established to pursue the introduction of DAB services in a concerted manner world-wide and it became the World DAB Forum in 1997 [1]. As a result of developments within the Eureka 147 project, the DAB Standard or DAB Specification in the form of EN 300401 was approved by the European Telecommunications Standards Institute (ETSI), which defines the characteristics of the DAB transmission signal, including audio coding, data services, signal and service multiplexing, channel coding and modulation [2].

DAB multiplexes several audio programmes (of 24 KHz/48 KHz PCM) into a so called-ensemble with a bandwidth of 1.536 MHz, where the number of programmes per ensemble is flexible and depends on individual programme bandwidth requirements [5].

2.2 DAB - A Brief History

The first digital sound broadcasting systems providing CD-like audio quality was developed in early 1980s using Satellite technology. The system employed very low data compression and was not suitable for mobile reception. It used frequency in the range 10-12 GHz. Therefore it was not possible to provide service to large number of listeners. It was realized terrestrial digital sound broadcasting would do the job and to develop this new digital solution an international research project was necessary. So, in 1986 few organizations from France, Germany, United Kingdom and The Netherlands signed an agreement to cooperate in the development of a new standard and with this Eureka-147 project was born [2] [6].

Members of European Broadcasting Union (EBU), who were the part of work on the satellite delivery of digital sound broadcasting to mobiles in the frequency range between 1 and 3 GHz, also joined the Eureka-147 project. Later International Telecommunications Union (ITU-R) and the European Telecommunications Standards Institute (ETSI) started the standardization process.

Following goals were set up for DAB from the beginning with the sole aim of quality audio for mobile reception:

- High quality digital audio services (near CD quality).
- Well suited for mobile reception in vehicles, even at higher speeds.
- Efficient frequency spectrum usage
- Transmission capacity for ancillary data.
- Low transmitting power.
- Terrestrial, cable and satellite delivery options.
- Easy tuning of receivers.
- Large coverage area than current AM and FM systems.

With the goals set, following transmission method were proposed to achieve above mentioned goals:

- One narrow-band system.
- Single carrier spread-spectrum system.
- One multicarrier OFDM (Orthogonal Frequency Division Multiplex) system.
- One frequency-hopping system.

Eureka 147 consortium alone started choosing the most appropriate transmission method based on thorough simulation and field test. Results showed that broadband solutions performed better than the narrow-band proposal, while the frequency-hopping solution was considered too demanding with respect to network organization. Since the spread-spectrum was not developed as hardware therefore coded Orthogonal Frequency Division Multiplex (COFDM) system was chosen finally.

The next issue was audio coding standard selection for the Eureka-147 project. By that time the MPEG (Moving Pictures Expert Group) had already been standardized for data compression both video and audio coding. The solutions proposed by the Eureka-147 were sent to the MPEG Audio group to be evaluated with other several options from other countries. The performance offered by the methods submitted by the Eureka consortium was clearly superior so they were standardized by the MPEG as MPEG Audio Layers I, II and III. It took a long time until the final decision to which standard should be used for DAB was taken [4]. Finally, Layer II, also known as MUSICAM was chosen.

Another important specification for the DAB was the bandwidth consideration. From a network and service area planning point of view, one transmitter with the 7 MHz bandwidth of a TV channel was too much inflexible, but showed very good performance in a multipath environment [2]. Therefore considerable reduction in transmission bandwidth was necessary. In Canada experiments with the COFDM system also revealed that performance degradation begins around 1.3MHz and lower. Therefore appropriate bandwidth for a DAB channel was fixed at 1.5 MHz

With this one 7 MHz TV channel can be divided into four DAB blocks, each carrying ensembles of five to seven programmes.

The first DAB standard was achieved in 1993 and then in 1995 the ETSI adopted DAB as the only European standard for digital radio. The Eureka 147 DAB standard as digital radio is accepted Worldwide except, USA and Japan. In USA the National Association of broadcasting rejected the Eureka 147 DAB system because of strong opposition against the system requiring new spectrum and ensembling of several programmes into one transmitter. USA uses the idea of system approach named ‘In-Band-On-Channel (IBOC)’. Japan has its own developed national solution called Terrestrial Integrated Services Digital Broadcasting (ISDB-T).

2.3 What is DAB?

Digital Audio Broadcasting (DAB) is a new digital radio system that delivers radio services from the studio to the receiver. DAB is intended to deliver very high quality digital audio programmes and data services to fixed, mobile and portable receivers which can use simple whip antennas. It was developed in the 1990s by the Eureka 147 DAB project. DAB is very well suited for mobile reception and provides very high robustness against multipath reception. It allows use of single frequency networks (SFNs) for high frequency efficiency.

DAB uses COFDM technology that makes it resistant to Multipath fading and inter symbol interference (ISI). FM reception can be badly affected by shadowing [3] (i.e. the blocking or screening of the signals by tall buildings and hills which lie in the direction of the transmitter) and by passive echoes (the arrival at the receiver of delayed “multipath” signals which have been reflected from tall buildings and hills). DAB is tolerant to all these types of interferences with the use of simple whip antenna.

The RF signal amplitude at the receiver input varies over time, which is called signal fading. This can be slow fading and fast fading. In wireless telecommunications, Multipath is the propagation phenomenon that results in radio signals reaching the receiving antenna by two or more paths. Causes of multipath include ionospheric reflection and refraction. Reflection from water bodies and terrestrial objects such as mountains, buildings, etc. The effects of multipath include constructive and destructive interference and phase shifting of the signal. This causes Rayleigh fading. This multipath effect is illustrated in Figure 2.1

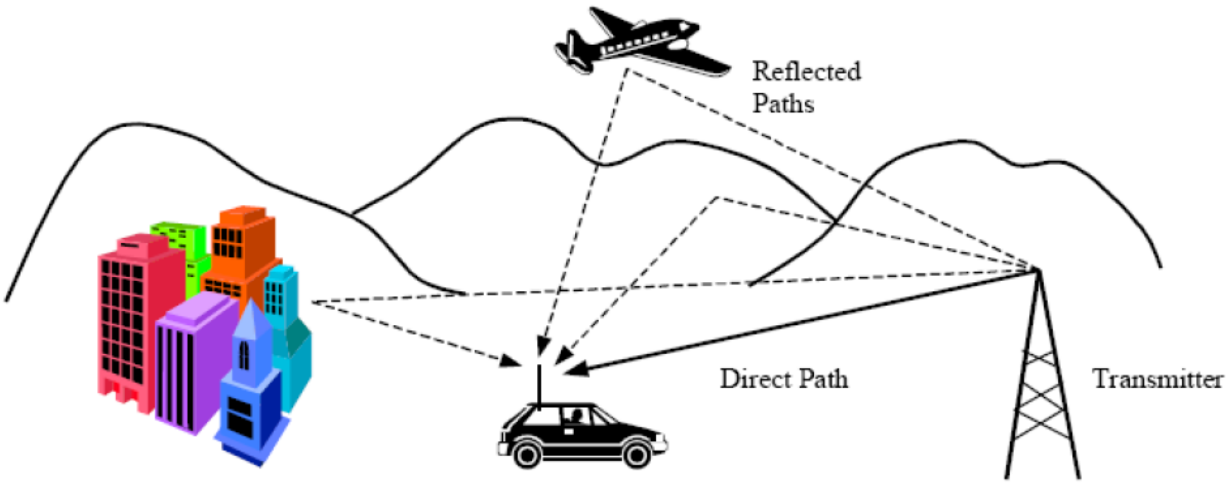

Figure 2. 1: Effect of multipath on mobile receiver.

The multipath effect like Doppler spread, diffraction, reflection etc., as shown in Figure 2.1 is absent in DAB since it employs advanced digital techniques such as OFDM multicarrier modulation, rate-compatible punctured convolutional codes (RCPC) and time-frequency interleaving.

2.3.1 Advantages of DAB

The Eureka 147 DAB system is the most significant advancement in radio broadcasting technology since the introduction of FM and AM. It offers both listeners and broadcasters a unique combination of benefits and opportunities [2] [7]. These include the following:

- High quality audio services (near CD quality).
- Both music and data services (such as text information, news, graphics, still or moving pictures) can be received using the same receiver.
- The DAB system has universal and well standardized system layout and a wide range of receiving equipment including fixed (stationary), mobile and portable radio receivers.
- Efficient use of the available radio frequency spectrum.

- Provides flexible bit rates between 8 and 384 Kbit/s.
- DAB services can be transmitted in a flexible multiplex configuration.
- DAB transmitter networks can be designed as Single Frequency Networks (SFNs).
- Low transmitting power.
- High robustness against Multipath reception.
- Larger coverage area than current AM and FM systems.

2.4 Principle of DAB system

The working principle of DAB system in illustrated in the conceptual block diagram shown in Figure 2.2. It gives details of important DAB system blocks. The overall DAB transmission system can be divided into a number of functional blocks that process the input signal to produce complete DAB transmitted signal.

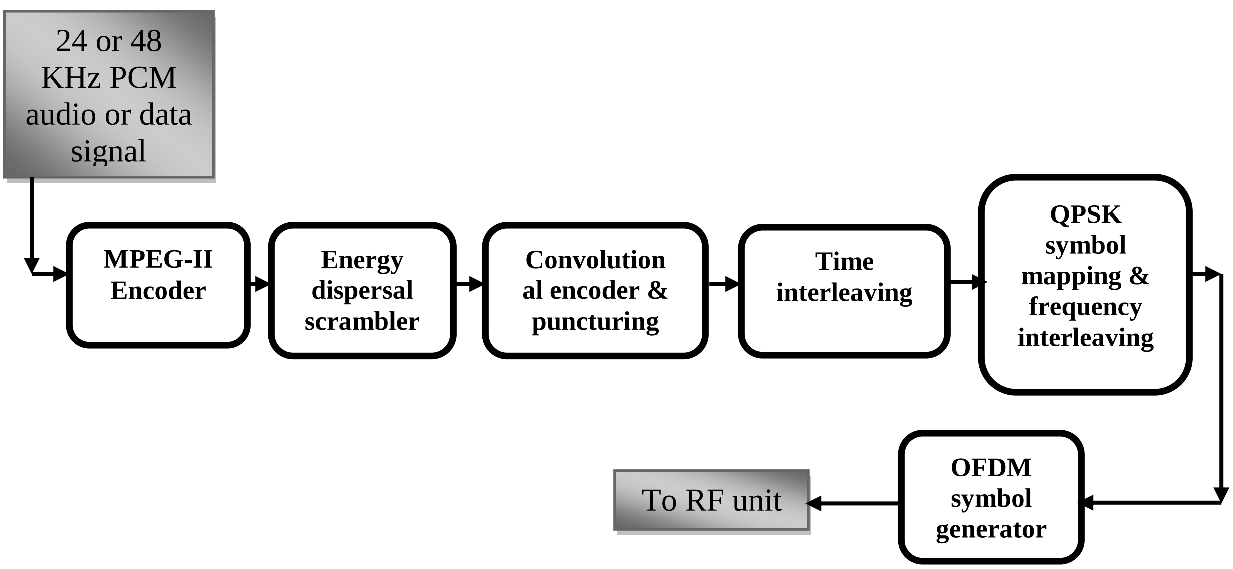

Figure 2. 2: Complete DAB transmitter block diagram [1].

At the input of the system the analog signals such as audio and data services are MPEG layer-2 encoded and then scrambled. In order to ensure appropriate energy dispersal in the transmitted signal, the individual inputs of the energy dispersal scramblers shown in Figure 2.2 shall be scrambled by a modulo-2 addition with a pseudo-random binary sequence (PRBS), prior to convolutional encoding [1]. The PRBS shall be defined as the output of the feedback shift register and shall use a polynomial of degree 9, defined by:

$$P(X) = X^9+X^5+1 \tag{2.1}$$

The scrambled bit stream is then subjected to Forward Error Correction (FEC) employing punctured convolutional codes with code-rates in the range 0.25-0.88. The coded bit-stream is then time interleaved and multiplexed with other programmes to form Main Service Channel (MSC) in the Main Service Multiplexer. The output of the multiplexer is then combined with service information in the Fast Information Channel (FIC) to form complete DAB frame.

Then after differential QPSK modulation with frequency interleaving of subcarriers in the frame is passed to OFDM signal generator basically inverse Fourier transform (IFFT) to form final DAB transmission signal.

2.5 Technical Overview

The Eureka 147 DAB system is very reliable, multi-programme, digital radio broadcasting system, intended mainly for robust reception by mobile, portable and fixed receivers, using simple antennas. ETSI, EN 300 401 [1] specifies the complete specification for the DAB transmitted signal.
The Eureka 147 DAB system consists of three main elements. These are:

- Source coding.
- Channel coding, Multiplexing and Transmission frame.
- COFDM Modulation.

The following sub-sections describe in detail the Technical aspects of DAB system that makes it highly robust against multipath affects providing CD quality audio services.

2.5.1 Source Coding

Source coding employs MUSICAM (Masking Pattern Universal Sub-band Integrated Coding And Multiplexing) audio coding that uses the principle of Psycho acoustical masking as specified for MPEG-2 Audio Layer-II encoding. This exploits the knowledge of the properties of human sound perception, particularly, the spectral and temporal masking effects of the ear. Principle of MUSICAM audio coding system is that it codes only audio signal components that the ear will hear, and discards any audio component that, according to the Psycho acoustical model, the ear will not perceive [7].

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

For a sampling frequency of 48 KHz, 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 24 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 [1].

2.5.2 Channel Coding, Multiplexing and Transmission Frame

The DAB system allows great flexibility in the choice of the proper error protection for different applications and for different physical transmission channels. Using rate compatible punctured convolutional (RCPC) codes, 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.

Channel coding is based on punctured convolutional forward-error-correction (FEC) which allows both equal and Unequal Error Protection (UEP), matched to bit error sensitivity characteristics [1]. The UEP is primarily designed for audio but can be used for data. The EEP can be used for audio as well as for data. Basic idea of RCPC channel coding is to generate first the mother code. The daughter codes will be generated by omitting certain redundancy bits. The channel coding is based on a convolutional code with constraint length 7. The octal forms of the generator polynomials are 133, 171, 145 and 133, respectively. The encoder can be thought as shift register shown in Figure 2.3.

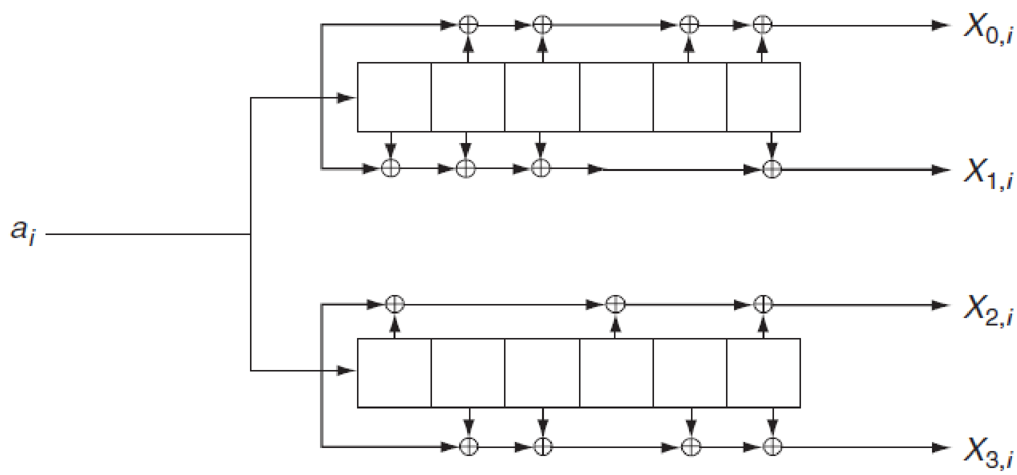

Figure 2. 3: Channel encoder for the DAB mother code [2].

The mother code has the code rate R =1/4, that is for each data bit a_i the encoder produces four coded bits $x_{0,i}$, $x_{1,i}$, $x_{2,i}$, and $x_{3,i}$. The mother code is defined by [1]:

$$x_{0,i} = a_i \oplus a_{i-2} \oplus a_{i-3} \oplus a_{i-5} \oplus a_{i-6} \quad (2.2)$$

$$x_{1,i} = a_i \oplus a_{i-1} \oplus a_{i-2} \oplus a_{i-3} \oplus a_{i-6} \quad (2.3)$$

$$x_{2,i} = a_i \oplus a_{i-1} \oplus a_{i-4} \oplus a_{i-6} \quad (2.4)$$

$$x_{3,i} = a_i \oplus a_{i-2} \oplus a_{i-3} \oplus a_{i-5} \oplus a_{i-6} \quad (2.5)$$

for i = 0, 1, 2, ..., I+5. The parameter I depends on audio bit rate.

A code rate of 1/2 or 1/3 can be obtained by applying appropriate puncturing vectors given in the DAB standard [1]. RCPC codes provide possibility of Unequal Error Protection (UEP) of a data stream. It is possible to save capacity and add as much redundancy necessary by using RCPC codes. UEP is particularly useful for MPEG-1 and MPEG-2 Audio Layer-II data. They are organized in frames of length 24 ms. For audio data with a sampling frequency of 48 KHz, the DAB system allows 14 different data rates between 32 and 384 Kbps. The protection profiles for all these date rates are grouped into five Protection Levels PL1 to PL5. PL1 is the most robust protection level, PL5 the least robust one. All protection levels except PL5 are designed for mobile reception.

In section 2.5, it was discussed that individual programme are initially encoded, error protected by applying FEC and then time interleaved. These outputs are then combined together to form a single data stream ready for transmission. This process is called as Multiplexing. In DAB several programmes are multiplexed into a so-called ensemble with a bandwidth of 1.536 MHz.

The DAB signal frame has the following structure that helps in efficient receiver synchronization. It is illustrated in Figure 2.4

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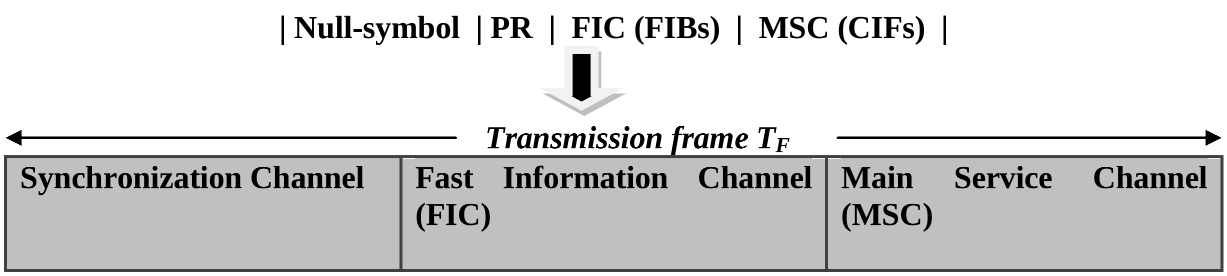

Figure 2. 4: DAB transmission signal frame structure.

According to Figure 2.4 the first symbol is the Synchronization channel consisting of Null symbol and the phase reference symbol. The next symbol must be FIC channel and last symbol is the MSC. MSC forms the useful payload of the DAB frame.

Where: PR = Phase Reference symbol, FIC = Fast Information Channel, FIB = Fast Information Block, MSC = Main Service Channel, CIF = Common Interleaved Frame, CU = Capacity Unit
1 CU = 64 bits, 1 CIF = 864 CU = 55.296 Kbits, 1 FIB = 256 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 24 ms or an integer multiple of it. The structure of transmission mode-II is very simple. The transmission frame length is 24 ms. The first two OFDM symbols constitute the synchronization channel. The next three OFDM symbols carry the data of the Fast

Information Channel (FIC) that carry the information regarding multiplex configuration and transmitted programmes [2]. The next 72 OFDM symbols carry the data of Main Service Channel (MSC). Figure 2.5 shows the transmission frame structure that is also valid for TMs -I and IV.

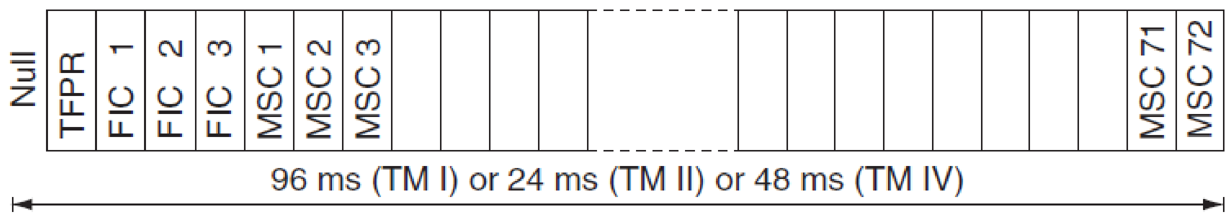

Figure 2. 5: Transmission Frame structure [4].

The DAB transmission frame has three channels as described below [1]:

1) **Main Service Channel (MSC):** This is used to carry audio and data service components. The MSC is a time interleaved data channel divided into a number of sub-channels which are individually convolutionally coded, with equal or unequal error protection. Each sub-channel may carry one or more service components. The MSC is made up of CIFs. The organization of the sub-channels and service components is called the multiplex configuration. The MSC of the DAB system has a gross capacity of 2.304 Mbps.

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

3) **Synchronization channel:** It consists of two symbols i.e., Null Symbol, during which no information is transmitted and Time frequency phase reference symbol (TFPR) which has predetermined modulation. Synchronization channel is used internally within the transmission system for basic demodulator functions, such as transmission frame

synchronization, automatic frequency control, channel state estimation, and transmitter identification.

2.5.3 COFDM Modulation

The main advantage of the DAB system developed in the European Eureka-147 standard is its ability to deliver high quality audio (near CD quality) services to mobile receivers under different channel conditions. This is because of the use of rugged transmission technology called the Coded Orthogonal Frequency Division Multiplexing (COFDM). This is the heart of Digital Audio Broadcasting. COFDM modulation combines the multi-carrier modulation technique OFDM (Orthogonal Frequency Division Multiplexing) with convolutional channel coding in such a way that the system can exploit both time and frequency diversity. This is achieved by interleaving data symbols, in the time and frequency domains, prior to transmission [8].

2.6 OFDM Theory

A mobile channel is characterized by a multipath fading environment. Due to which the received signal contains not only the direct line-of-sight radio wave but also a large number of reflected waves that arrives at the receiver at different times. Delayed signals are a result of reflections from trees, hills, mountains, vehicles and building. These reflected delayed waves interfere with the direct wave causing inter symbol interference (ISI) and thereby loss of information and degradation of network performance [9] [10].

Frequency Division Multiplexing (FDM) was used for a long time to carry more than one signal over a telephone line. FDM divides the channel bandwidth into sub channels and transmits multiple relatively low rate signals by carrying each signal on a separate carrier frequency. To ensure that the signal of one sub channel did not overlap with the signal from an adjacent one, some guard-band was necessary which is an obvious loss of spectrum and hence bandwidth.

In order to overcome the problem of multipath fading environment and bandwidth efficiency OFDM technology was proposed. OFDM stands for *Orthogonal Frequency Division*

Multiplexing. OFDM is a combination of modulation and multiplexing. OFDM is based on a parallel data transmission scheme that reduces the effect of multipath fading and makes the use of complex equalizers unnecessary. OFDM is derived from 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.

OFDM converts frequency selective fading channel into N flat fading channels, where N is the number of sub-carriers. Othogonality is maintained by keeping the carrier spacing multiple of 1/Ts by using Fourier transform methods, where Ts is the symbol duration.

The OFDM symbol in baseband is given by following equation 2.2, where N= no. of sub-carriers

$$\mathrm{sbb}(t) = \sum_{k=0}^{N-1} \frac{1}{\sqrt{N}} . \left(\mathrm{dk}\ e^{\frac{j2\pi kt}{T}}\right) \text{ if } 0 \leq t \leq T \tag{2.6}$$

The basic OFDM transmitter and receiver are illustrated in Figure 2.6.

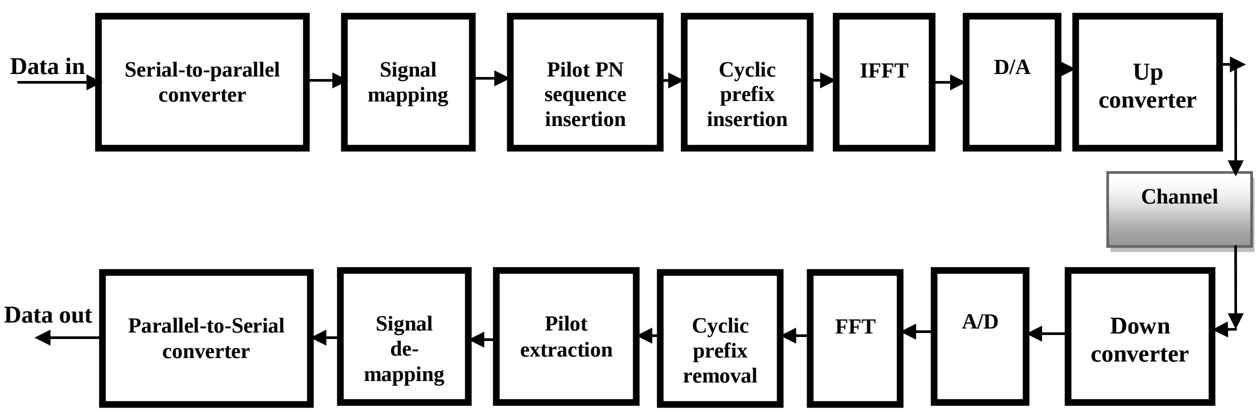

Figure 2. 6: Basic block diagram of OFDM system.

