DAB SYSTEM: TRANSMITTER, RECEIVER AND SVM CLASSIFIER

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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UNDER THE GUIDANCE OF

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Date: 25.05.2011

CERTIFICATE

This is to certify that the thesis entitled**, "DAB: Transmitter, Receiver and SVM Classifier "** submitted by Mr. **Ashish Agarwal** in partial fulfillment of the requirements for the award of Master of Technology Degree in Electronics and Communication Engineering with specialization in **"Telematics and Signal Processing"** during session 2010-11 at the 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.

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Abstract

The aim of this thesis is to implement a Digital Audio Broadcast (DAB) system in base-band. In this work I designed a DAB transmitter for mode II, following the ETSI standards for DAB transmitter was designed. This mode has been chosen for simulation because of its suitability in the local area terrestrial broadcasting and to be a model that presents other transmission mode implementation. The physical modulation part of the DAB for transmission modes II as well as its receiver is implemented.

The prime focus of this thesis is on the reception side. In the receiver side, receiver synchronization has been implemented and a new classification method for the QPSK mapping proposed. Support vector Machine (SVM) is used as a QPSK Classifier and as a channel estimator for DAB system. It uses the continuous learning algorithm, for training and testing. This learning algorithm needs one-time training for the classification of first DAB symbol, and for the remaining symbols the system learns the pattern from the previous symbol estimated by classifier.

A frame-based processing was used in this study. Performance studies for AWGN and Rayleigh channels have been conducted. Bit error rate (BER) has been considered as the performance index.

The result obtained showed that the implemented system work successfully in AWGN channel. For Rayleigh fading channel, the system performance is desirable in urban area, below the speeds of 50-70 Kmph of the receiver.

All the simulation work is implemented using the Microsoft Windows Operating system and MATLAB.

1.1 Introduction

Guglielmo Marconi conducted his first experiments in 1895, with wireless telegraphy in Italy. A century later, radio broadcasting became one of the most widespread mass media. At this age, all the broadcast transmission standard was based on analog AM and FM. These radio services provide good quality of services for fixed reception but for mobile reception system was badly affected by multipath propagation. This causes fading and occasional loss of signals. Today we live in a world of digital communication systems and services. These analog standards are now failing to provide the quality of services to the listeners they expect. For that reason, the universal digital multimedia broadcasting system Eureka 147 DAB was developed.

Digital Audio Broadcasting (DAB) is completely digital method of delivering radio services from studio to the receiver. It is the development of European consortium called Eureka 147. It having all capability to replace existing AM and FM broadcast services. DAB is very well suited for mobile reception and provide very high robustness against multipath reflection and inter symbol interference (ISI). DAB is designed to operate in any frequency range in the VHF and UHF range for terrestrial, satellite, hybrid (satellite with complementary terrestrial) and cable delivery. DAB works very differently from conventional broadcasting systems. Most of the system components such as perceptual audio coding, channel coding and modulation, multiplex management or data transmission protocols are new solutions.

1.2 Thesis Outline

Chapter 2: This chapter describes the introduction to DAB and the advantages of DAB. This chapter gives the introduction to the technical overview of the DAB an a details of DAB modes and parameters. This chapter also gives the introduction to OFDM.

Chapter 3: This chapter gives the detailed description to the simulation model of DAB transmitter in mode II based to the ETSI standard.

Chapter 4: This chapter gives the detailed description of the DAB receiver. It also provides an introduction to SVM and its use as a QPSK classifier and a