2.6.1 Interpretation of IFFT & FFT

The Fast Fourier Transform is a very efficient mathematical method for calculating DFT. It can be easily implemented in integrated circuits at fairly low cost. With the advances in VLSI and

DSP technology the implementation cost of OFDM is drastically reduced since heart of OFDM is merely IFFT/FFT operation. But the complexity of performing an FFT is dependent on the size of the FFT. The direct evaluation of an N-point DFT using the following formula:-

$$X(k) = \sum_{n=0}^{N-1} \mathrm{x(n)}.\, \mathrm{e}^{-\frac{\mathrm{j}2\pi \mathrm{nk}}{\mathrm{N}}} \tag{2.7}$$

where k=0,1,2…….,N-1.

require 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. Moreover FFT algorithm works efficiently when N is a power of 2, therefore the number of sub-carriers is usually kept as power of 2.

IFFT/FFT operation ensures that sub-carriers do not interfere each other. IFFT is used at the transmitter to obtain the time domain samples of the multicarrier signal. FFT is used to retrieve the data sent on individual sub-carriers. Therefore OFDM has a very simple implementation capability.

2.6.2 Guard time and Cyclic prefix

In order to overcome the problem of multipath fading environment and hence inter symbol interference ISI, it is common practice in OFDM technology to add guard interval between OFDM symbols. The guard interval is formed by a cyclic continuation of the signal so the information in the guard interval is actually present in the OFDM symbol. Guard interval makes the system robust against multipath delay spread. The guard interval is actually added by taking the copy of the last portion of the OFDM symbol and placing it at the start of the symbol as illustrated in Figure 2.7.

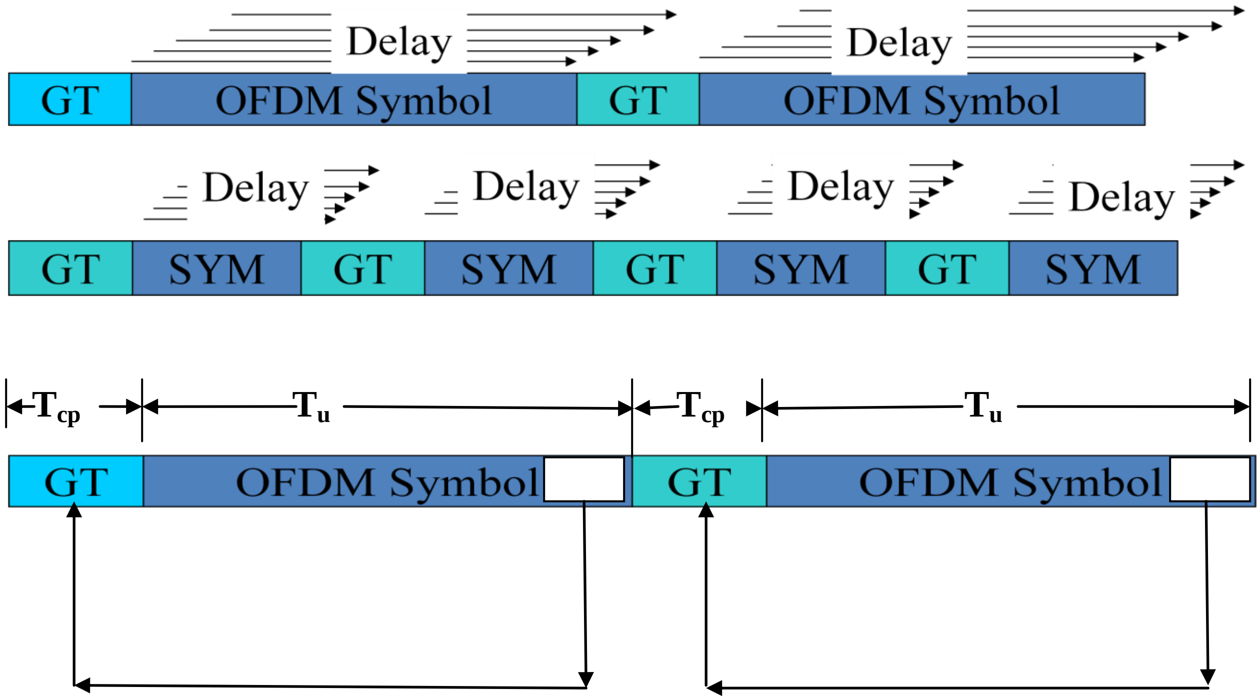

Figure 2. 7: Guard time and Cyclic prefix.

The last T_g portion of the symbol is appended and transmitted during the Guard time. T_u is the OFDM symbol time without guard interval. T_{cp} is the duration of the copied information in the guard interval using cyclic prefix.

Therefore total symbol time $Ts = T_u + T_{cp}$ (2.8)

Guard time needs to be greater than maximum delay spread otherwise ISI results. The guard time also eliminates the need of a pulse shaping filter, and it reduces the sensitivity to time synchronization problems. Cyclic prefix helps in maintaining orthogonality between sub-carriers by converting linear convolution into circular convolution in multipath environment and also avoids ICI (Inter channel interference).

2.6.3 Error Control Coding

Error control coding or channel coding transforms the signal to improve communications performance by increasing the robustness against channel impairments like noise, interference, fading, etc. This is accomplished by adding redundant bits to the original data. The probability of error for a particular signaling scheme is a function of signal-to-noise ratio at the receiver input and the information rate.

There are three broad classification of error control coding [10],

1. Automatic Repeat reQuest (ARQ):

   These are full duplex, only error detection codes. In ARQ method when an error is detected in a received signal, the receiver sends a request to the transmitter for re-transmission. Thus obvious disadvantage of these codes is the time delays those results due to request and repeat signals making ARQ inappropriate for real time systems. Example: CRC (Cyclic Redundancy Check) codes.

2. Forward Error Correction (FEC):

   These are simplex codes, having inbuilt capability of error detection and as well as error correction. Here re-transmission of data is not necessary. FEC is widely used for real time systems such as DAB, DVB-T, WiMAX, etc. Examples: Block codes, Convolutional codes.

3. Hybrid ARQ (ARQ+FEC):

   These are full duplex, error detection and correction codes. Hybrid ARQ performs better in poor signal conditions compared to ordinary ARQ.

From section 2.5 it was seen that after punctured convolutional coding, the OFDM symbols are generated. This accounts for the name "Coded" in COFDM. Also error correction process works well if the incoming bits are random. Therefore time and frequency interleaving is applied to the coded bits to help FEC to work properly and combat the effect of deep fades that may occur in the wireless channel.

The coding gain is given by the following general equation:

Gain [dB] = $(Eb/No)_{uncoded}$[dB] - $(Eb/No)_{coded}$[dB] (2.9)

2.7 DAB Transmitted Signal

The DAB transmitted signal [1] is built up around a transmission frame structure consisting of synchronization channel, the FIC and the MSC. This has been presented in Figure 2.4. The transmitted frame duration is denoted by T_F. Each transmission frame consists of a sequence of OFDM symbols. The number of OFDM symbols in a transmission frame depends on the transmission modes which will be explained in section 2.9. The first two OFDM symbols in each transmission frame are kept reserved for the synchronization channel.

The standard defines that the first OFDM symbol 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 Ts. Each of these OFDM symbols have set of equally spaced carriers, with carrier spacing 1/Tu.

The main DAB transmitted signal s(t) [1] is defined by the following formula:

$$s(t) = Re\left\{e^{2j\pi fct}\sum_{m=-\infty}^{\infty}\sum_{l=0}^{L}\sum_{k=-\frac{K}{2}}^{\frac{K}{2}} z_{m,l,k} \times g_{k,l}\left(t - mT_F - T_{NULL} - (L-1)T_S\right)\right\} \quad (2.10)$$

With ,

$$g_{k,l}(t) = \{\, e^{\frac{2j\pi k(t-\Delta)}{T_U}} . Rect\left(\frac{t}{T_S}\right) \qquad for\ l = 1,2,\dots\dots,L \quad (2.11)$$

But $g_{k,l}(t) = 0$ for $l = 0$ and $Ts = Tu + \Delta$

where,

- L is the number of OFDM symbols per transmission frame (the Null symbol being excluded);
- K is the number of transmitted carriers;
- T_F is the transmission frame duration;

T_{NULL} is the Null symbol duration;

Ts is the duration of the OFDM symbols of indices l=1,2,3,……., L;

T_U is the inverse of carrier spacing;

Δ is the duration of time interval called guard interval;

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

f_c is the central frequency of the signal.

All these parameters are given in Table 2.2 for each transmission mode [1], presented in next section.

2.8 DAB Modes and System Parameters

The Eureka 147 DAB [1] 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 use of these transmission modes depends on the network configuration and operating frequencies. This makes the DAB system operate over a wide range of frequencies from 30 MHz to 3 GHz.

As discussed in section 2.6.2 DAB transmission frame consists of FICs which is made up of FIBs and MSCs which is made up of CIFs. Table 2.1 shown below gives details of number of FIBs and CIFs for each transmission mode.

Table 2. 1: DAB transmission frame composition [1].

Transmission mode	Duration of transmission frame	Number of FIBs per transmission frame	Number of CIFs per transmission frame
I	96 ms	12	4
II	24 ms	3	1
III	24 ms	4	1
IV	48 ms	6	2

Table 2.2 shown below gives the details of DAB system parameters for all the four transmission modes. 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 [1] .

Parameter	Transmission mode -I	Transmission mode -II	Transmission mode -III	Transmission mode -IV
K	1536	384	192	768
L	76	76	153	76
T_F	196608 T 96 ms	49152 T 24 ms	49152 T 24 ms	98304 T 48 ms
T_{NULL}	2656 T 1297 ms	664 T 324 µs	345 T 168 µs	1328 T 648 µs
Ts	2552 T 1246 ms	638 T 312 µs	319 T 156 µs	1276 T 623 µs
T_U	2048 T 1 ms	512 T 250 µs	256 T 125 µs	1024 T 500 µs
Δ	504 T 246 µs	126 T 62 µs	63 T 31 µs	252 T 123 µs
Max. RF	375 MHz	1.5 GHz	3 GHz	750 MHz
Carrier spacing	1 kHz	4 kHz	8 kHz	2 kHz
FFT length	2048	512	256	1024

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

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

Transmission mode –III	is available for satellite broadcasting below 3 GHz (UHF L-Band).
Transmission mode –IV	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

2.9 Conclusion

The chapter discussed the detailed theoretical background of the Eureka 147 DAB system. It presented the working principle and technical overview of the DAB system. It also discussed about the COFDM technology which is the heart of DAB system.

The four DAB modes of transmission along with their standard parameters were also presented. The next chapter presents the simulation model in detail and its implementation in the MATLAB environment.

Chapter 3
THE SIMULATION MODEL

3.1 Introduction

This chapter describes the detailed method for modeling of the DAB transmission and receiving system strictly in accordance with the ETSI DAB standard [1] described in previous chapter. Transmission mode –II has been used in the simulation so all the standard parameters of this mode has been selected. After successful design of mode –II, all other modes has also been simulated.

MATLAB has been used as the software for simulation since it is very easy to understand having very good interactive environment that enables programmer to perform computationally intensive tasks faster than any other programming languages. It is well suited for design and analysis of complete digital communication systems.

All the simulation work has been developed in the baseband transmission and frame based processing is used. The simulation results are shown only for transmitted mode –II. Before simulating the main DAB system some basic simulation has been presented for performance of BPSK, QPSK and QAM in AWGN & Rayleigh fading channels. Therefore the results from this simulation are first presented in chapter 4. The DAB system was designed and simulated without MPEG audio coding, scrambling, time interleaving, and ADC/DAC and up/down converter. All the simulation results have been presented in chapter 4.

3.2 DAB Simulation Model

Figure 3.1 presents the complete details of the DAB system modeled which was simulated in MATLAB environment.

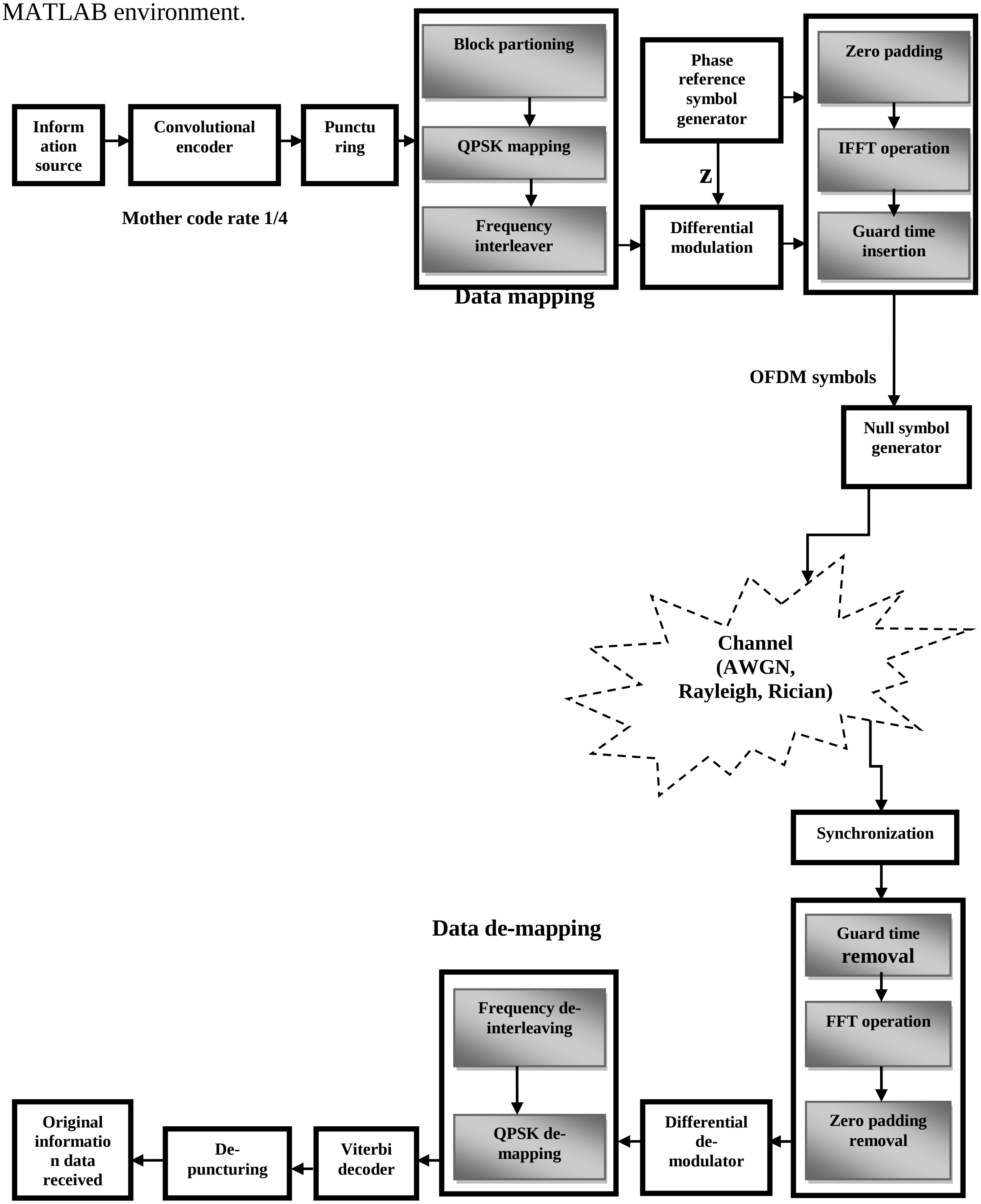

Figure 3. 1: Complete block diagram of DAB system for Simulation [1] [4].

3.3 Information source

This is the first block in the transmitter section of the DAB system model for simulation. It generates random binary data bit sequence for FIC and MSC. Therefore the data for one transmission frame is given by:

$$DATA\_bits = FIC\_DATA + MSC\_DATA \tag{3.1}$$

As discussed in section 2.9 each transmission frame for mode –II has 76 OFDM symbols. First OFDM symbol is reserved for phase reference symbol, next 3 OFDM symbols are for FIC and rest 72 symbols are for MSC channel. For 24 ms transmission frame there is only 1 CIF that constitutes the MSC data and each FIC has 3 FIBs.

Total data bits for FIC and MSC can be calculated from parameters given in Table 2.2. The number of sub-carriers for transmission mode –II is 384. Since QPSK mapping is done therefore there are 2 bits per carrier. Thus bits per OFDM symbol equals 768 bits (384*2).

FIC_DATA = No. of OFDM symbols*bits/OFDM symbol => 3*768 = 2304 bits.
MSC_DATA = No. of OFDM symbols*bits/OFDM symbol => 72*768 = 55296 bits.

The total data bits for each transmission frame can be easily calculated from (3.1) as 57600 bits. The MATLAB function “randint” was used to generate the random data bits for transmission. Randint(M) generates an M-by-M matrix of random binary numbers "0" and "1" with equal probability.

3.4 Convolutional Encoder

The output data stream Tx_bits from previous block is input to convolutional encoder. Channel coding is based on punctured convolutional forward-error-correction (FEC) which allows both Equal and Unequal Error Protection (UEP) described in section 2.6.2. DAB system has a convolutional encoder with constraint length 7 and octal forms of generator polynomials are 133, 171, 145 and 133, respectively [1]. The mother code has a rate R =1/4 which means that each input bit is protected by four bits. Thus expand rate is 1/R.

No MATLAB code has been written for this block. The built in function “convenc” was used to generate convolutionally encoded bits. MATLAB Communications Toolbox [14] supports feed forward or feedback convolutional codes that can be described by a trellis structure or a set of generator polynomials. It uses Viterbi algorithm to implement hard-decision and soft-decision decoding. The reader is referred to [11] for detailed information regarding constraint length, generator polynomial and trellis structure.

The output of this block is coded bit stream having gross data rate of 230400 bps since mother code has a rate R=1/4.

3.4.1 Puncturing

According to [1] some predefined code-bits generated by the mother code in section 3.4 is not transmitted. This process is called puncturing. Puncturing has the same effect as encoding with an error correction code with a higher rate, or less redundancy. Puncturing considerably increases the flexibility of the system without significantly increasing system complexity.

3.4.1.1 Puncturing of FIC

The puncturing procedure for Fast Information Channel (FIC) is fixed and provides a high protection level against channel impairments. The DAB standard [1] defines 24 puncturing vectors with an equivalent code rate (refer to Appendix A).

The first 4I=3072 bits of the serial mother code at the output of convolutional encoder should be split into 24 consecutive blocks of 128 bits. This is for audio bit rate of 32 Kbit/s. The first 21 blocks will be punctured according to puncturing index PI=16. The remaining three blocks will be punctured by puncturing index PI=15. This gives a code rate of approximately 1/3. Finally, the last 24 bits of the mother code should be punctured according to puncturing vector given by :

$$\mathbf{V_T} = (1100\ 1100\ 1100\ 1100\ 1100\ 1100) \tag{3.2}$$

Resulting 12 bits are called tail bits. All these blocks are then grouped together and with tail bits appended to the last block. The resulting word is called punctured codeword and has 2304 bits. Zero padding bits shall be appended at the end of punctured codeword for certain puncturing schemes to ensure a word length of 64 bits [1].

For $\mathbf{V_{PI,i}} = 0$, the corresponding bit should be taken out of the block and shall not be transmitted.
For $\mathbf{V_{PI,i}} = 1$, the corresponding bit should retained and shall be transmitted.

3.4.1.2 Puncturing of MSC

The puncturing of Main Service Channel (MSC) is defined in terms of protection profiles and protection levels. As discussed in section 2.6.2 there are five protection levels from PL1 to PL5 defined for different audio bit rates. Unequal Error Protection (UEP) will be explained which is designed for audio but can also be used for data. Protection level 1 has been implemented in simulation.

For audio bit rate of 32 Kbit/s the first 4I=3072 bits of the serial mother code at the output of convolutional encoder is split into 24 consecutive blocks of 128 bits. The first L_1 blocks are punctured according to puncturing index PI_1. The next L_2 blocks by puncturing index PI_2. The next L_3 blocks with puncturing index PI_3. The remaining L_4 blocks are punctured according to PI_4. Finally, the last 24 bits of the mother code is punctured according equation 3.5 given above. Table 3.2 shown below explains the protection profiles for five protection levels for audio rate of 32 Kbit/s.

Table 3. 1: Five protection levels for audio rate of 32 Kbit/s [1].

Audio bit rate (Kbit/s)	P	L_1 L_2 L_3 L_4	PI_1 PI_2 PI_3 PI_4	Number of padding bits
32	5	3 4 17 0	5 3 2 -	0
32	4	3 3 18 0	11 6 5 -	0
32	3	3 4 14 0	15 9 6 8	0
32	2	3 4 14 3	22 13 8 13	0
32	1	3 5 13 3	24 17 12 17	4

3.4.2 Concatenated coding

The virtually error free channel can be achieved by concatenated coding using a convolutional code (‘inner code’) together with a Block code (‘outer code’) [2]. This technique also improves the BER performance in different transmission channels as will be shown in subsequent sections. Figure 3.2 shown below explains the concept of concatenated coding.

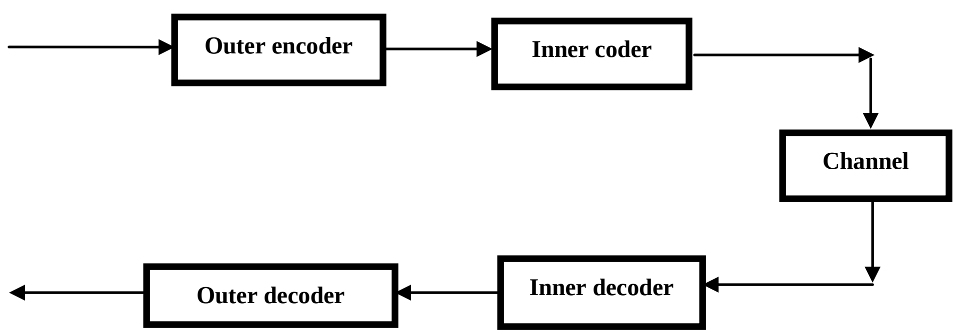

Figure 3. 2: Block diagram of concatenated coding.

Typically, the inner code is not a block code but a soft-decision convolutional Viterbi-decoded code with a short constraint length. For the outer code, a longer hard-decision block code, frequently Reed Solomon with 8-bit symbols, is selected. This project uses linear block code, cyclic code, hamming code and BCH (Bose-Chaudhuri-Hocquenghem) code as the outer encoder. The MATLAB function ‘encode’ is used for linear block code, cyclic code and hamming code and ‘bchenc’ for BCH code.

Code word length n was taken 511 and message length to be 502 for the former and (n. k) was taken to be (511,439) for the later which gives error correction capability t = 8.

3.5 Data Mapping

The three main task of this block are block partitioning, QPSK symbol mapping and frequency interleaving. The following sub-sections describe these operations in detail.

3.5.1 Block partitioning

The task of this sub-block is to divide the punctured convolutional code bits at its input into blocks of data which will be associated to OFDM symbols. This operation is transmission mode dependent. The convolutional codeword of size 57600 bits is divided into 75 consecutive blocks of 768 bits to be transmitted as OFDM symbols of index l=2,3,4,……76.

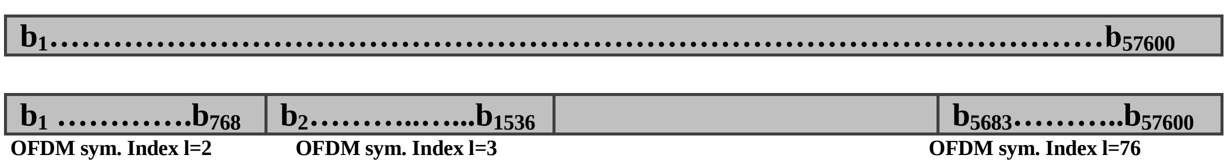

Figure 3.3: Method of Block partitioning.

3.5.2 QPSK mapping

The QPSK digital symbol mapping block is responsible for mapping (in parallel) serial bit stream in each data block into a digital constellation according to QPSK modulation scheme given in DAB standard [1], defined as:

$$q_{l,n} = \frac{1}{\sqrt{2}}[(1 - 2b_{l,n}) + j(b_{l,n+k})] \tag{3.3}$$

Where n = 1, 2,…..,K and l = 2,3,4,……,76. and K is the number of carriers used.

Each data block of 768 bits is mapped onto the 384 complex coefficients for one OFDM symbol of duration T_S. The first 384 bits will be mapped to the real parts of the 384 QPSK symbols, the last 384 bits will be mapped to the imaginary parts. QPSK encodes two bits per symbol. Therefore output of this block is complex consisting of 75 blocks of 384 bits.

Figure 3.4 shows the simulated QPSK constellation mapping without noise addition. It can be seen that there are no distortions in mapped symbols.

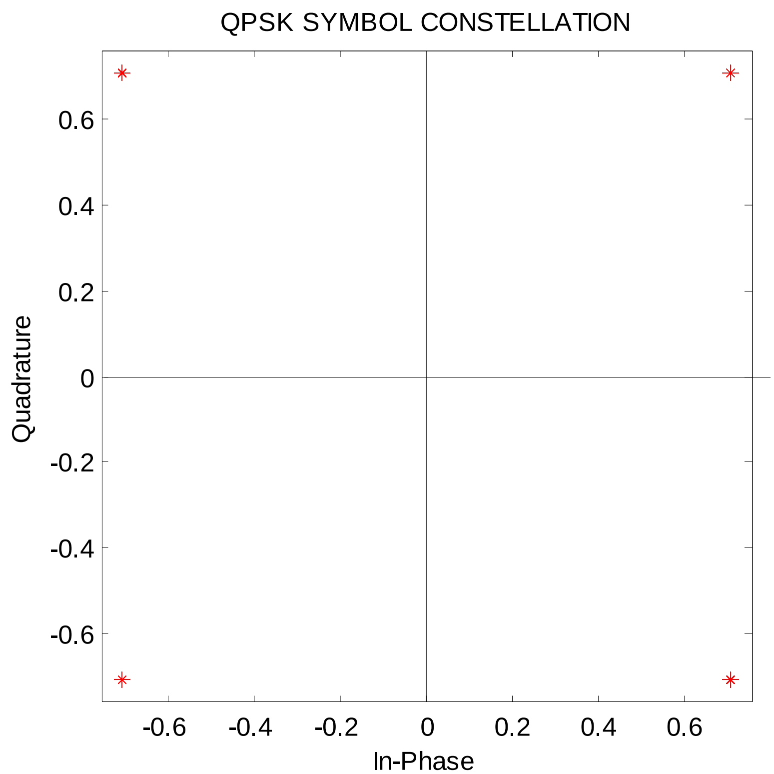

Figure 3. 4: QPSK constellation diagram.

Figure 3.5 presents the QPSK constellation plot contaminated with AWGN noise with 20 dB SNR. It may be seen that symbols are distorted.

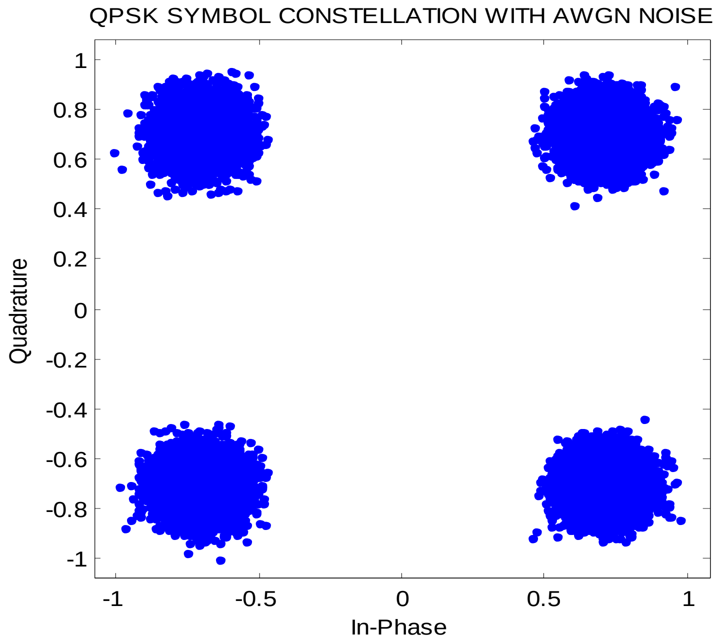

Figure 3. 5: QPSK constellation diagram with AWGN noise.

3.5.3 Frequency interleaving

Frequency interleaving is used to eliminate the effects of selective fading. It offsets any deep fades that occur in the wireless channel by spreading the data bits over the sub-carrier channels. Frequency interleaving [1] defines the correspondence between the index n of the QPSK symbols q(l, n) and the carrier index k ($-K/2 \le k < 0$ and $0 < k \le K/2$). The QPSK symbols shall be re-ordered according to the following relation:

$y(l,k) = q(l,n)$ for $l = 2,3,4,\ldots,L$ and with $k = F(n)$ (3.4)

where F is a function defined in the next terms for transmission mode –II.

Let $\pi(i)$ be a permutation in the set of integers i = 0, 1, 2,...,511 obtained from the following relation:

$\pi(i) = 13\,\pi(i-1) + 127 \pmod{512}$ and $\pi(0) = 0$; for i = 1, 2, ...,511. (3.5)

$d_n = \pi(i)$ (excluding 256) (3.6)

The frequency interleaving rule between QPSK symbols and carrier index is defined as

$$k = F(n) = d_n - 256 \tag{3.7}$$

The interleaving rule is illustrated in Table 3.2 below:

Table 3. 2: Frequency interleaving rule for transmission mode-II [1] .

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

Figure 3.6 and Figure 3.7 presents the effect of with and without frequency interleaving for real part of QPSK symbols.

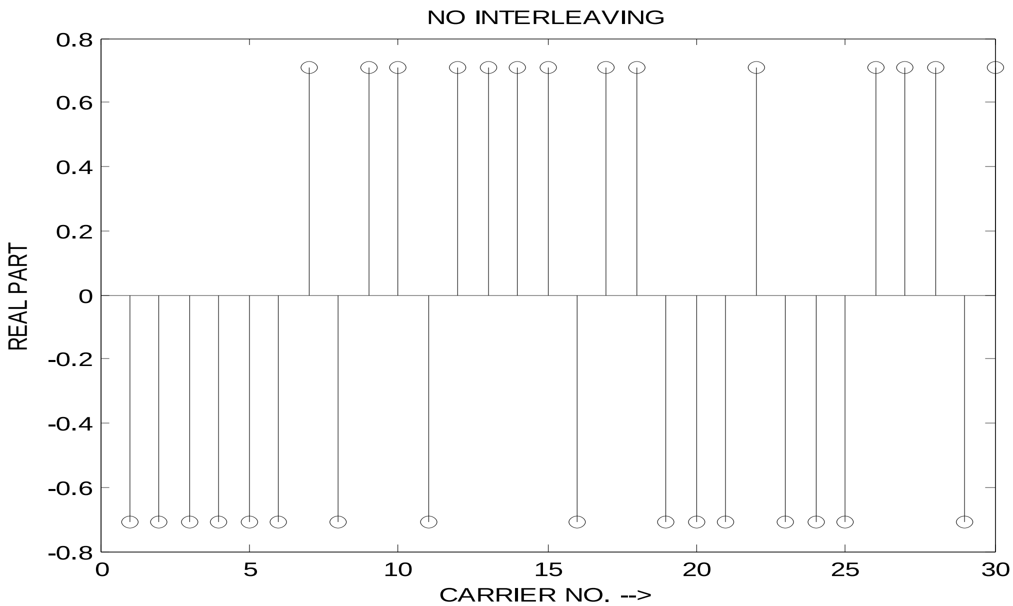

Figure 3. 6: Without frequency interleaving.

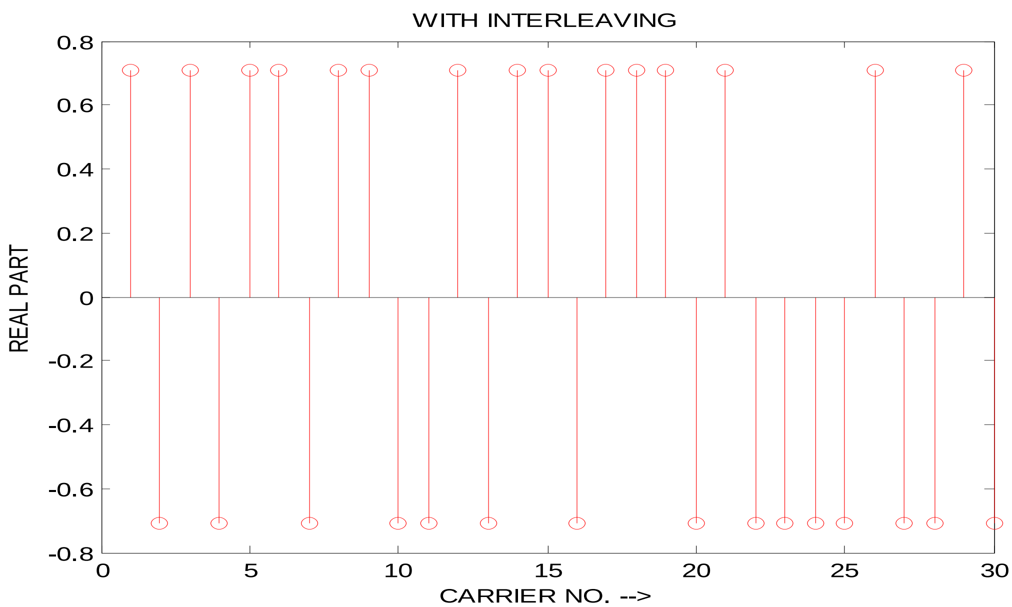

Figure 3. 7: With frequency interleaving.

3.6 Phase Reference Symbol Generator

According to DAB standard [1] the first OFDM symbol (without taking account Null symbol) in the transmission frame is the phase reference symbol which helps in receiver synchronization.

Since it occurs once in a frame therefore the detection of this symbol can be used for frame synchronization.

It serves as reference for the differential modulation for the next OFDM symbols in the transmission frame. The phase reference symbol is defined [1] in the following expression:

$$z_{l,k} = \begin{cases} e^{j\varphi_k} & for -\frac{K}{2} \leq k < 0 \; and \; 0 < k \leq \frac{K}{2} \\ 0 & for \; k = 0 \end{cases} \tag{3.8}$$

where

$$\varphi_{k=\frac{\pi}{2}(\,h_{i,k-k'}+n)} \tag{3.9}$$

The values of indices i, k’ and the parameter n are given as functions of the carrier index k for all the DAB transmission modes (refer to Appendix B). The values of the parameter $h_{i,j}$ is given as a function of its indices i and j is also given in Appendix B.

Figure 3.8 shows the simulated output for real part of the phase reference symbol which appears to be noise like waveform. Figure 3.9 presents its constellation diagram.

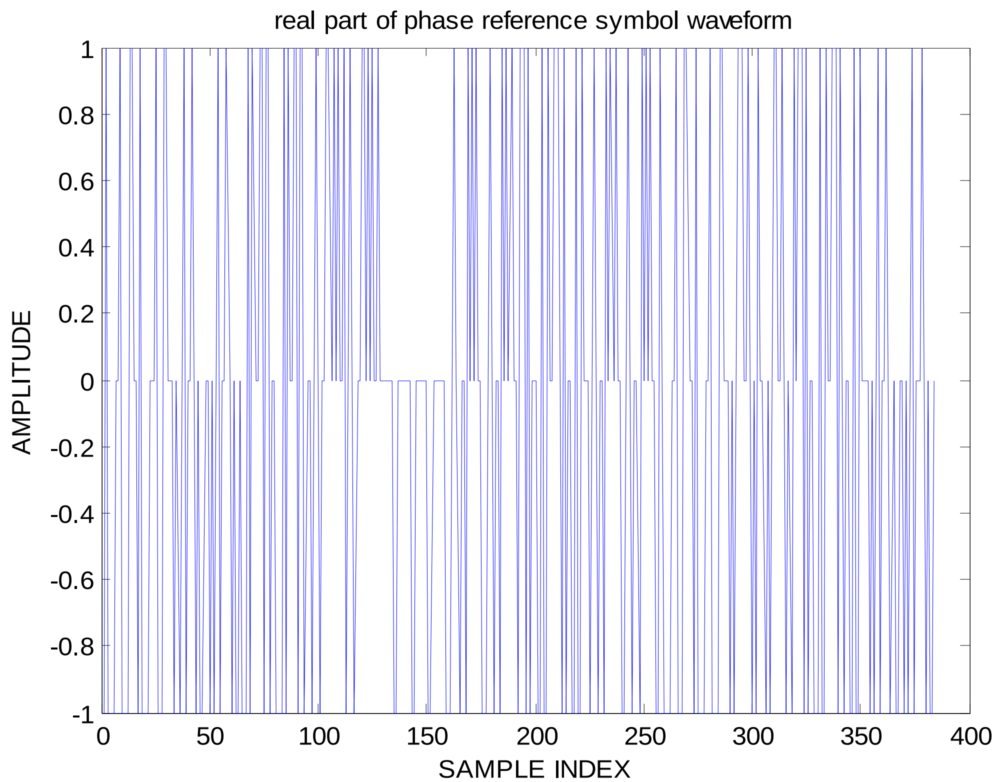

Figure 3. 8: Real part of the Phase reference symbol.

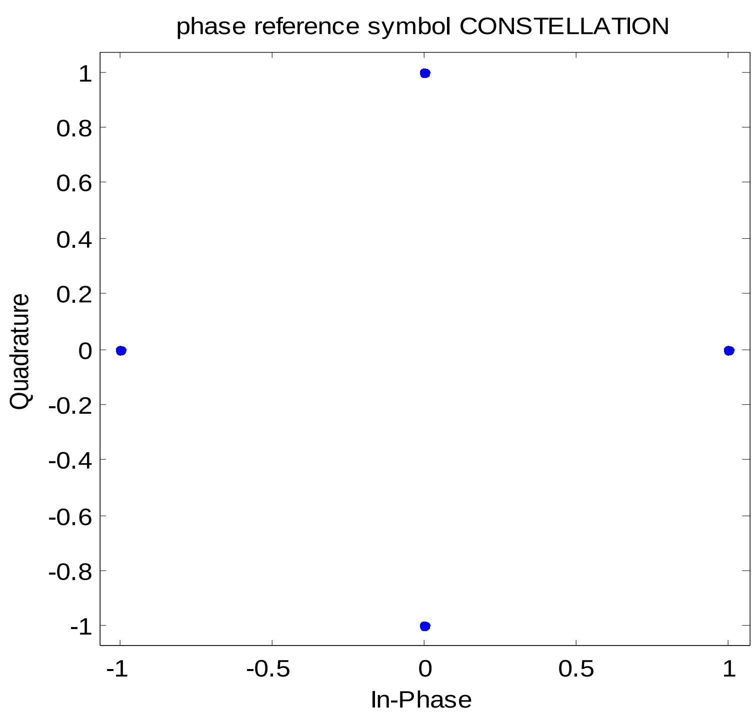

Figure 3. 9: Phase reference symbol constellation diagram.

3.7 Differential Modulation

In mobile communications the multipath effect can degrade the phase of the carriers. The solution to this problem is to send the information as the difference between the phases of two symbols. This is accomplished by this block which performs $\pi/4$ shifted differential QPSK modulation.

In this modulation scheme there is no absolute phase reference, each symbol is referenced only against the previous symbol. This greatly simplifies the decoder. Apart from being bandwidth efficient the additional $\pi/4$ phase shift resolves phase ambiguity of ordinary D-QPSK.
Differential modulation is applied to QPSK symbols on each carrier and is defined [1] by:

$$z_{l,k}=z_{l-1,k}\times y_{l,k} \qquad (3.10)$$

where $l=2,3,4,\ldots,L$ and $-K/2 \leq k \leq K/2$.
z is the complex D-QPSK symbol and y is the input QPSK symbol.

3.8 OFDM Symbol Generator

This block is the heart of the DAB system as discussed in section 2.6.3. The OFDM technology in DAB makes it robust against multipath fading environment delivering high quality audio services. The detail of OFDM symbol generation will be explained in following sub-sections.

3.8.1 Zero padding

As discussed in section 2.7.1 the IFFT/FFT algorithm works efficiently if the number of carriers is a power of two. The output D-QPSK symbol from section 3.7 has length of 384. The DAB system parameters presented in Table 2.2 shows that the FFT length for transmission mode-II is 512. Therefore zero padding is necessary to make it power of two (i.e., from 384 to 512).
To work with 512 FFT lengths this sub-block adds 128 zeros to each D-QPSK symbol block as illustrated in Figure 3.10 below.

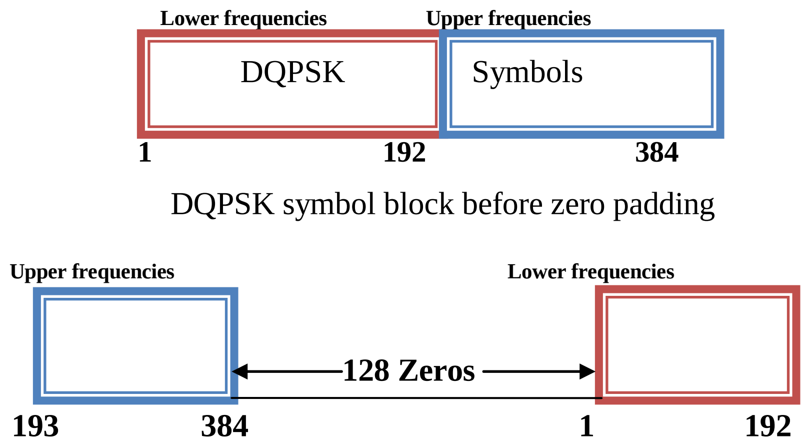

Figure 3. 10: D-QPSK symbol block before and after zero padding and rearrangement.

3.8.2 IFFT operation

This sub-block is the heart of OFDM technology. It performs 512 point IFFT operation on each D-QPSK block obtained after zero padding. Thus frequency domain samples in each D-QPSK block is transformed into time domain samples. MATLAB function ‘ifft’ has been used.

3.8.3 Guard time insertion

This sub-block is responsible for making OFDM symbols resistant to inter symbol interference. It takes copy of last 126 samples equal to guard interval (according to Table 2.2) from each OFDM symbol and place it at the beginning of the OFDM symbol. This makes the length of OFDM symbol equal to 638 samples equivalent to OFDM symbol duration T_S.

3.9 Null Symbol Generator and final DAB frame

This is the last block of the DAB transmitter and its addition completes the final DAB frame structure for transmission. Null symbol has duration of T_{NULL} equivalent to 664 samples. Thus 664 zeros are generated by using MATLAB function ‘zeros’ and appended at the beginning of the frame. It may be recalled that null symbol is the first OFDM symbol. During null symbol period no information is transmitted. Figure 3.11 shows the simulated final complex DAB frame.

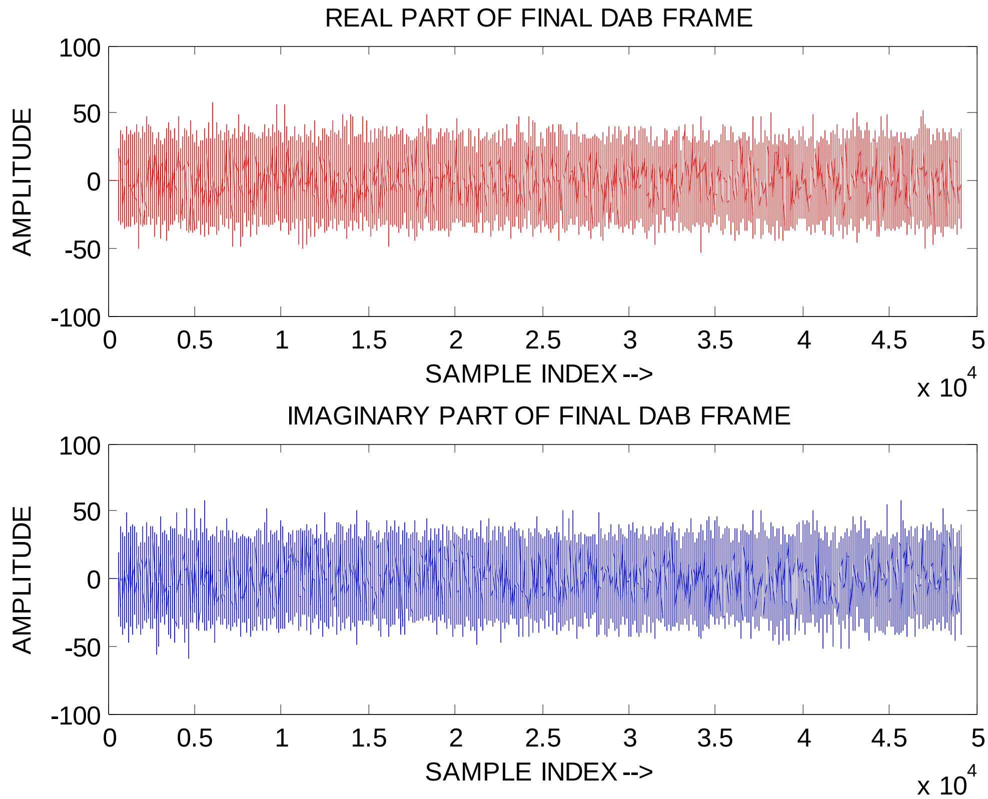

Figure 3. 11: Simulated DAB frame in time domain.

3.10 Channel

Channel is the physical transmission medium through which the final DAB signal generated by the transmitter section is passed for BER performance analysis. This is accomplished by selecting AWGN channel, Rayleigh fading channel and Rician channel from the "Channels" library in the MATLAB Communications Toolbox [12].

The mobile radio channel is characterized by time variance and frequency selectivity. The Doppler frequency shift is the main parameter for mobile channels. Because of relative speed between the mobile receiver and the fixed transmitter there results a frequency shift in the incoming signal. The maximum Doppler frequency shift (in Hz) is calculated by the following formula [2]:

$$f_D = \frac{v}{c} f_o = \frac{1}{1080} \frac{f_o}{MHz} \frac{v}{km/hr} Hz \tag{3.11}$$

Where f_o is the transmission frequency and v is the speed of the vehicle.

Figure 3.12 shows the received VHF signal level for a car moving at 192 km/hr as a function of time for a carrier frequency of 225 MHz.

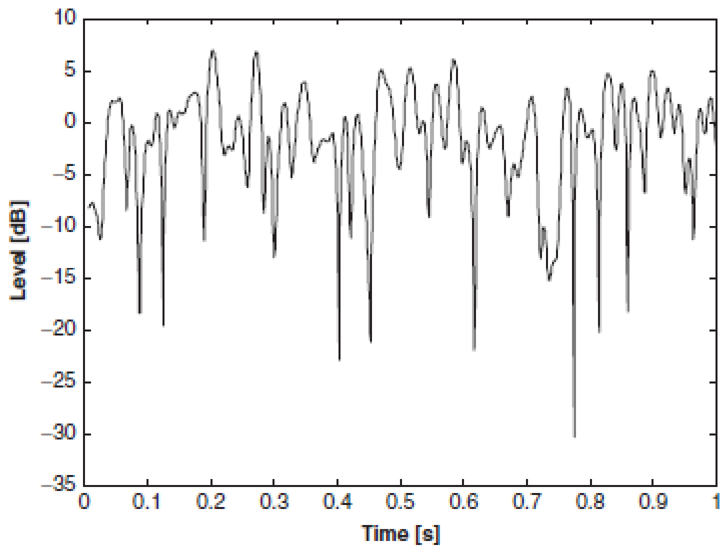

Figure 3. 12: Time variance due to multipath channel [2].

Table 3.3 shown below gives Doppler frequencies for different vehicle speed and carrier frequency.

Table 3. 3: Doppler frequencies for different vehicle speeds [2].

f_{dmax}	V= 48 km/hr	V= 96 km/hr	V=192 km/hr
f_o=225 MHz	10 Hz	20 Hz	40 Hz
f_o=900 MHz	40 Hz	80 Hz	160 Hz
f_o=1500 MHz	67 Hz	133 Hz	267 Hz

3.11 Spectrum Characteristics

The power spectral density (PDF) $P_k(f)$ [1] of each carrier at frequency $f_k = f_c + k/T_u$ $(-\frac{K}{2} \leq k < 0 \text{ and } 0 < k \leq \frac{K}{2})$ is defined as :

$$P_K(f) = \left[\frac{\sin\pi(f-f_k)T_S}{\pi(f-f_k)T_S}\right]^2 \tag{3.12}$$

The overall PSD is the sum of power spectral densities of each carrier. The bandwidth of DAB signal as specified in the standard [1] is 1.536 MHz, therefore any signal component outside the nominal bandwidth can be removed by appropriate filtering. Figures presented below show the simulated theoretical DAB signal spectrum (based on equation 3.12) for all the four transmission modes.

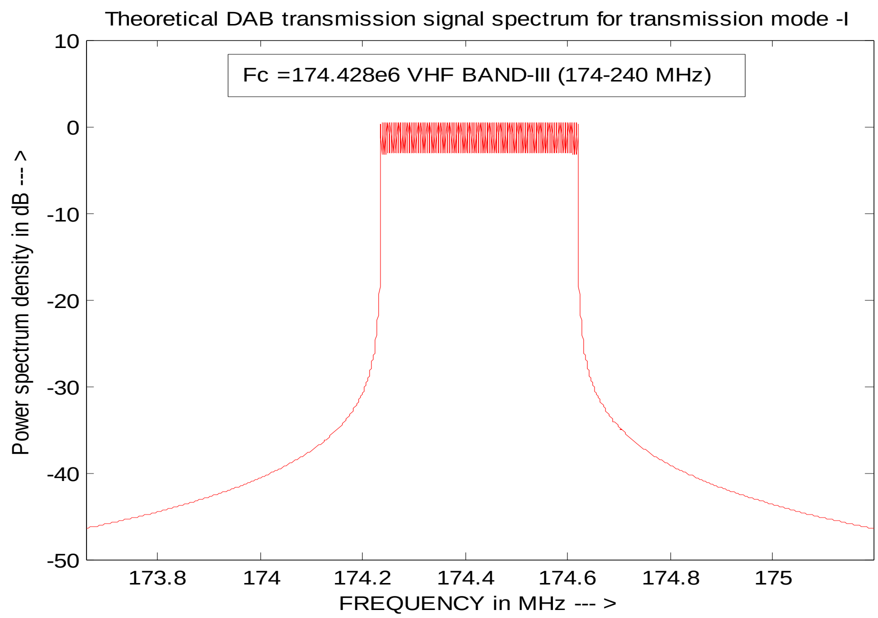

Figure 3. 13: DAB signal spectrum for TM-I.

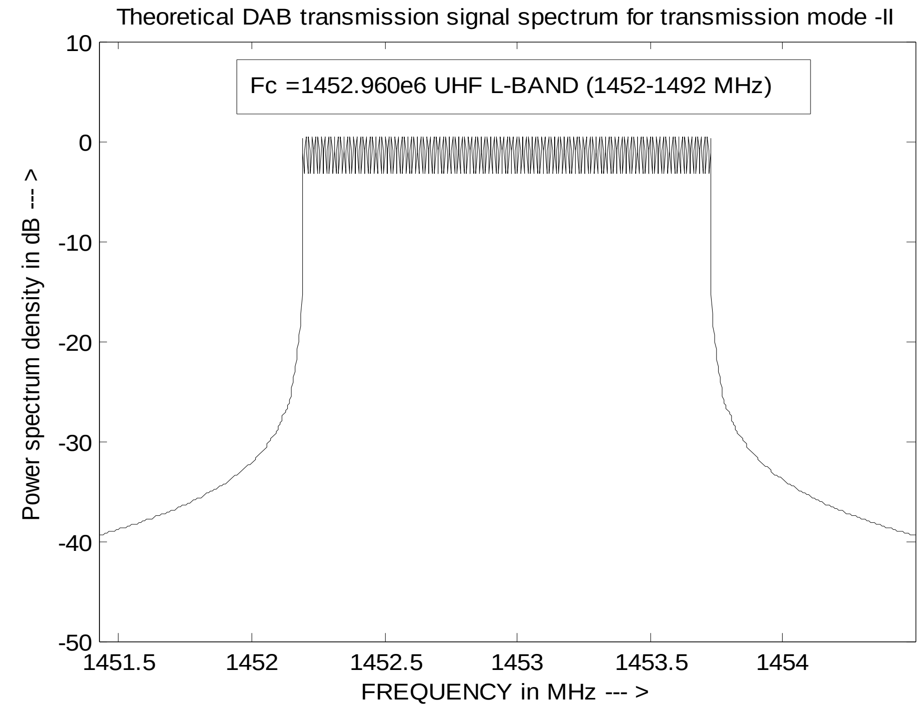

Figure 3. 14: DAB signal spectrum for TM-II.

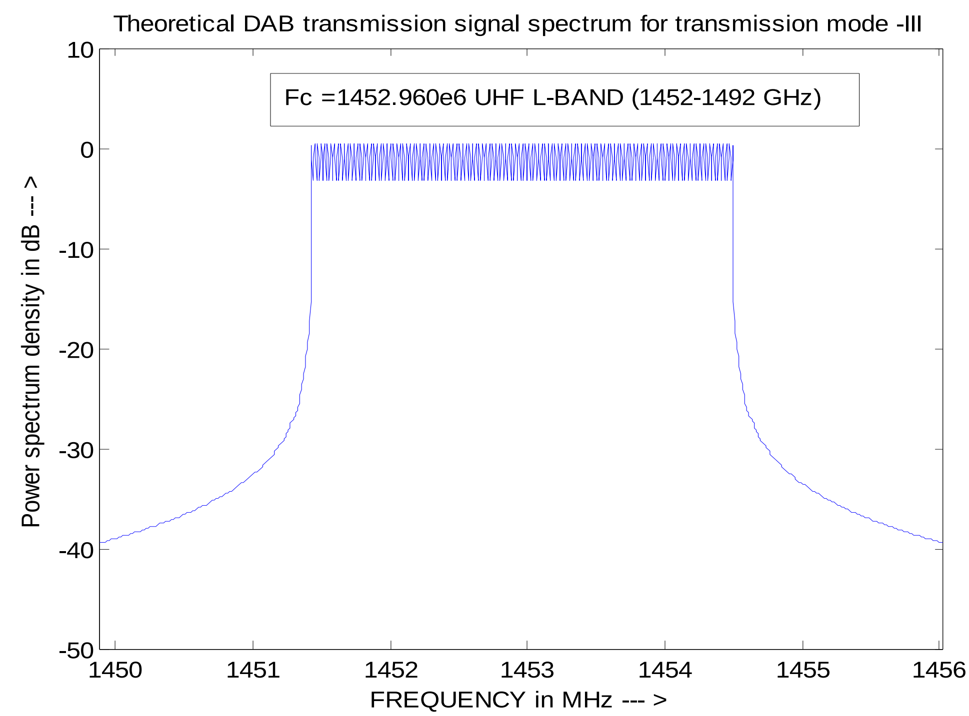

Figure 3. 15: DAB signal spectrum for TM-III.

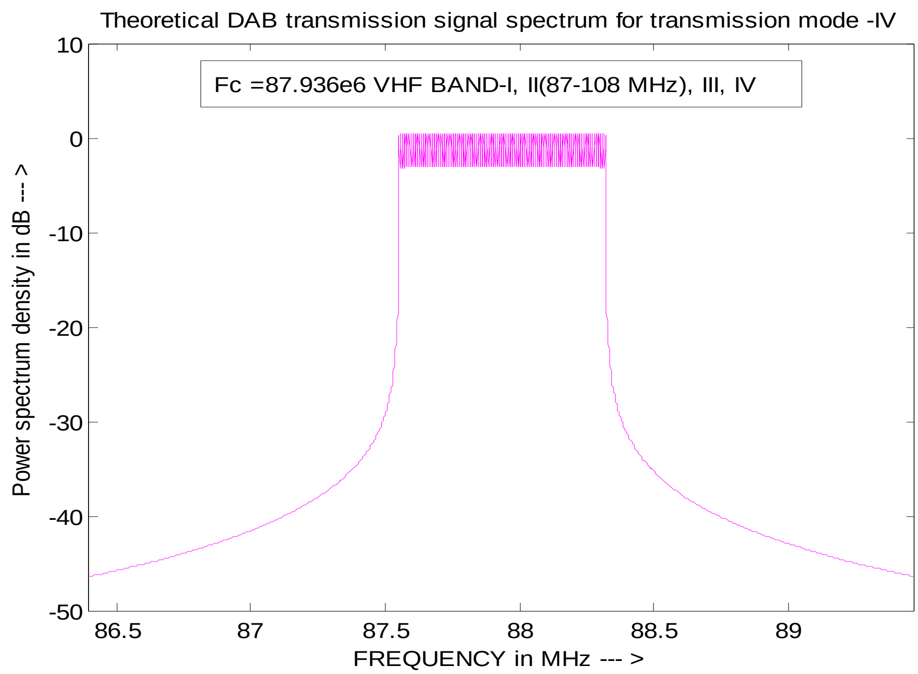

Figure 3. 16: DAB signal spectrum for TM-IV.

The practical values of centre frequency Fc for simulation was obtained from [3]. It may be evaluated from above Figures that all the four transmission modes have a bandwidth of 1.536 MHz which is in exact conformity with the DAB standard [1].

The analysis of simulated DAB signal spectrum will be presented next. SNR (signal-to-noise ratio) was taken to be 15 dB. The bandwidth of the signal is represented in baseband mode.

The final DAB frame after Null symbol addition was used to obtain the spectrum of the transmitted spectrum and the output of the channel before synchronization was used to obtain the received signal spectrum (according to Figure 3.1).

Figure 3.17 presents the transmitted DAB signal spectrum.

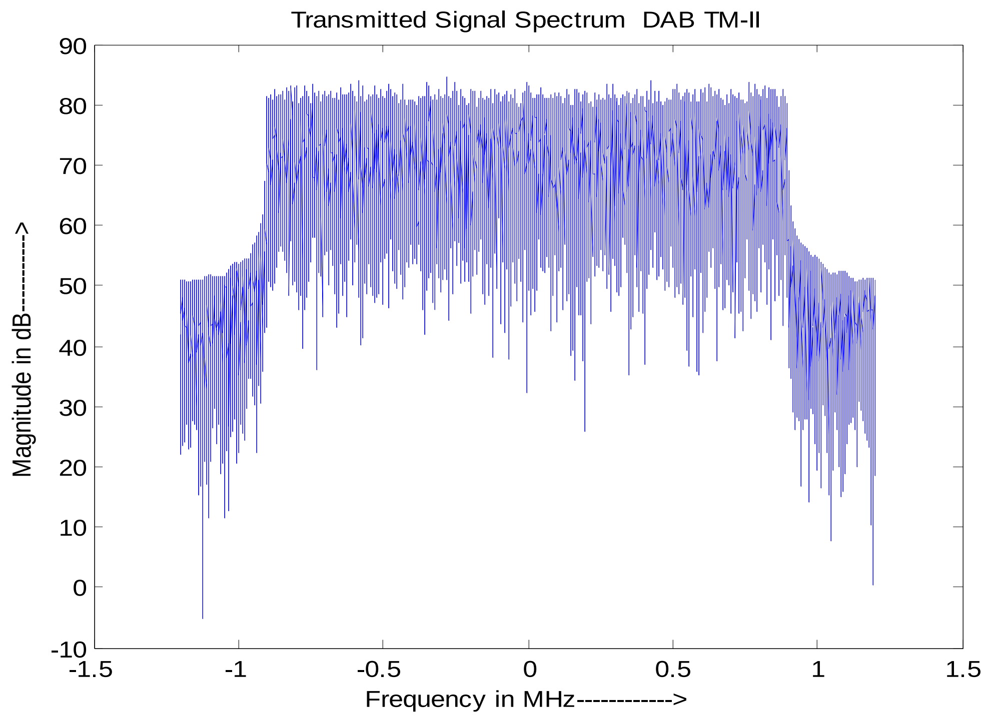

Figure 3. 17: Simulated transmitted signal spectrum.

Figure 3.18 shows the received DAB signal spectrum (mode-II) in AWGN channel.

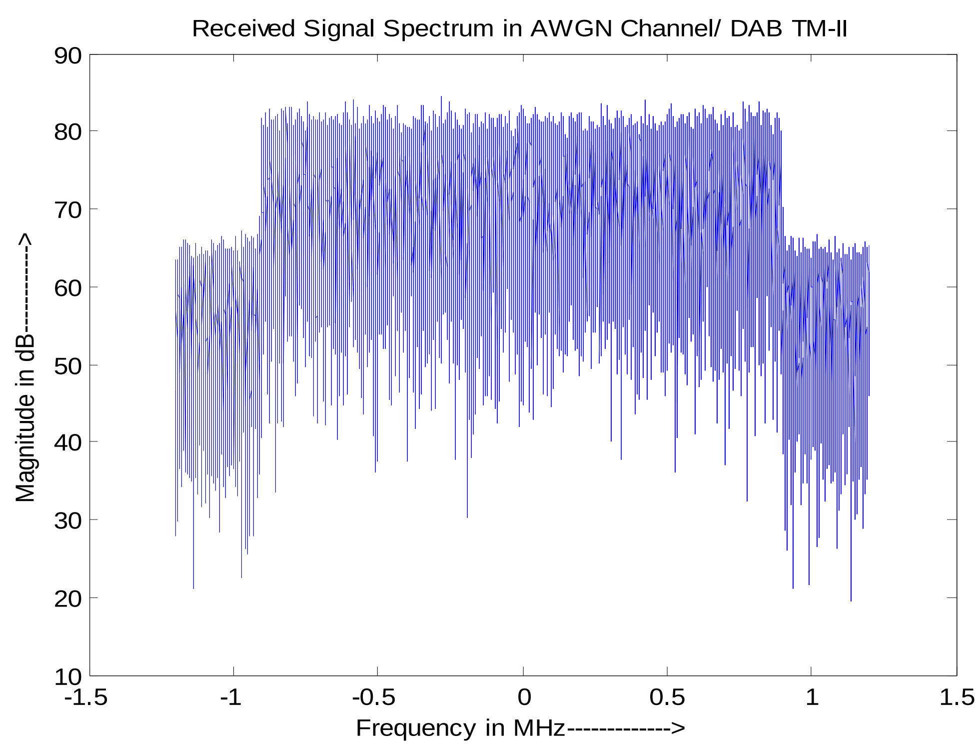

Figure 3. 18: Simulated received signal spectrum in AWGN channel.

Figure 3.18 reveals that received signal spectrum in AWGN channel has approximately the same power level as transmitted signal.

Figure 3.19 shows the received DAB signal spectrum (mode-II) in Rayleigh fading channel.

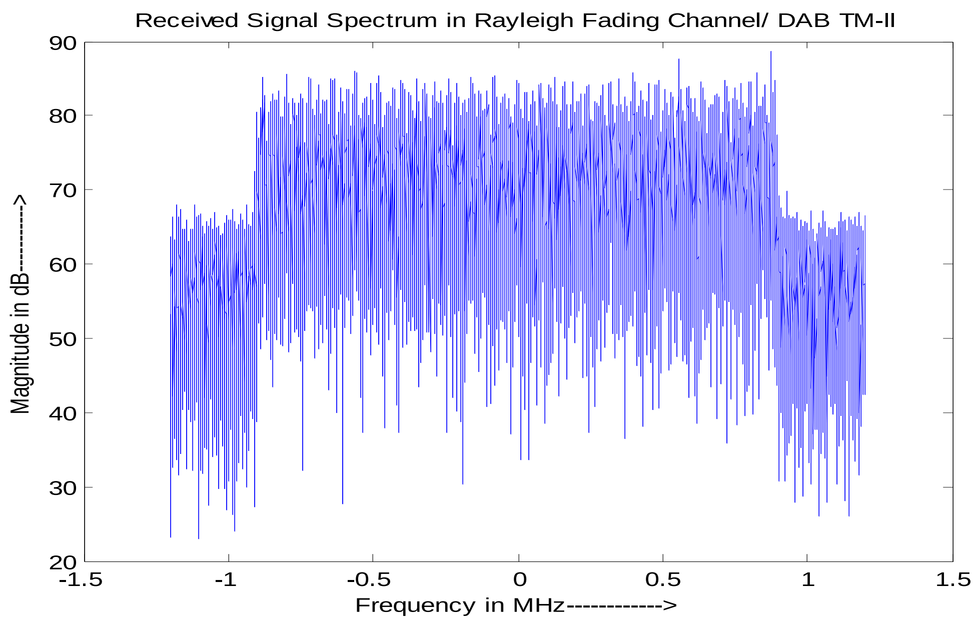

Figure 3. 19: Simulated received signal spectrum in Rayleigh fading channel.

Figure 3.19 shows that the power level of received signal spectrum in Rayleigh fading channel is 2 dB less than the transmitted signal.

Figure 3.20 presents the received DAB signal spectrum (mode-II) in Rician channel.

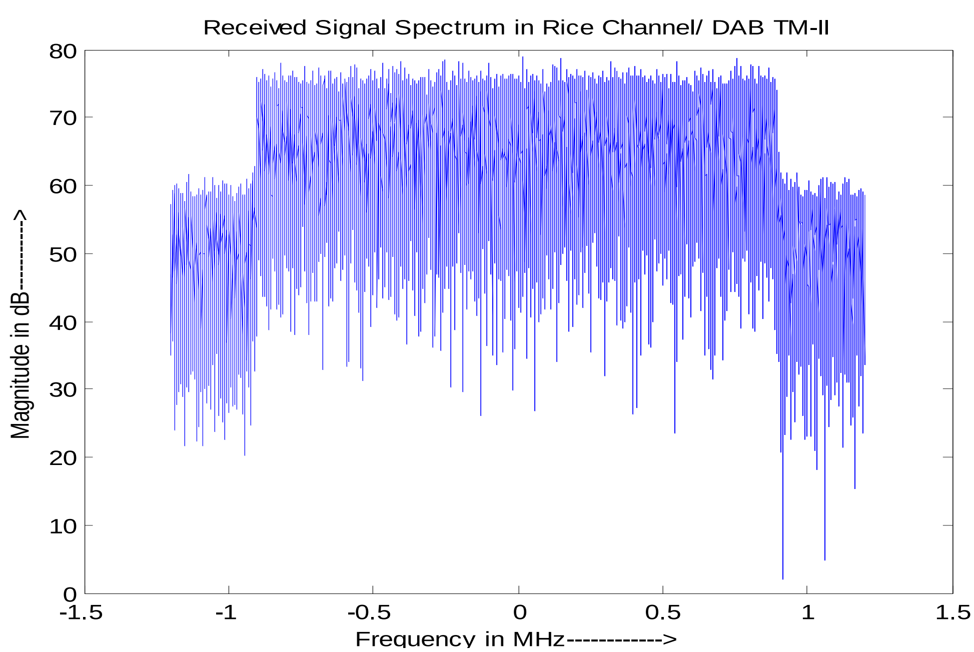

Figure 3. 20: Simulated received signal spectrum in Rician channel.

Finally, the Figure 3.20 shows that the power level of received signal spectrum in Rician channel is 5 dB less than the transmitted signal. Thus network gain is less in case of Rician channel.

3.12 Receiver Synchronization

The ETSI DAB standard [1] explains only about the transmitter side of Digital audio broadcasting. The DAB receiver will be designed exactly in reverse way of the task performed for the transmitter.

All digital communication systems require proper synchronization for decoding of the received signal in order to produce the original information transmitted. The synchronization block is used to locate precisely each DAB frame, so that the demodulation can be performed frame by frame, symbol by symbol. DAB system explores the Null symbol and the phase reference symbol for synchronization purpose. Figure 3.21 illustrates the process of receiver synchronization.

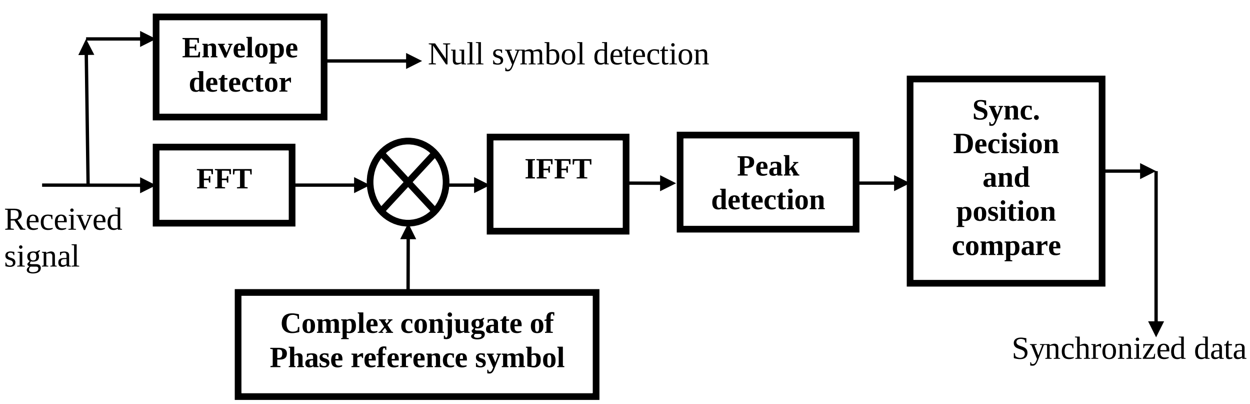

Figure 3. 21: Block diagram of Symbol and Frame synchronization [13] [4].

3.12.1 Fine time synchronization

Fine time synchronization or symbol timing synchronization [4] is performed by calculating the Channel Impulse Response (CIR) based on the actually received time frequency phase reference symbol (PRS) and the specified PRS stored in the receiver. To estimate the CIR, training Sequences (PRS in case of DAB system) are used. This means that a part (or the whole) of the transmitted signal is known from the receiver. As the receiver knows which signal is supposed to be observed, it can evaluate the distortion induced by the propagation channel and the modulation & demodulation stages.

Fine time synchronization is based on the phase reference symbol which is the dedicated pilot symbol in each DAB transmission frame [1]. Since the modulation each carrier is known [14], multiplication of received PRS with complex conjugate of PRS at the receiver results in cancellation of the phase modulation of each carrier. The phase reference symbol can be converted to impulse signal or CIR can be obtained by an IFFT operation of the resultant product as illustrated in following formula:

$$\text{CIR} = \text{IFFT}\{\text{Received PRS} \bullet \text{PRS}^*\} \quad (3.13)$$

Where PRS* is the complex conjugate of the phase reference symbol.

The peak of the impulse signal obtained from 3.13 will give position of the start of the PRS compared to a set threshold (T) providing symbol timing as well as frame timing. According to Figure 3.21 from the received signal a data sample block of FFT length is taken. Then FFT operation is performed on the block to convert the samples into frequency domain. Since FFT window length is 512 and size of PRS at the receiver is 384 (mode-II) therefore zero padding removal and data rearrangement has to be done [4]. The resulting sample block is of size 384 same as PRS. Now sample block is multiplied by the complex conjugate of the PRS known at the receiver which is then transformed into impulse signal in time by performing IFFT operation on the product (refer to equation 3.16).

The highest peak detection will indicate the start position of the PRS. To get a precise synchronization decision the peak obtained from every sample block taken from the received signal is compared to set threshold level (T). When the detected peak is less than the threshold level, then the peak found is not the desired peak and does not indicate the accurate start of the PRS. So the loop process has to be continued by taking the next sample block till the desired peak is obtained. The peak will be greater than the threshold only for the sample block which has phase reference symbol in it, since PRS have a high correlation with itself.

The Threshold level is determined by observing the magnitude [4] of the highest peak obtained by multiplication of the PRS with its complex conjugate and IFFT applied to the product, both in presence and absence of noise. The Figure 3.22 presents the phase reference impulse symbol in presence and absence of noise.

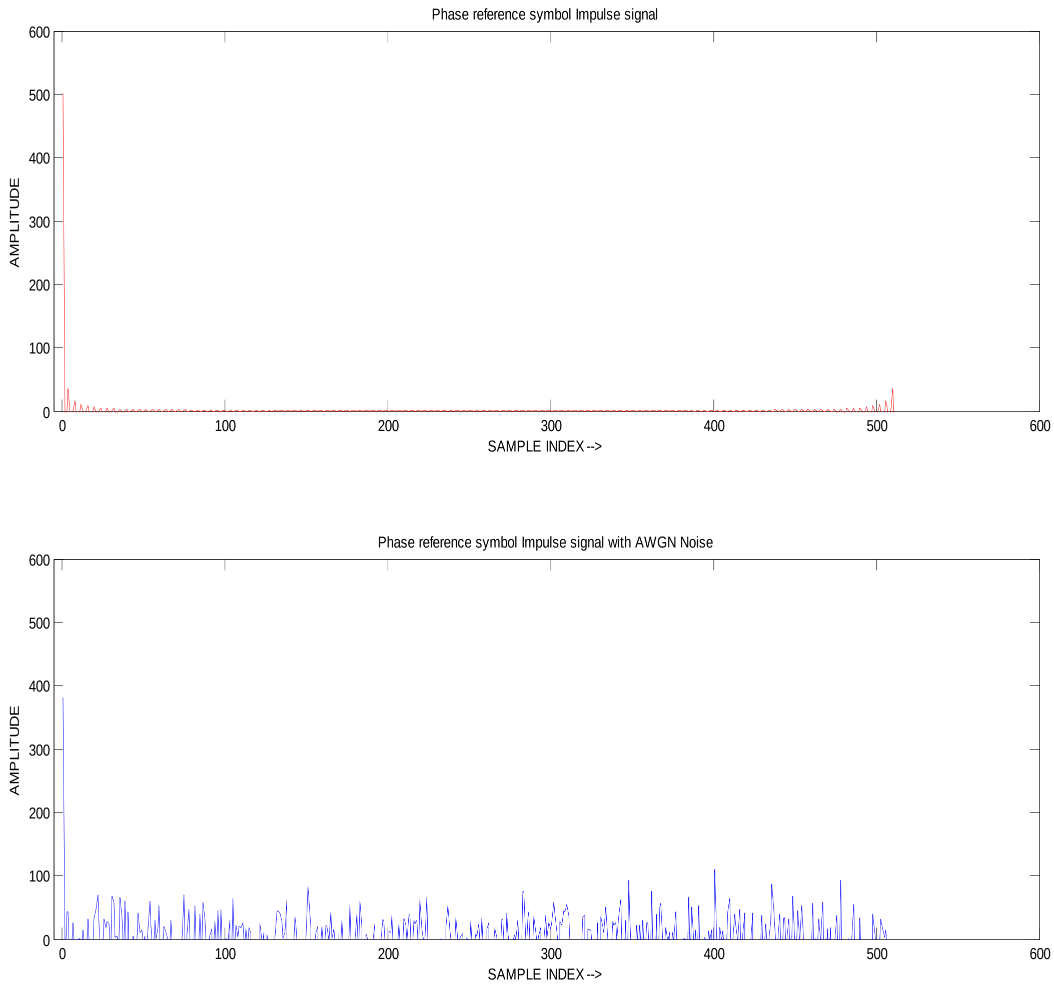

Figure 3. 22: Threshold determination using PRS.

From Figure 3.22 it is observed that the highest peak is obtained in the absence of noise therefore the threshold level was set to be greater than half the magnitude of this peak. This ensures that noise peak will not be mistaken as desired peak during peak detection. Threshold level was set to be T= 380.

The Figure 3.23 presented below shows the successful detection of the desired peak.

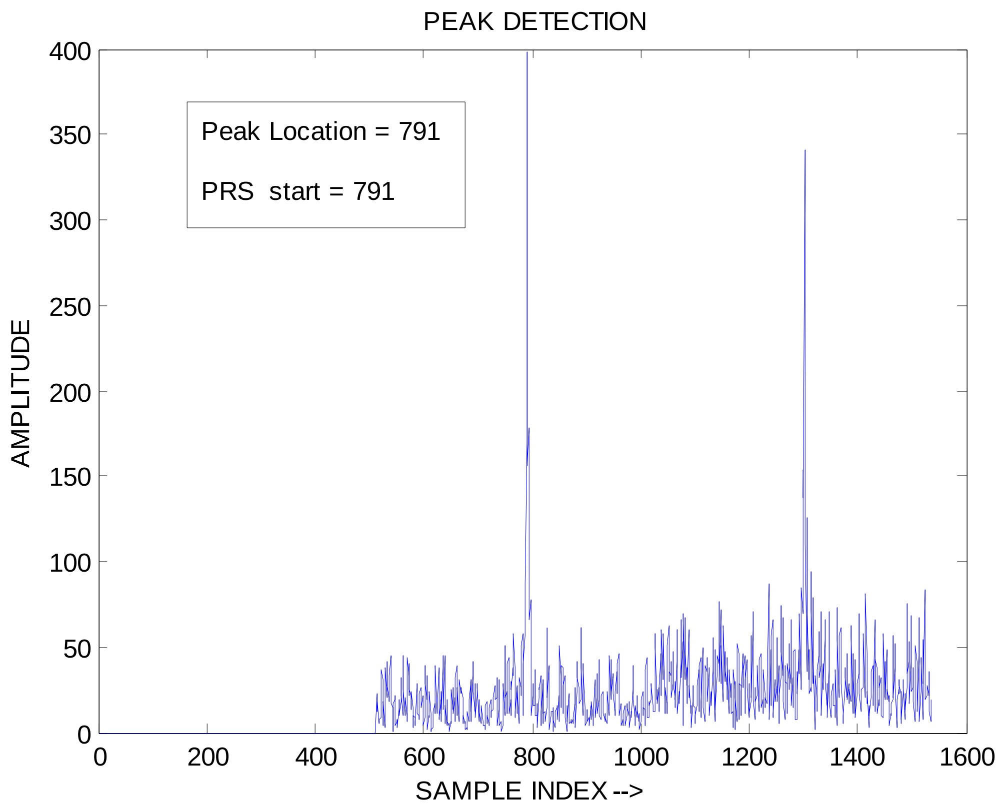

Figure 3. 23: Desired peak detection.

From the above figure it can be evaluated that the highest peak is located at sample index 791. From section 3.8.3 and 3.9 it is well known that the first symbol in the DAB frame is a Null symbol of size 664 zeros followed by a guard interval of 126 samples of PRS. Thus sum of null symbol and guard interval samples equals 790, therefore the peak is located exactly at the starting point of useful phase reference symbol. The useful OFDM symbol duration T_U (refer to Table 2.2) does not include the guard time interval.

3.12.2 Coarse time synchronization

It gives the rough frame timing by envelope detection of the received signal i.e., detecting the null symbol by comparing average signal power during null symbol period T_{NULL} with a set threshold.

From the received signal, a data block of size 664 (equal to T_{NULL}) samples is taken to measure the average signal power [4]. When this average signal power is less than half of the average transmitted signal power the null symbol has been detected, which indicates the start of the new frame. This method of frame synchronization based on null symbol detection is not suited for low SNR conditions because high noise power will provide incorrect frame timing estimate. Therefore phase reference symbol detection is ideally well suited for correct symbol timing and frame timing. Since according to DAB standard [1] the phase reference symbol occurs once in each transmission frame .

3.13 OFDM Symbol Demodulator

The OFDM symbol demodulator block demodulates the OFDM symbols from the synchronized DAB frame obtained from section 3.12. The demodulation is done by removing the null symbol and the phase reference symbol. The details of the OFDM symbol demodulator is discussed in the following sub-sections.

3.13.1 Guard time removal

This sub-block performs the reverse operation of the cyclic prefix done at the transmitter. It removes the guard interval from each OFDM symbols. Thus the output of this sub-block is the useful OFDM symbol which is given input to the FFT block.

3.13.2 FFT operation

After guard time removal the OFDM symbols has a size of 512 which is equal to the FFT length specified for mode-II in the DAB standard [1]. This sub-block performs the FFT operation on each OFDM symbol obtained from section 3.13.1. Thus we get back frequency domain samples after FFT operation.

3.13.3 Zero padding removal

This sub-block removes the 128 zeros that was padded to each D-QPSK symbol block in the transmitter and rearranges the data in proper form as illustrated in the Figure 3.24.

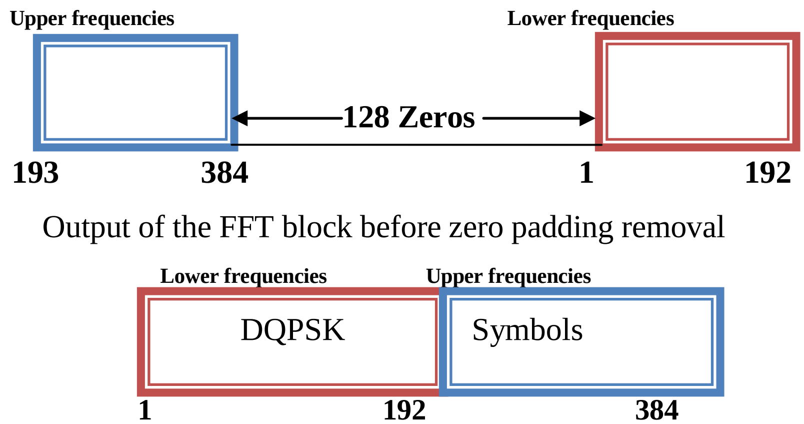

Figure 3. 24: Zero padding removal and data rearrangement.

3.14 Differential Demodulation

Differential demodulation of the carriers is performed by applying complex multiplication by the complex conjugated amplitude of the received D-QPSK symbol blocks from OFDM symbol demodulator. This is explained in the following equation [4]:

$$y_{l,k} = z_{l,k} \times z_{l-1,k}^{*} \tag{3.14}$$

Initialization of this process is done using received phase reference symbol. ‘y’ is the output of this block, ‘z’ is the received D-QPSK symbol blocks from OFDM symbol demodulator and ‘z*’ is the complex conjugate of the differential phase reference.

3.15 Data De-mapping

This block transforms the received QPSK symbols after differential demodulation back into bits. The consists of two sub-blocks named Frequency de-interleaving and QPSK de-mapping. The details of these are provided in the following sub-sections.

3.15.1 Frequency de-interleaving

Frequency interleaving was performed at the transmitter to nullify transmission disturbances such as selective fade. This sub-block performs the reverse of the frequency interleaving by re-arrangement of the bits to get QPSK symbol block identical with the output of QPSK mapping.

3.15.2 QPSK de-mapping

This sub-block converts the received complex QPSK symbols from the frequency de-interleaver output into bits. When the sign of the real part of the complex QPSK symbol is negative, the decoded bit is '1' and when it is positive the decoded bit is '0'. The same rule is applicable to imaginary part of the complex QPSK symbol [4]. The decoded bits should be arranged in the similar manner in which bit was used for QPSK mapping. The decoded I phase component bit will be assigned to index 1 to 384 and Q phase component bits to index 385 to 768.

3.16 Viterbi Decoder

To minimize the transmission errors 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 1/4 for channel coding [1]. This mother code was punctured with different puncturing vectors (refer to Appendix A) to obtain a wide range of possible code rates to adapt to the channel characteristics.

For decoding these codes the Viterbi algorithm will be used [26], which offers best performance according to the maximum likelihood criteria. The input to the Viterbi decoder will be hard-decided bits that are '0' or '1', which is referred to as a hard decision. No MATLAB code has been written for this block instead the function 'vitdec' is used for Viterbi decoding purpose.

3.16.1 De-puncturing

For increasing the code rate of the convolutional code from 1/4 to 1/3 for FIC data, a fixed punctured scheme was applied. The whole MSC data was punctured with the same puncturing vector and also according to protection level 1 as discussed in section 3.4.1. This last sub-block of the simulation model performs the process of de-puncturing to get back the original information transmitted.

3.16.2 Concatenated decoding

As discussed in section 3.4.2, the concept of concatenated coding has been applied for improving the BER performance of the DAB system. This was accomplished by using linear block code, cyclic code, hamming code and BCH (Bose-Chaudhuri-Hocquenghem) code as the outer encoder.

This sub-section performs concatenated decoding using the MATLAB function 'decode' for linear block code, cyclic code and hamming code and 'bchdec' for BCH code [15].

3.17 Conclusion

The chapter discussed about the details of transmitter and receiver side of the DAB system model for transmission mode-II implemented in MATLAB environment. The simulation results and performance analysis has been presented in next chapter.

Chapter 4 SIMULATION RESULTS AND DISCUSSION

4.1 Introduction

This chapter presents simulation results along with the bit error rate (BER) analysis for AWGN channel, Rayleigh fading channel and Rice channel. The simulation parameters are taken as per the DAB standard [1] for transmission mode-II. These parameters have been presented in Table 2.2.

The BER performance of uncoded DAB system is compared with the FEC coded DAB system. BER performance of DAB system with & without frequency interleaving and with and without puncturing is also analyzed. Before simulating the DAB system some basic simulations had been conducted for BPSK, QPSK and QAM modulation in AWGN & Rayleigh fading channels.

All simulations were performed in MATLAB 7.6.0 (R2008a).

4.2 Basic Simulation results

Before simulating the main DAB system in transmission mode-II, some basic simulations were performed including BER performance for BPSK, QPSK and QAM in AWGN and Rayleigh fading channel.

The simulation parameters [9] are presented below:

No. of random bits generated : 230400

SNR : -3 to 30 dB in steps of 1dB

Theoretical BER for BPSK & QPSK in AWGN : $\frac{1}{2}erfc(\sqrt{E_b/N_o})$ (4.1)

Theoretical BER for BPSK & QPSK in Rayleigh : $\frac{1}{2}\left[1-\frac{1}{\sqrt{1+\frac{1}{E_b/N_o}}}\right]$ (4.2)

Theoretical BER for 16-QAM in AWGN : $\frac{3}{8}erfc\left(\sqrt{\frac{2}{5}E_b/N_o}\right)-\frac{9}{64}erfc^2\left(\sqrt{\frac{2}{5}E_b/N_o}\right)$ (4.3)

Theoretical BER for 16-QAM in Rayleigh channel : $\frac{3}{8}\left[1-\frac{1}{\sqrt{1+\frac{5}{(2E_b/N_o)}}}\right]$ (4.4)

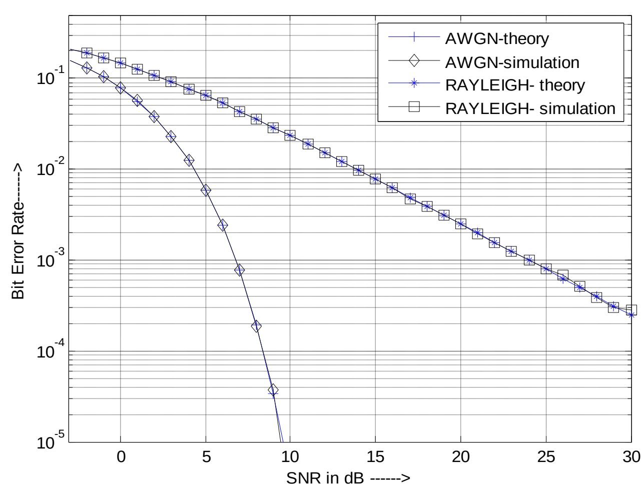

Figure 4. 1: BER performance for BPSK modulation in AWGN & Rayleigh fading channel.

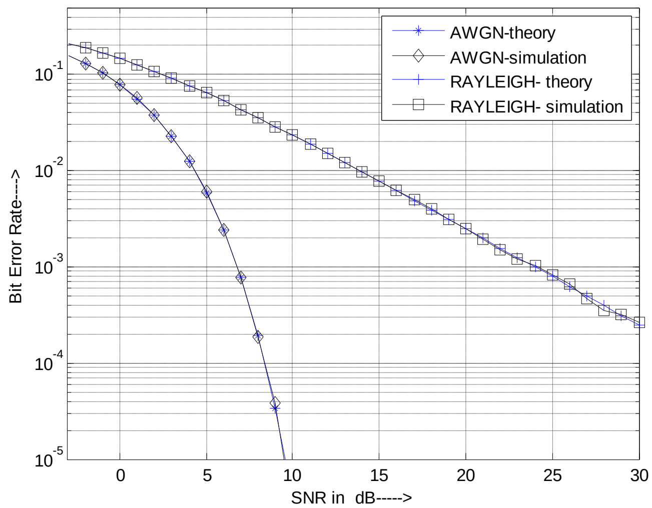

Figure 4. 2: BER performance for QPSK modulation in AWGN & Rayleigh fading channel.

As can be seen from Figure 4.1 and 4.2 that both theoretical and experimental BER are in good agreement with each other, respectively for BPSK and QPSK modulation in AWGN and Rayleigh fading channel.

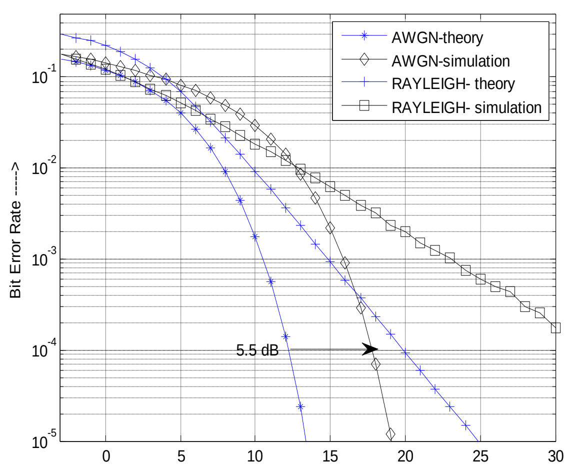

Figure 4.3: BER performance for 16-QAM modulation modulation in AWGN & Rayleigh fading channel.

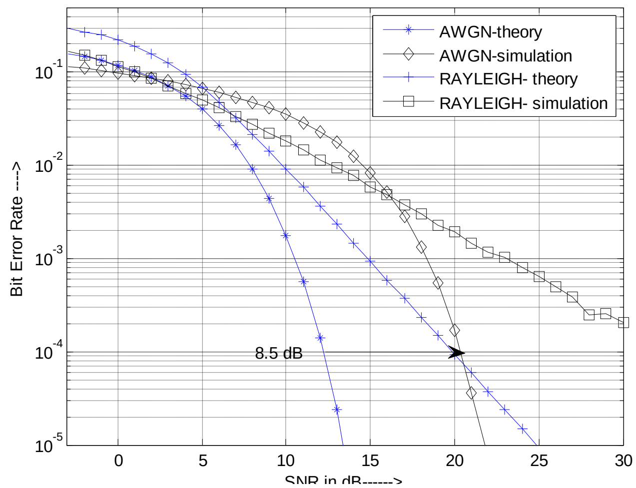

Figure 4. 4: BER performance of 32-QAM in AWGN & Rayleigh fading channel.

Figure 4.3 shows that for 16-QAM modulation in AWGN channel, experimental BER needs an additional transmitted signal power of 5.5 dB compared to theoretical BER to achieve a BER of 10^{-4}. And in rayleigh channel to achieve a BER of 10^{-4} experimental BER needs an additional transmitted signal power of 13 dB compared to theoretical BER. Figure 4.4 reveals that for 32-QAM modulation in AWGN channel to achieve a BER of 10^{-4} experimental BER needs an additional transmitted signal power of 8.5 dB compared to theoretical BER. In rayleigh channel to achieve a BER of 10^{-4} experimental BER needs an additional transmitted signal power of 16 dB compared to theoretical BER. It is also conluded that as SNR increases, bit error rate reduces and the use of channel coding could improve the BER performance.

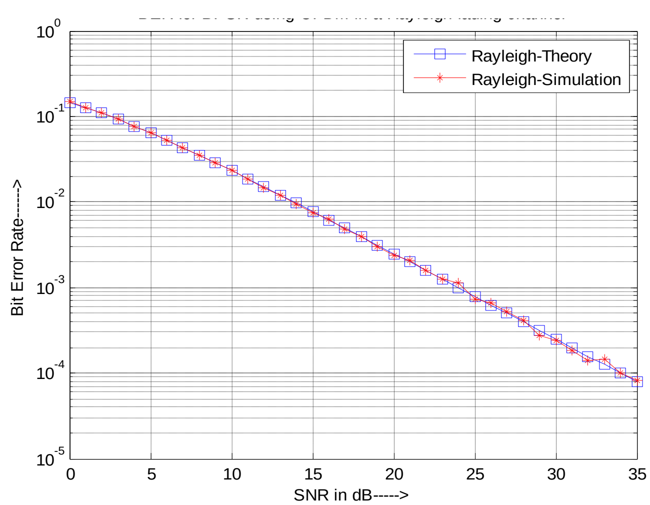

Figure 4. 5: BER performance of OFDM using BPSK modulation in Rayleigh fading channel.

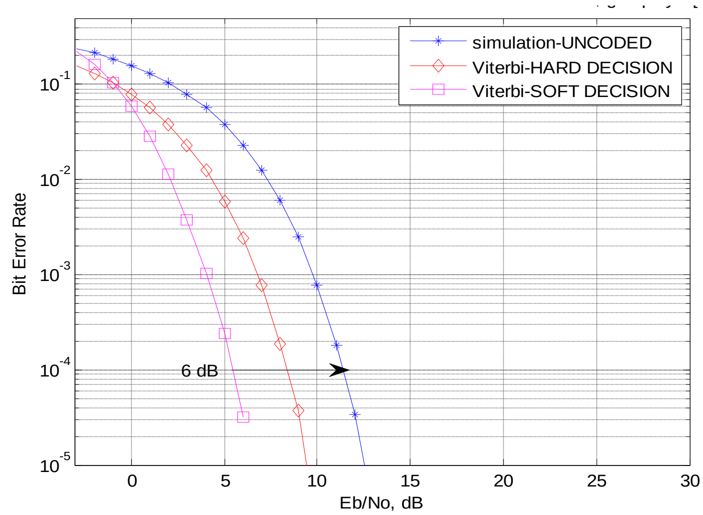

Figure 4. 6: BER performance for BPSK modulation in AWGN channel using convolutional code.

Figure 4.5 show that both theoretical and simulated results show similar performance for OFDM using BPSK modulation. The parameters for the result shown in Figure 4.5 were taken to be as:

FFT size	:64
No. of sub-carriers	:52
No. of random bits generated	:230400
SNR	:0 to 35 dB in steps of 1dB

Figure 4.6 shows that to achieve a BER of 10^{-4}, the coded BPSK with Viterbi 3-bit soft decision decoding gives a coding gain of 2.5 dB and 6 dB compared with Viterbi hard decision decoding & uncoded BPSK modulation, respectively. Convolutional code with constraint length L=3, code rate 1/2 and generator polynomial in octal (7, 5) was used as parameters for this simulation.

4.3 Simulation Results for DAB mode-II in AWGN channel

After performing basic simulations for BER performance of BPSK and QPSK modulation in AWGN and Rayleigh fading channel, now the simulation results for DAB mode-II in AWGN channel will be presented. First of all the correctness of the DAB simulation model given in Figure 3.1 will be tested. The simulation parameters have been taken as per the DAB standard [1] for the mode-II.

The probability of bit error for π/4 D-QPSK is given by the following expression [7]:

Theoretical BER for π/4 D-QPSK in AWGN: $\frac{1}{2} erfc(\sqrt{0.5858 \times (E_b/N_o})$ (4.5)

Figure 4.7 presents the system performance.

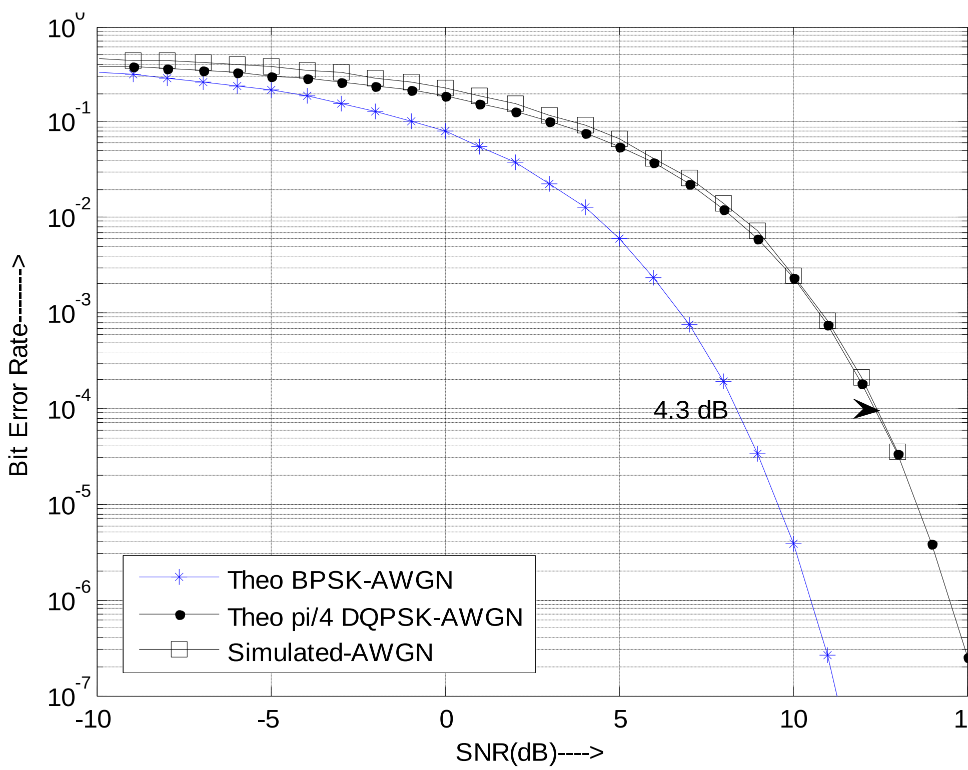

Figure 4. 7: BER performance of DAB mode-II in AWGN channel.

It can be from Figure 4.7 that both experimental and theoretical BER plots are same and almost overlapping each other. This justifies that the DAB system model simulated is perfectly implemented. The result also indicates that to achieve a BER of 10^{-4} theoretical $\pi/4$ D-QPSK needs an additional SNR of 4.3 dB compared to theoretical BPSK.

After verifying the correctness of the DAB system model, next the fine time synchronization under worst SNR of -11 dB will be tested. Figure 4.8 presents the peak detection with SNR = -11 dB.

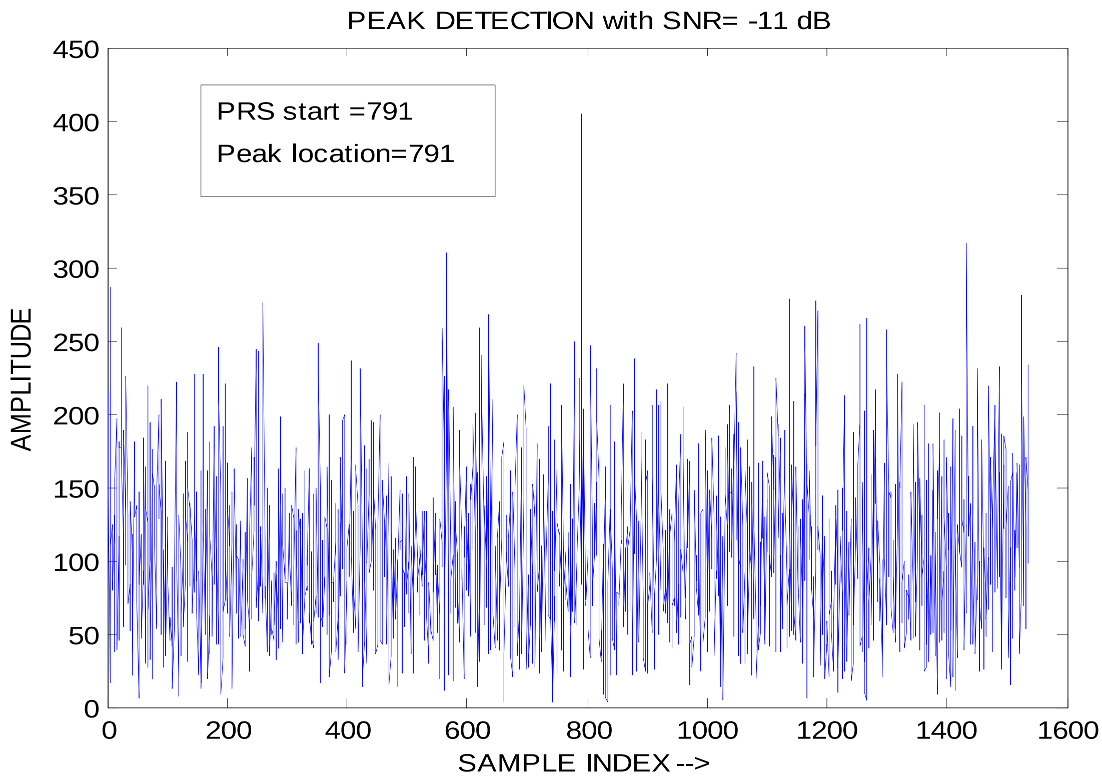

Figure 4. 8: Peak detection for fine time synchronization at very low SNR.

From the above Figure 4.8 it can be evaluated that the highest peak is located at sample index 791 even at worst SNR at -11 dB providing correct fine time synchronization. From chapter 3 it is well known that the first symbol in the DAB frame is a Null symbol of size 664 zeros followed by a guard interval of 126 samples of PRS. Thus sum of null symbol and guard interval samples equals 790, therefore the peak is located exactly at the starting point of useful phase reference symbol.

The performance of DAB system with FEC coding will be analyzed next. Convolutional code with constraint length 7 and generator polynomials 133, 171, 145 and 133 was taken as simulation parameters. No puncturing has been applied. Decoding was done with Viterbi algorithm. Figure 4.9 presents the result for the DAB system with FEC coding.

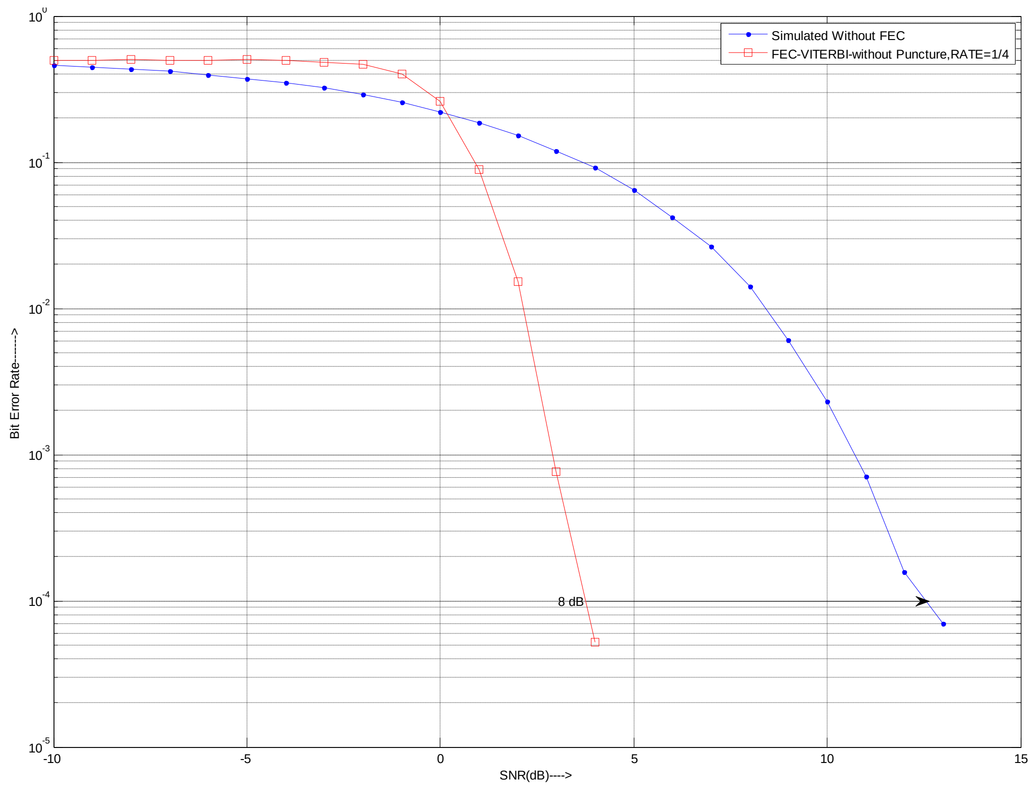

Figure 4. 9: BER performance with and without FEC coding.

From the Figure 4.9 it can be seen that the use of the channel coding improves the BER performance of the DAB system. It can be evaluated from above figure that to achieve a BER of 10^{-4} coded DAB system without puncturing gives a coding gain of approximately 8 dB compared with the uncoded system.

After verifying improvement in the BER performance using channel coding, the performance with and without frequency interleaving will be investigated next. The performance results are presented in Figure 4.10.

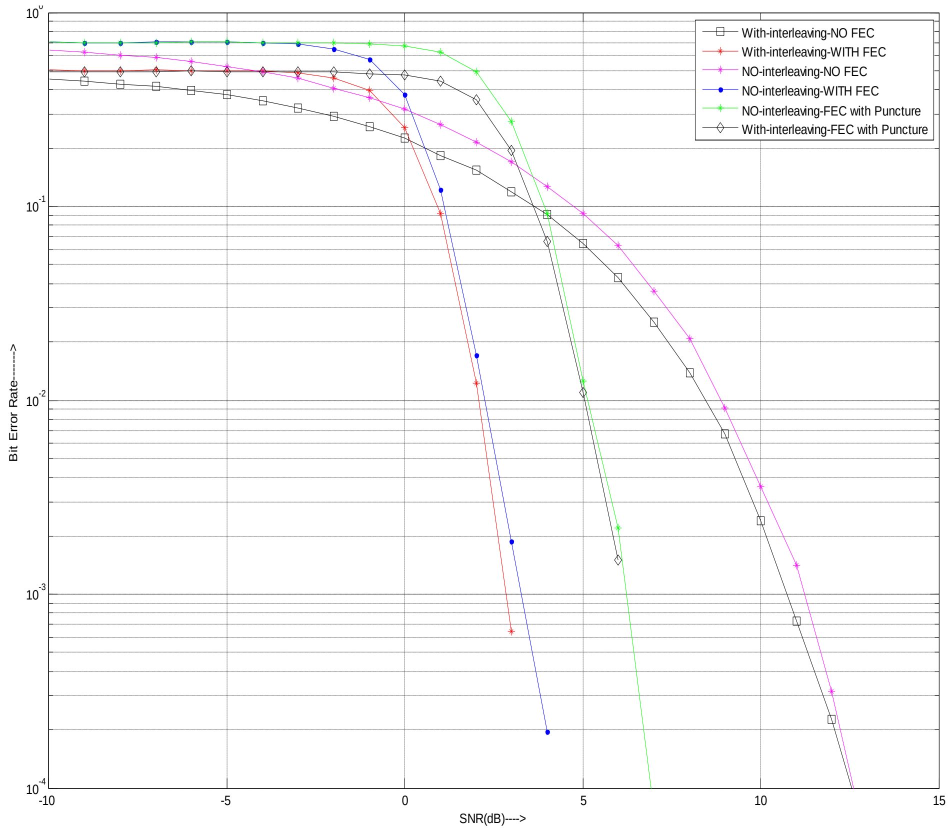

Figure 4. 10: BER performance with and without FEC coding & with and without interleaving.

Figure 4.10 clearly shows that interleaving is necessary for the channel coding to function properly. Also this offsets any deep fades that may occur in the wireless channel. Interleaving spreads the data bits over the sub-carriers. The wireless channel as a wideband channel and rarely encounters a flat, consistent response across the entire spectrum. As deep fades affect more than one sub-carrier channel, a block data is affected. By spreading adjacent bit across the channels, the bits have been re-arranged in their proper order. The effect of fade is reduced.

The performance result for different coding rates is presented next. A particular coding rate is implemented by a corresponding puncturing vector (refer to Appendix A) defined in the DAB standard [1]. Figure 4.11 presents result for DAB mode-II applied with different coding lengths 8/11, 8/12, 8/16, 8/24 and 8/32.

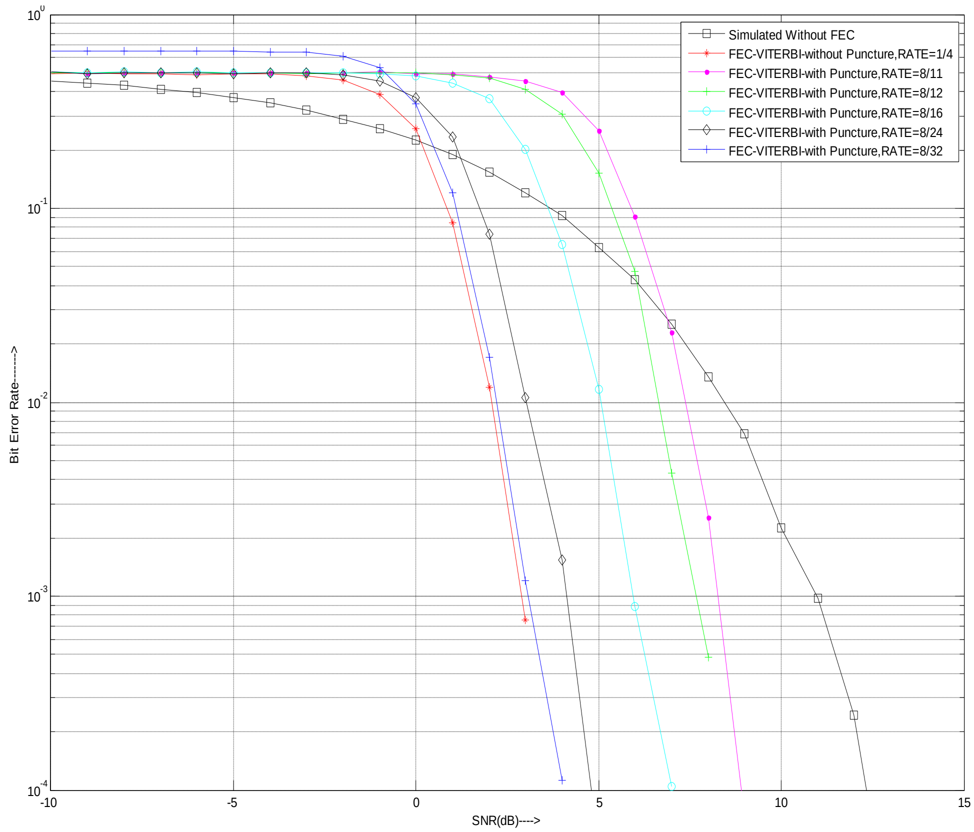

Figure 4. 11: BER performance with different coding rates in AWGN channel.

Figure 4.11 reveals that as we increase the coding rate (or transmitting less redundancy) we need more transmitted signal power to get a better BER performance. But at the same time flexibility of the system also increases and good high rate convolutional codes are generated from low rate mother code. It may also be observed that FEC with puncturing is required for improving system performance in different transmission channels and can be decoded by the basic Viterbi decoder without increase in decoder complexity.

The performance of DAB system using concatenated coding technique was investigated next. For this outer coding employed Block coding such as Linear, Cyclic and Hamming code and inner coding was accomplished by convolutional codes. Codeword length was taken as 511 and Message length to be 502. The performance results are presented in Figure 4.12.

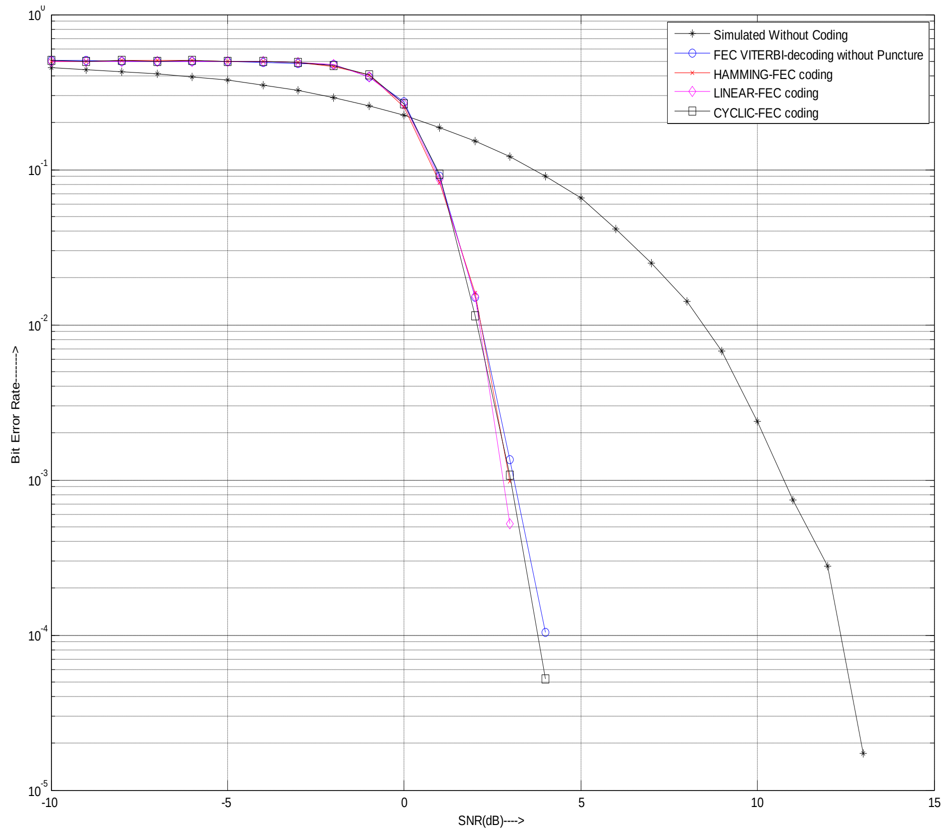

Figure 4. 12: BER performance using concatenated coding in AWGN channel.

It can be seen from Figure 4.12 that concatenated coding (employing outer block coding and inner convolutional coding) improves the BER performance marginally compared with only convolutional coding. It may be observed that use of linear block coding as outer code performs well compared with cyclic and hamming code giving a coding gain of about 0.5 dB.

After analysing the BER performance using block coding techniques, the performance analysis using BCH coding as the outer coding and convolutional coding as the inner coding. Codeword length was taken as 511 and Message length to be 493 for error correcting capability of two and 511 and 439 for error correcting capability of eight as simulation parameters. Figure 4.13 and 4.14 presents the results for BCH coding.

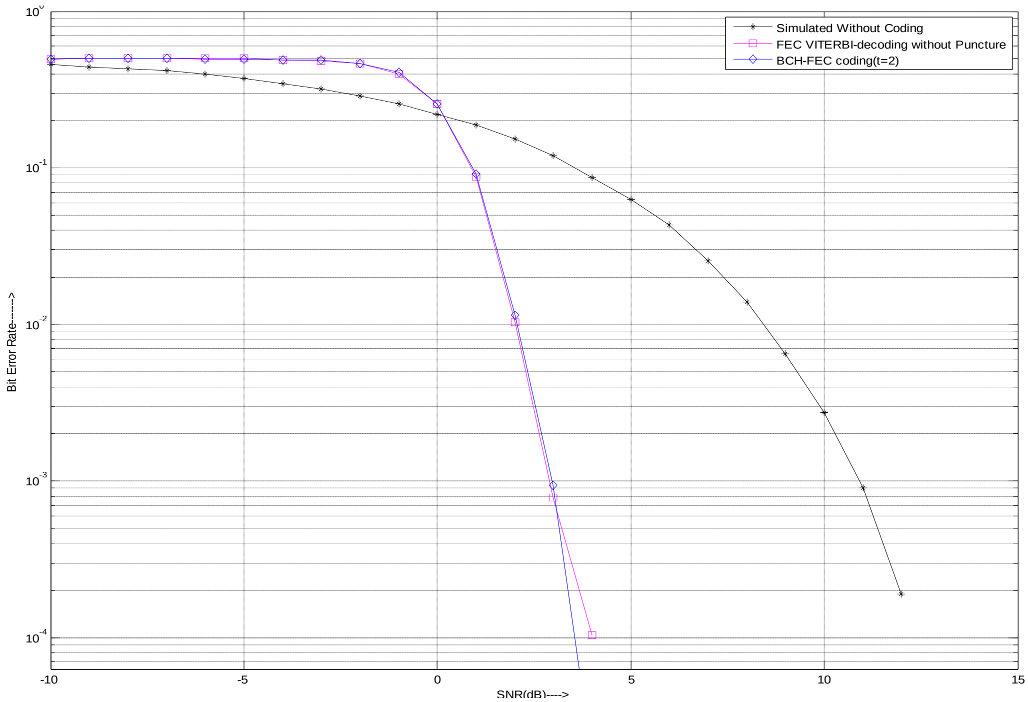

Figure 4. 13: BER performance using concatenated coding in AWGN channel with t=2.

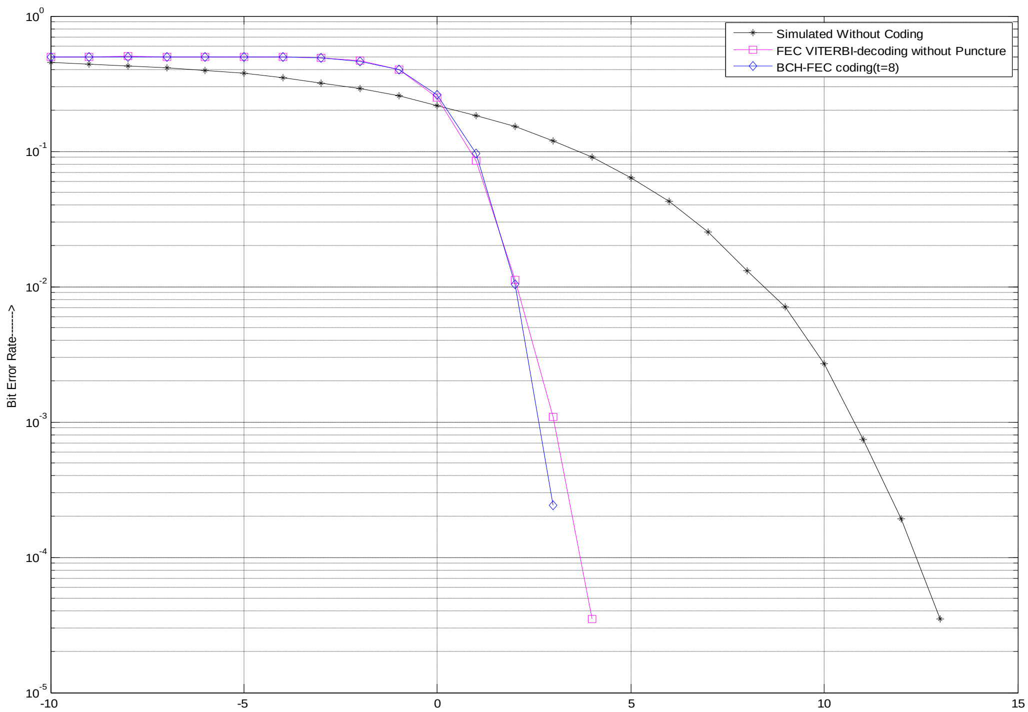

Figure 4. 14: BER performance using concatenated coding in AWGN channel with t=8.

From Figure 4.13 and 4.14 presents that concatenated coding (employing outer BCH coding and inner convolutional coding) improves the BER performance marginally compared with only convolutional coding. Here it is seen that for BCH using error correction capability from 2 to7 start to show improvement at BER of 10^{-4}. But for t=8 a coding gain of about 0.5 dB is observed form Figure 4.14 compared to FEC coding.

The BER performance of the channel coding using hard and 4-bit soft decision Viterbi decoding is investigated next. No puncturing has been applied. Convolutional coding with mother code rate ¼ was used. The 15 Quantization levels were taken to be as [0.1000, 0.1179, 0.1389, 0.1638, 0.1931, 0.2276, 0.2683, 0.3162, 0.3728, 0.4394, 0.5179, 0.6105, 0.7197, 0.8483, and 1.0000].

Figure 4.15 presents the performance results for hard-soft Viterbi decoding.

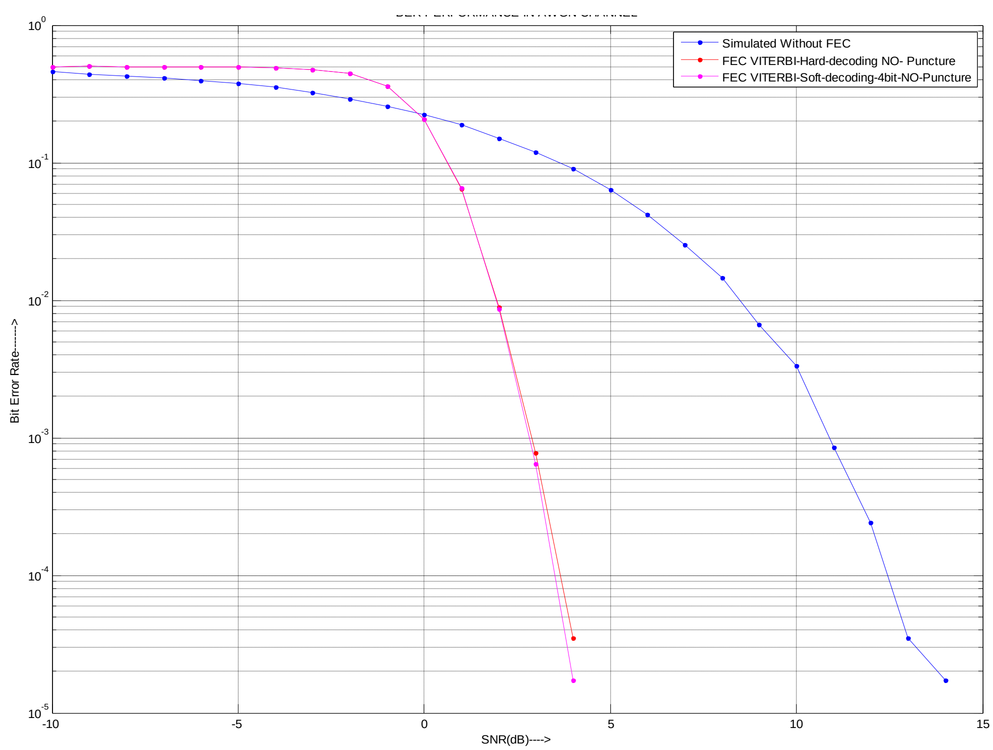

Figure 4. 15: BER performance using hard-soft Viterbi decoding in AWGN channel.

Figure 4.15 indicates that Viterbi 4-bit soft decision decoding gives a slightly better BER performance of about 0.2 dB compared with hard decision decoding. No puncturing has been applied. Important parameter for soft decision decoding is the Quantization levels. It is seen that for different Quantization levels the performance vary greatly. It is finally concluded that use of adaptive Quantization levels would give a much better performance.

The DAB standard [1] defines five protection levels for encoding of the MSC but coding of FIC is fixed. Protection level 1 is considered to be most efficient among other profiles in case of mobile reception. Therefore performance of DAB system using Protection level 1 is investigated next. The performance results are presented in Figure 4.16.

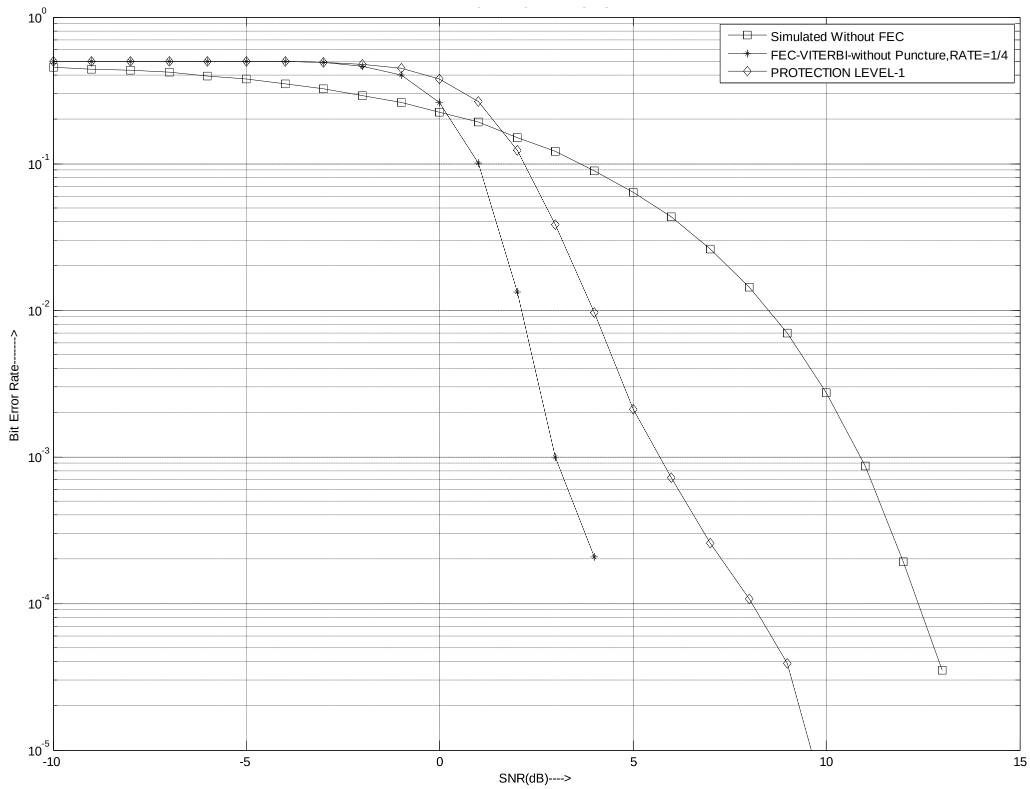

Figure 4. 16: BER performance using Protection level 1 in AWGN channel.

Figure 4.16 shows that channel coding of MSC with Protection level 1 makes the system resistant to channel impairments but at the cost of higher SNR value. The actual performance of Protection level 1 will be investigated in a fading channel in next section.

4.4 Simulation Results for DAB mode-II in Rayleigh fading channel.

After investigating the BER performance of DAB mode-II in AWGN channel, the performance analysis in Rayleigh fading channel will be considered. The simulation parameters have been taken as per the DAB standard [1].

First the performance of the system using different coding rates was considered. Figure 4.17 presents result for DAB mode-II applied with different coding lengths 8/11, 8/12, 8/16, 8/24 and 8/32.

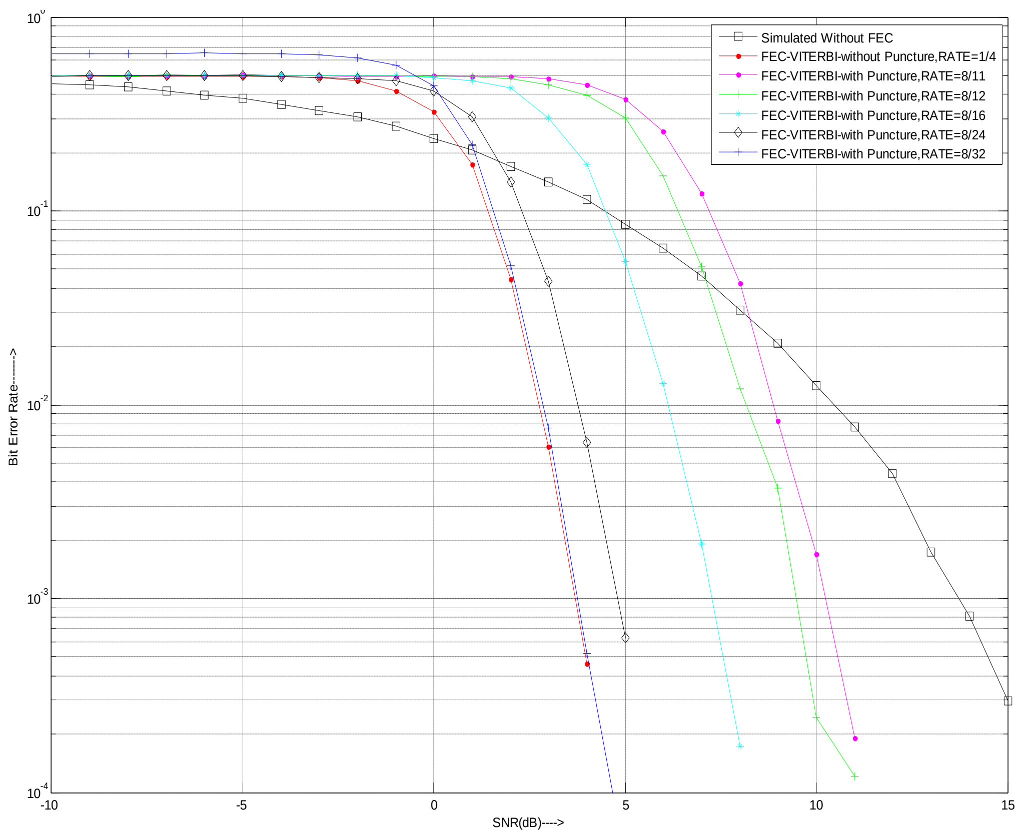

Figure 4. 17: BER performance with different coding rates in a fading channel.

Figure 4.17 reveals that as we increase the coding rate (or transmitting less redundancy) we need more transmitted signal power (compared to AWGN case) to get a good BER performance. For

FEC punctured with puncturing pattern PI=16 (code rate 8/24) needs a SNR of 5 dB in AWGN channel compared to SNR of 7 dB in Rayleigh fading channel to get a BER of 10^{-4}.
Similarly the performance of the system was investigated with and without frequency interleaving. Figure 4.18 presents the performance result.

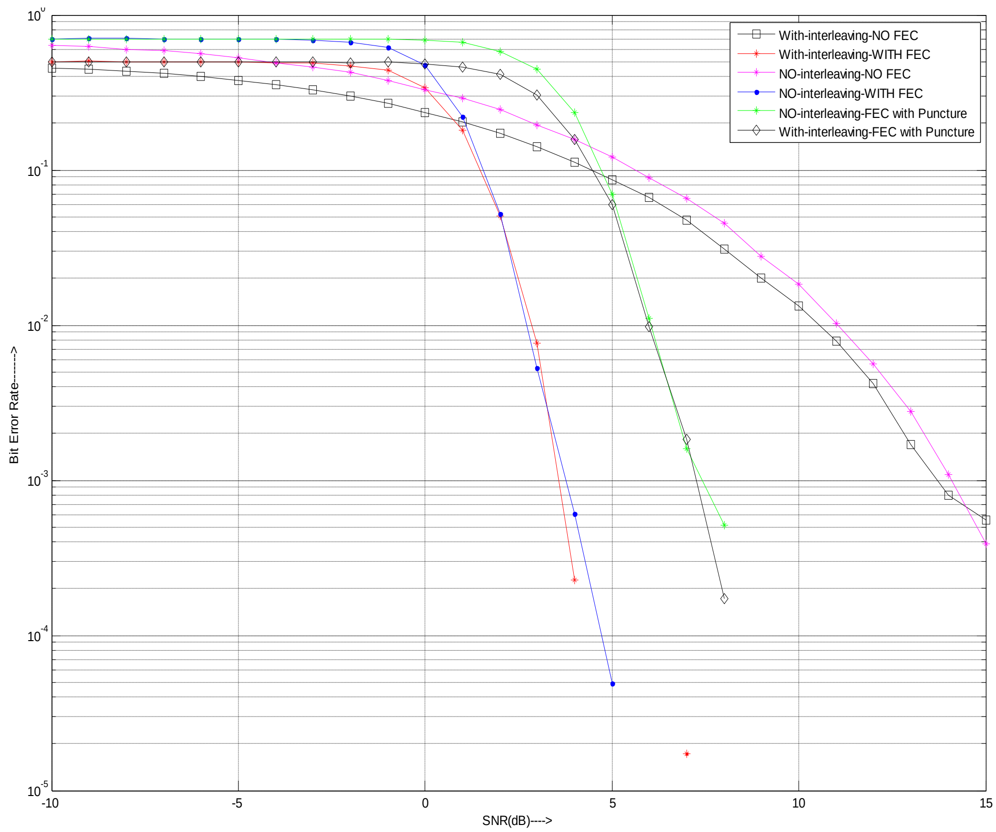

Figure 4. 18: BER performance with and without interleaving in a fading channel.

Figure 4.18 clearly shows that interleaving is necessary for the channel coding to function properly. Also this eliminates any deep fades that may occur in the wireless channel. Again in a fading channel it may be observed that the SNR required is high for a given BER performance in comparison to AWGN channel. It is seen that to achieve a given BER performance a higher SNR is required for without interleaving in comparison to with interleaving. A coding gain of 0.5 dB with interleaving is evaluated.

After investigating the performance using interleaving, the performance analysis with and without puncturing will be investigated next. For this the system was exposed to fading channel with Doppler frequency 20 Hz (i.e., v= 24 km/hr), 40 Hz (i.e., v= 48 km/hr) and 100 Hz (i.e., v= 120 km/hr) for a fixed transmission frequency, f=900 MHz. The performance result is presented in Figure 4.19, 4.20 and 4.21 respectively.

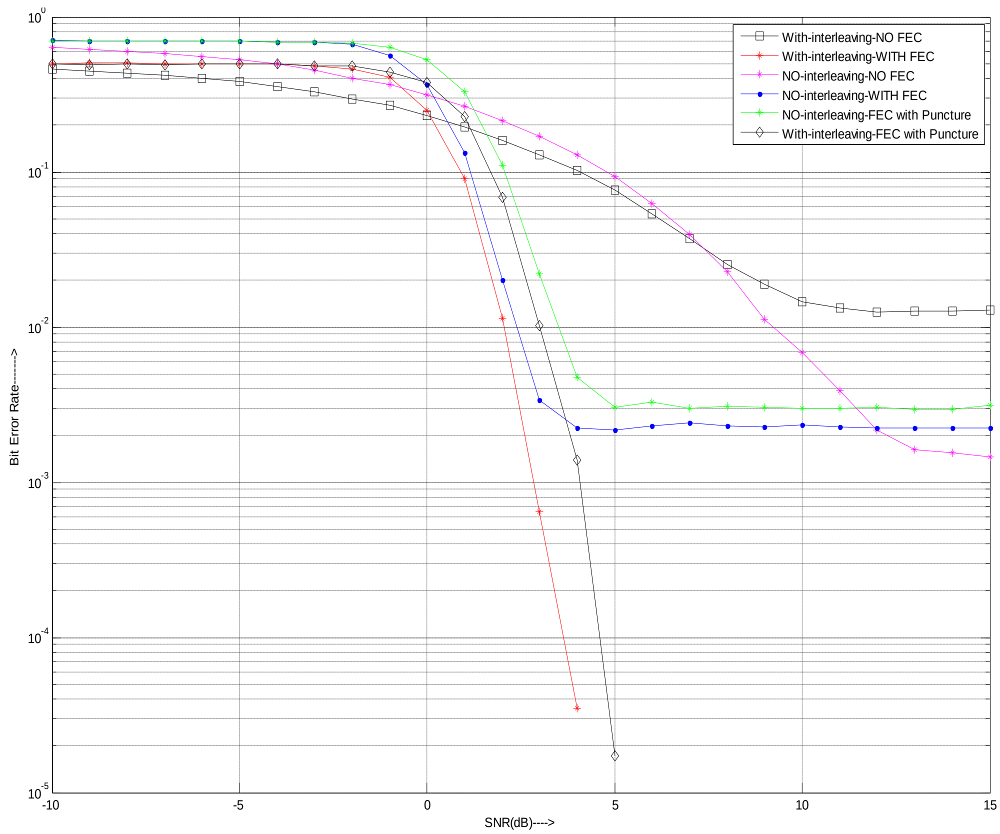

Figure 4. 19: BER performance with and without puncture in a fading channel with Doppler 20 Hz.

It may be observed that for flat- frequency Rayleigh channel with no Doppler shift in Figure 4.17 employing punctured FEC with puncturing pattern PI=16 (code rate 8/24) needs SNR of 7 dB to get a BER of 10^{-4}. The same system with Doppler shift of 20 Hz requires a SNR of 4.5 dB for the same BER performance. This shows that puncturing improves the system performance in different transmission channels.

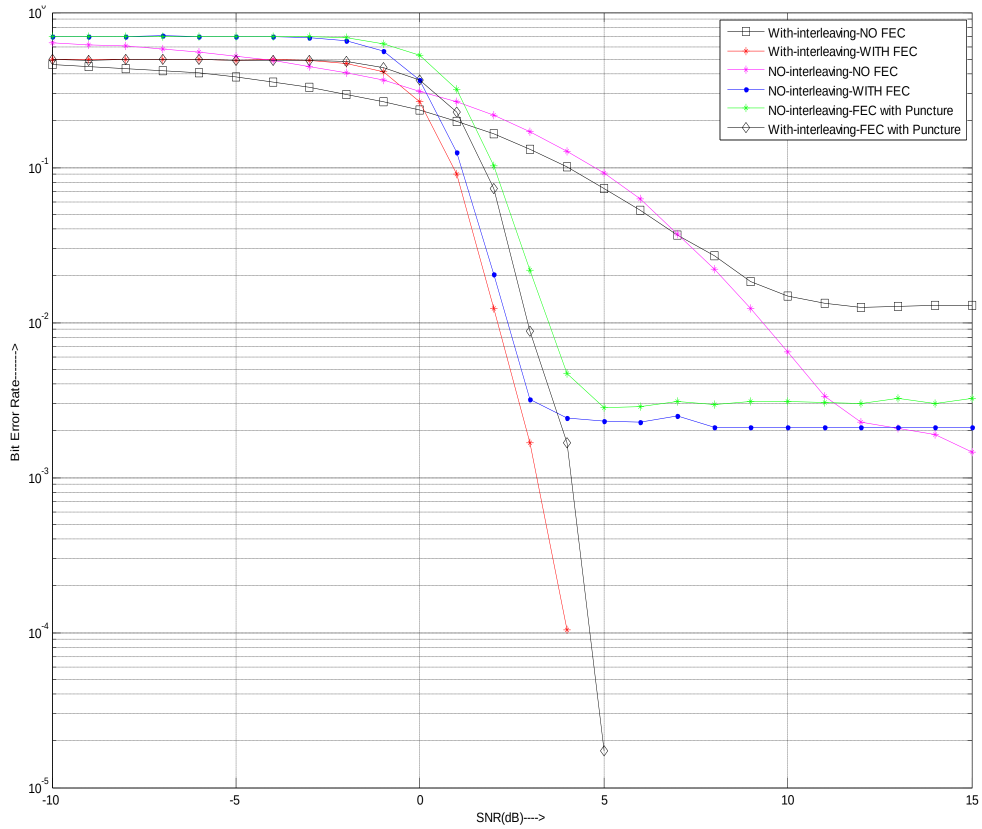

Figure 4. 20: BER performance with and without puncture in a fading channel with Doppler 40 Hz.

It may observed that with Doppler frequency of 40 Hz, FEC with puncture (code rate 8/24) does not show any increase in SNR for same BER performance of 10^{-4} compared to Doppler shift of 20 Hz.

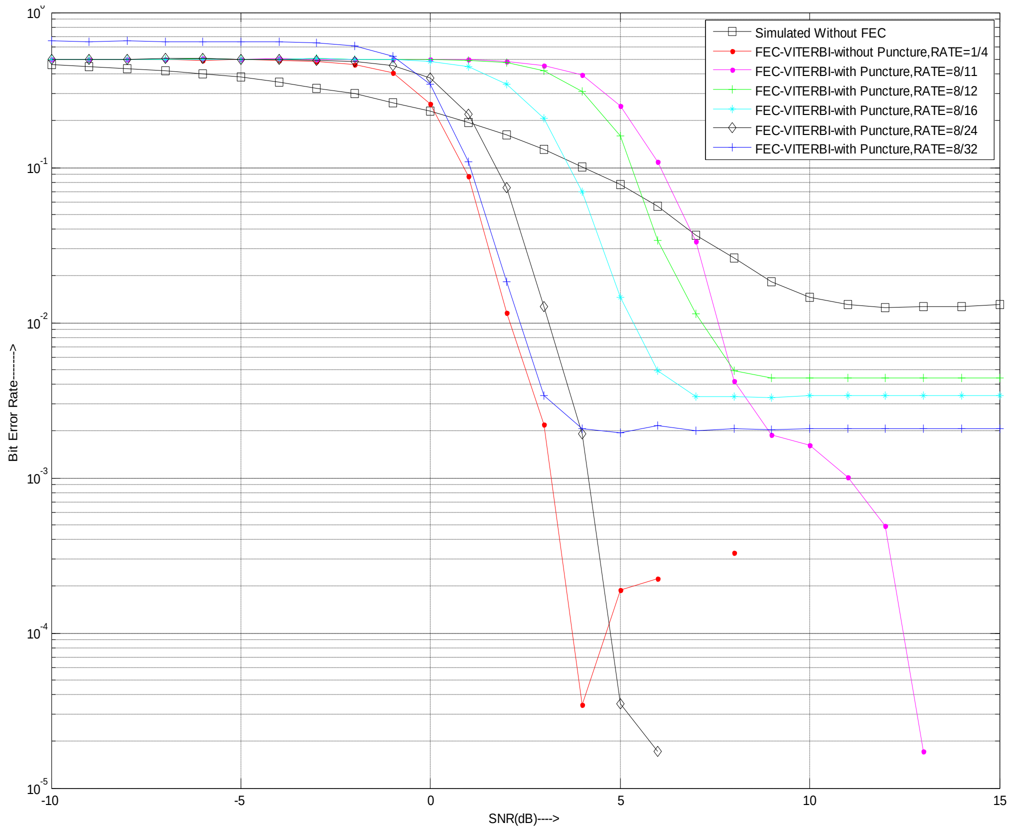

Figure 4. 21: BER performance with and without puncture in a fading channel with Doppler 100 Hz.

It is observed that the performance of FEC without puncturing deteriorates after SNR of 4 dB (no. of bit error increases) but FEC with puncture (code rate 8/24) does not show any increase in SNR for same BER performance of 10^{-4}. Thus FEC with puncturing is essential for better performance.

The effect of frequency selective (three path) Rayleigh fading channel with Doppler shift of 100 Hz on the BER performance will be investigated next. Figure 4.22 presents the performance result.

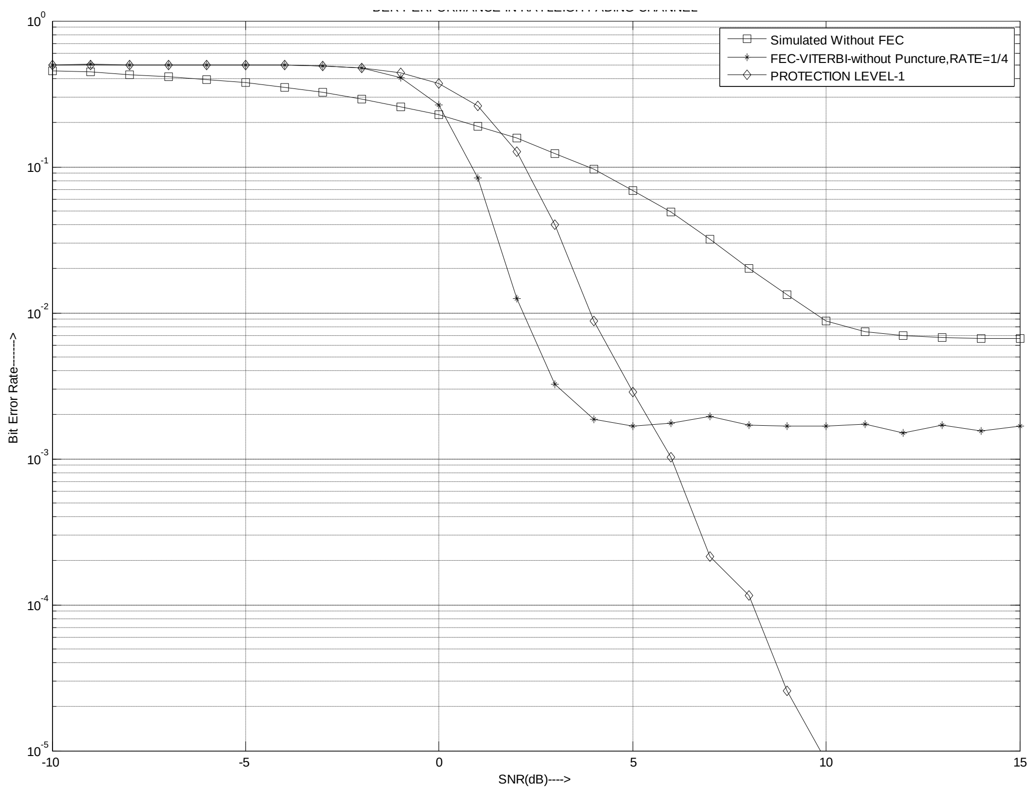

Figure 4. 22: BER performance with Protection level 1 in a fading channel.

In Figure 4.22 the system was exposed to frequency selective (three path) Rayleigh fading channel with Doppler shift of 100 Hz. It is observed that FEC without puncturing shows no improvement in BER performance after SNR of 4 dB but FEC with Protection level 1 needs a SNR of 8 dB for a BER performance of 10^{-4}. Therefore it indicates that protection level 1 is well suited for mobile reception.

The performance of the system with different block coding technique is presented next. Here concatenated coding technique is considered. For this outer coding employed Block coding such as Linear, Cyclic and Hamming code and inner coding was accomplished by convolutional codes. Codeword length was taken as 511 and Message length to be 502. The performance results are presented in Figure 4.23.

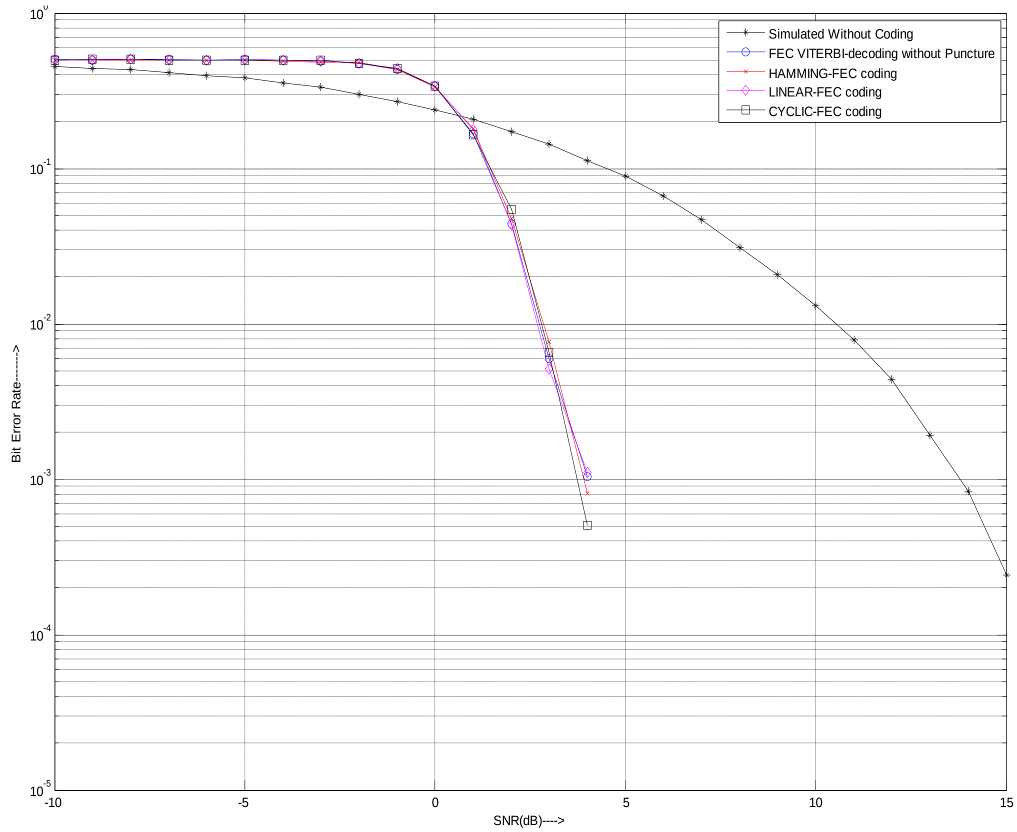

Figure 4. 23 BER performance with block coding in a fading channel.

It can be seen from Figure 4.23 that concatenated coding (employing outer block coding and inner convolutional coding) improves the BER performance marginally compared with only convolutional coding. It may be observed that use of cyclic block coding as outer code performs well compared with linear and hamming code giving a coding gain of about 0.5 dB.

After analysing the BER performance using block coding techniques, the performance analysis using BCH coding as the outer coding and convolutional coding as the inner coding. Codeword length was taken as 511 and Message length to be 439 for error correcting capability of eight as simulation parameters. Figure 4.24 presents the results for BCH coding.

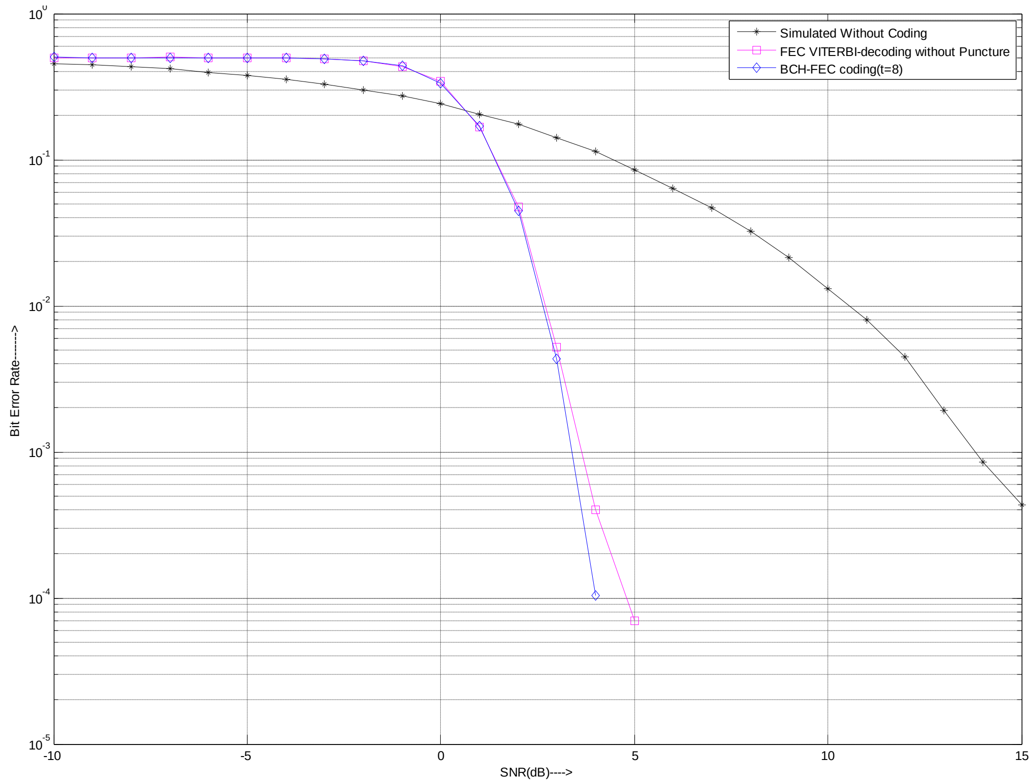

Figure 4. 24: BER performance with concatenated coding in a fading channel.

Figure 4.24 reveals that concatenated coding (employing outer BCH coding and inner convolutional coding) improves the BER performance marginally compared with only convolutional coding. It provides a coding gain of about 1 dB.

4.5 Simulation Results for DAB mode-II in Rician channel

After investigating the BER performance of DAB mode-II in Rayleigh fading channel, the performance analysis in Rician channel will be considered. The simulation parameters have been taken as per the DAB standard [1].

First the performance of the system using different coding rates was considered. Figure 4.25 presents result for DAB mode-II applied with different coding lengths 8/11, 8/12, 8/16, 8/24 and 8/32.

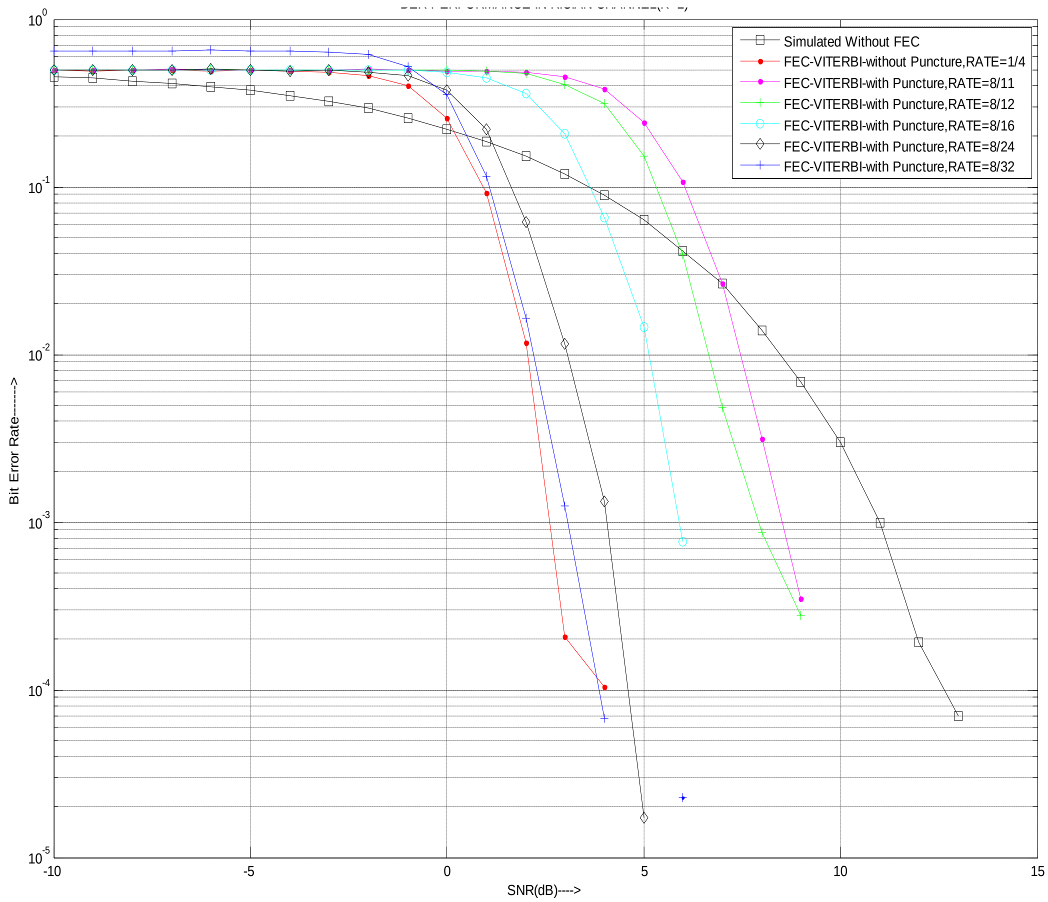

Figure 4. 25: BER performance with different coding rates in a Rician channel.

Figure 4.25 reveals that as we increase the coding rate (or transmitting less redundancy) we need more transmitted signal power to get a good BER performance. It may be seen that in Rice channel for FEC punctured with puncturing pattern PI=16 (code rate 8/24) provides a coding gain of 0.5 dB compared to AWGN channel and a coding gain of 2.5 dB compared to Rayleigh channel to get a BER of 10^{-4}.

Similarly the performance of the system was investigated with and without frequency interleaving. Figure 4.26 presents the performance result.

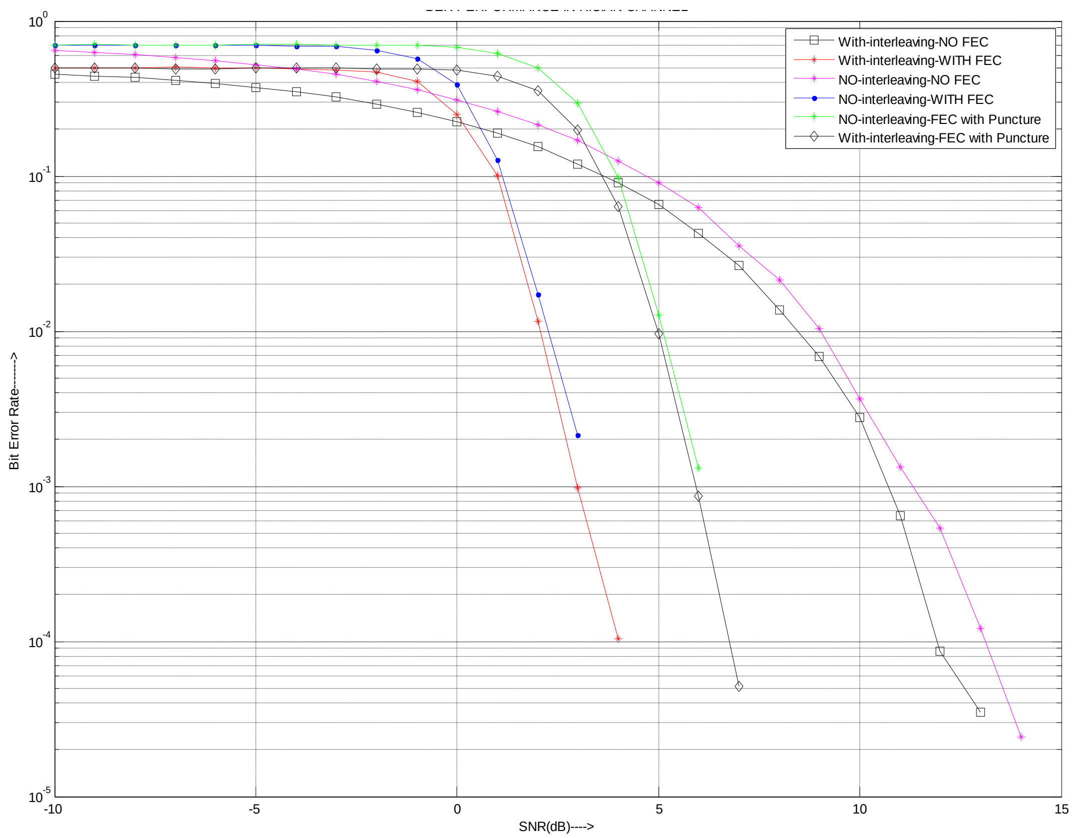

Figure 4. 26: BER performance with and without interleaving in a Rician channel.

Figure 4.26 clearly shows that interleaving is necessary for the channel coding to function properly. It is seen that to achieve a given BER performance a higher SNR is required for without interleaving in comparison to with interleaving. A coding gain of 0.5 dB with interleaving is evaluated.

The effect of flat- frequency Rician fading channel with Doppler shift of 100 Hz on the BER performance will be investigated next. Figure 4.27 presents the performance result.

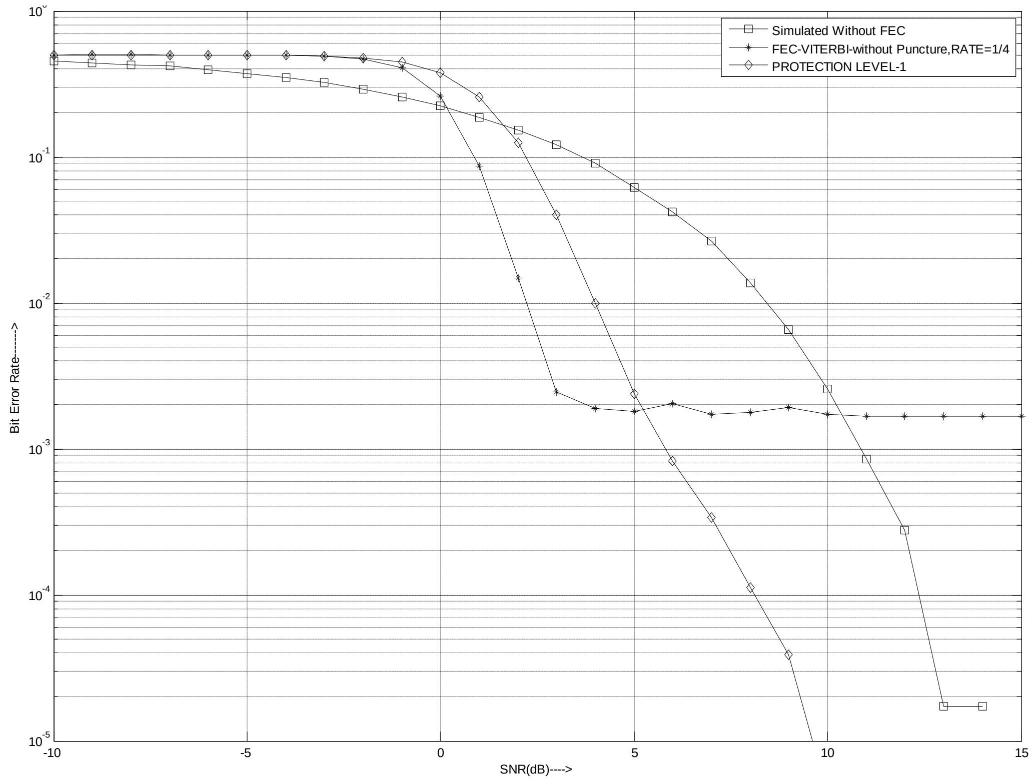

Figure 4. 27: BER performance with Protection level 1 in a Rician channel.

In Figure 4.27 the system was exposed flat- frequency static channel with Doppler shift of 100 Hz and k=1. It is observed that FEC without puncturing shows no improvement in BER performance after SNR of 3 dB but FEC with Protection level 1 needs a SNR of 8 dB for a BER performance of 10^{-4}. Therefore it indicates that protection level 1 provides robustness against channel impairments.

The performance of the system with different block coding technique is presented next. Here concatenated coding technique is considered. For this outer coding employed Block coding such as Linear, Cyclic and Hamming code and inner coding was accomplished by convolutional codes. Codeword length was taken as 511 and Message length to be 502. The performance results are presented in Figure 4.28.

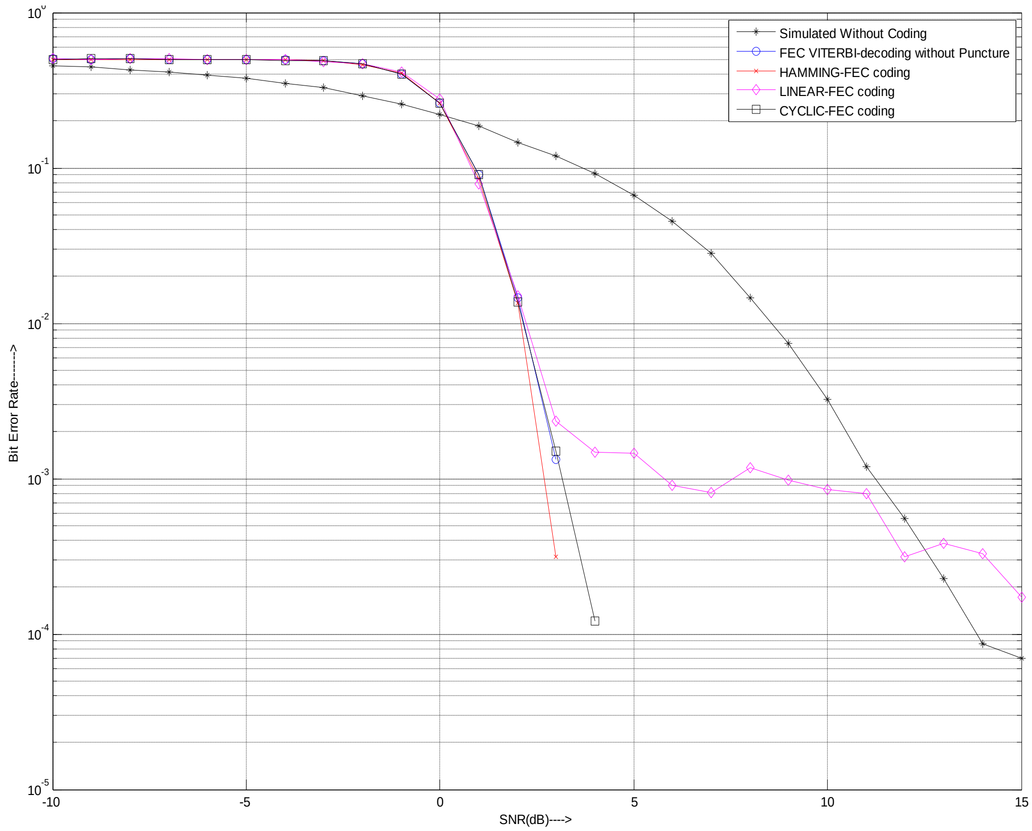

Figure 4. 28: BER performance with Block coding in a Rician channel.

It can be seen from Figure 4.28 that concatenated coding (employing outer block coding and inner convolutional coding) improves the BER performance marginally compared with only convolutional coding. It may be observed that use of hamming coding as outer code performs well compared with linear and cyclic code giving a coding gain of about 0.5 dB.

After analysing the BER performance using block coding techniques, the performance analysis using BCH coding as the outer coding and convolutional coding as the inner coding. Codeword length was taken as 511 and Message length to be 439 for error correcting capability of eight as simulation parameters. Figure 4.29 presents the results for BCH coding.

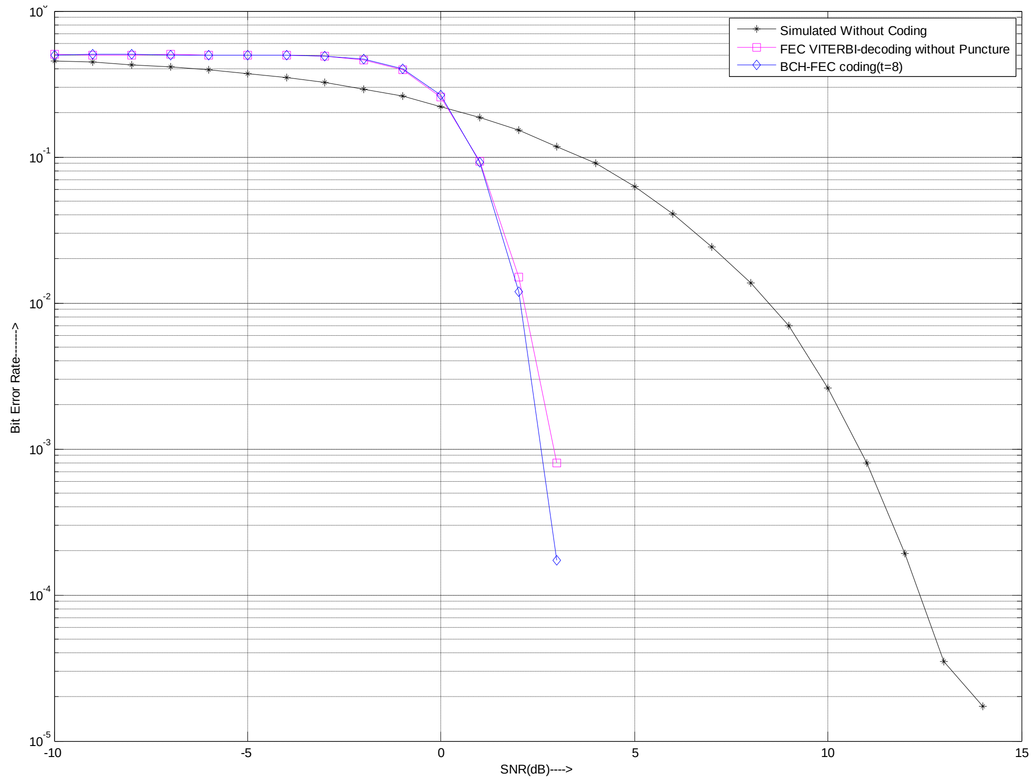

Figure 4. 29: BER performance with concatenated coding in a Rician channel.

Figure 4.29 indicates that concatenated coding (employing outer BCH coding and inner convolutional coding) improves the BER performance marginally compared with only convolutional coding. It provides a coding gain of about 0.5 dB.

4.6 Conclusion

The basic goal of this thesis has been achieved since the transmitted signal according to the DAB standard has been perfectly received at the receiver. The next chapter presents the final concluding remarks and scope of future work.

Chapter 5
CONCLUSION

5.1 Introduction

The basic goal of this thesis has been to develop, implement and evaluate BER performance of the DAB system in MATLAB environment. This thesis used previous work of [4] as reference for performance analysis of DAB under different coding techniques. This project developed a DAB base-band transmission system based on Eureka 147 specifications. The implementation was done in MATLAB environment. The simulation model included punctured convolutional encoder, frequency interleaver, DQPSK modulator, and OFDM transmitter (IFFT) in the Transmitter side and in the Receiver corresponding inverse operations have been carried out. The transmitted signal was exposed to AWGN, Rayleigh and Rician channel. BER was used as an performance criteria.

The Eureka 147 DAB system promises to be efficient radio broadcasting technology with significant advantages compared with the existing analog AM/FM radio. It will deliver high quality digital radio services, including wide range of value-added services to fixed, portable and mobile receivers. With the present trend of wireless technologies becoming digital, DAB will become increasingly important in the future radio broadcasting.

5.2 Performance Analysis

According to simulations, DAB appears to be suitable radio broadcasting technology for high performance in diverse transmission channels. From the simulation results it is observed that FEC is practically well suited for channel coding giving a coding gain of about 8 dB in AWGN channel, 12 dB in Rayleigh fading channel and 8.2 dB in Rician channel. The results reveal that interleaving is essential for the FEC to work properly and FEC with punctured technique increases the flexibility of the system in different communication channels.

From the results it is observed that use of concatenated channel coding (employing outer block coding and inner convolutional coding) further improves the BER performance of the DAB system. It was seen that to achieve a BER of 10^{-4}, system employing Block coding combined with FEC coding gave a coding gain of 0.5 dB compared with only FEC channel coding for all the three channels. Whereas system employing BCH combined with FEC coding gave a coding gain of 0.5 dB in AWGN channel & Rice channel and coding gain of 1 dB in fading channel for the same BER performance.

5.3 Scope of Future work

The work presented in this thesis can be extended in number of directions. One promising area is extension of these concepts for implementing DAB in Femtocell. The Viterbi decoding of the convolutional codes can be improved, just by using 4 bit soft decision decoding with adaptive selection of quantization levels. Another area could be use of adaptive symbol and frame synchronization using phase reference symbols. Last but not the least use of MPEG-4 audio coding may be applied for source coding.

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Appendix A

This appendix shows the 24 puncturing vectors according to the DAB standard [1].

	(V**$_{PI,0}$,…… ……., **V**$_{PI,31}$**)
PI=1: code rate:8/9	**1100 1000 1000 1000 1000 1000 1000 1000**
PI=2: code rate:8/10	**1100 1000 1000 1000 1100 1000 1000 1000**
PI=3: code rate:8/11	**1100 1000 1100 1000 1100 1000 1000 1000**
PI=4: code rate:8/12	**1100 1000 1100 1000 1100 1000 1100 1000**
PI=5: code rate:8/13	**1100 1100 1100 1000 1100 1000 1100 1000**
PI=6: code rate:8/14	**1100 1100 1100 1000 1100 1100 1100 1000**
PI=7: code rate:8/15	**1100 1100 1100 1100 1100 1100 1100 1000**
PI=8: code rate:8/16	**1100 1100 1100 1100 1100 1100 1100 1100**
PI=9: code rate:8/17	**1110 1100 1100 1100 1100 1100 1100 1100**
PI=10: code rate:8/18	**1110 1100 1100 1100 1110 1100 1100 1100**
PI=11: code rate:8/19	**1110 1100 1110 1100 1110 1100 1100 1100**
PI=12: code rate:8/20	**1110 1100 1110 1100 1110 1100 1110 1100**
PI=13: code rate:8/21	**1110 1110 1110 1100 1110 1100 1110 1100**
PI=14: code rate:8/22	**1110 1110 1110 1100 1110 1110 1110 1100**
PI=15: code rate:8/23	**1110 1110 1110 1110 1110 1110 1110 1100**
PI=16: code rate:8/24	**1110 1110 1110 1110 1110 1110 1110 1110**
PI=17: code rate:8/25	**1111 1110 1110 1110 1110 1110 1110 1110**
PI=18: code rate:8/26	**1111 1110 1110 1110 1111 1110 1110 1110**
PI=19: code rate:8/27	**1111 1110 1111 1110 1111 1110 1110 1110**
PI=20: code rate:8/28	**1111 1110 1111 1110 1111 1110 1111 1110**
PI=21: code rate:8/29	**1111 1111 1111 1110 1111 1110 1111 1110**
PI=22: code rate:8/30	**1111 1111 1111 1110 1111 1111 1111 1110**
PI=23: code rate:8/31	**1111 1111 1111 1111 1111 1111 1111 1110**
PI=24: code rate:8/32	**1111 1111 1111 1111 1111 1111 1111 1111**

Table A.1 Puncturing vectors used for puncturing FIC and MSC data.

Appendix B

This appendix gives the relation between indices i, k' and the parameter n and the carrier index k for the generation of phase reference symbol for all the four DAB transmission modes [1].

k_min	*k_max*	*k'*	*i*	*n*		*k_min*	*k_max*	*k'*	*i*	*n*
-768	-737	-768	0	1		1	32	1	0	3
-736	-705	-736	1	2		33	64	33	3	1
-704	-673	-704	2	0		65	96	65	2	1
-672	-641	-672	3	1		97	128	97	1	1
-640	-609	-640	0	3		129	160	129	0	2
-608	-577	-608	1	2		161	192	161	3	2
-576	-545	-576	2	2		193	224	193	2	1
-544	-513	-544	3	3		225	256	225	1	0
-512	-481	-512	0	2		257	288	257	0	2
-480	-449	-480	1	1		289	320	289	3	2
-448	-417	-448	2	2		321	352	321	2	3
-416	-385	-416	3	3		353	384	353	1	3
-384	-353	-384	0	1		385	416	385	0	0
-352	-321	-352	1	2		417	448	417	3	2
-320	-289	-320	2	3		449	480	449	2	1
-288	-257	-288	3	3		481	512	481	1	3
-256	-225	-256	0	2		513	544	513	0	3
-224	-193	-224	1	2		545	576	545	3	3
-192	-161	-192	2	2		577	608	577	2	3
-160	-129	-160	3	1		609	640	609	1	0
-128	-97	-128	0	1		641	672	641	0	3
-96	-65	-96	1	3		673	704	673	3	0
-64	-33	-64	2	1		705	736	705	2	1
-32	-1	-32	3	2		737	768	737	1	1

Table B.1 Relation between the indices i, k' and n and carrier index k for transmission mode- I [1].

k_min	*k_max*	*k'*	*i*	*n*		*k_min*	*k_max*	*k'*	*i*	*n*
-192	-161	-192	0	2		1	32	1	2	0
-160	-129	-160	1	3		33	64	33	1	2
-128	-97	-128	2	2		65	96	65	0	2
-96	-65	-96	3	2		97	128	97	3	1
-64	-33	-64	0	1		129	160	129	2	0
-32	-1	-32	1	2		161	192	161	1	3

Table B.2 Relation between the indices i, k' and n and carrier index k for transmission mode- II [1].

k_min	*k_max*	*k'*	*i*	*n*		*k_min*	*k_max*	*k'*	*i*	*n*
-96	**-65**	**-96**	**0**	**2**		**1**	**32**	**1**	**3**	**2**
-64	**-33**	**-64**	**1**	**3**		**33**	**64**	**33**	**2**	**2**
-32	**-1**	**-32**	**2**	**0**		**65**	**96**	**65**	**1**	**2**

Table B.3 Relation between the indices i, k' and n and the carrier index k for mode- III [1].

k_min	*k_max*	*k'*	*i*	*n*		*k_min*	*k_max*	*k'*	*i*	*n*
-384	**-353**	**-384**	**0**	**0**		**1**	**32**	**1**	**0**	**0**
-352	**-321**	**-352**	**1**	**1**		**33**	**64**	**33**	**3**	**1**
-320	**-289**	**-320**	**2**	**1**		**65**	**96**	**65**	**2**	**0**
-288	**-257**	**-288**	**3**	**2**		**97**	**128**	**97**	**1**	**2**
-256	**-225**	**-256**	**0**	**2**		**129**	**160**	**129**	**0**	**0**
-224	**-193**	**-224**	**1**	**2**		**161**	**192**	**161**	**3**	**1**
-192	**-161**	**-192**	**2**	**0**		**193**	**224**	**193**	**2**	**2**
-160	**-129**	**-160**	**3**	**3**		**225**	**256**	**225**	**1**	**2**
-128	**-97**	**-128**	**0**	**3**		**257**	**288**	**257**	**0**	**2**
-96	**-65**	**-96**	**1**	**1**		**289**	**320**	**289**	**3**	**1**
-64	**-33**	**-64**	**2**	**3**		**321**	**352**	**321**	**2**	**3**
-32	**-1**	-32	**3**	**2**		**353**	**384**	**353**	**1**	**0**

Table B.4 Relation between the indices i, k' and n and the carrier index k for mode- IV [1].

j	*0*	*1*	*2*	*3*	*4*	*5*	*6*	*7*	*8*	*9*	*10*	*11*	*12*	*13*	*14*	*15*
$h_{0,\ ,j}$	**0**	**2**	**0**	**0**	**0**	**0**	**1**	**1**	**2**	**0**	**0**	**0**	**2**	**2**	**1**	**1**
$h_{1,\ ,j}$	**0**	**3**	**2**	**3**	**0**	**1**	**3**	**0**	**2**	**1**	**2**	**3**	**2**	**3**	**3**	**0**
$h_{2,\ ,j}$	**0**	**0**	**0**	**2**	**0**	**2**	**1**	**3**	**2**	**2**	**0**	**2**	**2**	**0**	**1**	**3**
$h_{3,\ ,j}$	**0**	**1**	**2**	**1**	**0**	**3**	**3**	**2**	**2**	**3**	**2**	**1**	**2**	**1**	**3**	**2**

j	*16*	*17*	*18*	*19*	*20*	*21*	*22*	*23*	*24*	*25*	*26*	*27*	*28*	*29*	*30*	*31*
$h_{0,\ ,j}$	**0**	**2**	**0**	**0**	**0**	**0**	**1**	**1**	**2**	**0**	**0**	**0**	**2**	**2**	**1**	**1**
$h_{1,\ ,j}$	**0**	**3**	**2**	**3**	**0**	**1**	**3**	**0**	**2**	**1**	**2**	**3**	**2**	**3**	**3**	**0**
$h_{2,\ ,j}$	**0**	**0**	**0**	**2**	**0**	**2**	**1**	**3**	**2**	**2**	**0**	**2**	**2**	**0**	**1**	**3**
$h_{3,\ ,j}$	**0**	**1**	**2**	**1**	**0**	**3**	**3**	**2**	**2**	**3**	**2**	**1**	**2**	**1**	**3**	**2**

Table B.5 Time-Frequency-Phase parameter *h* values [1].

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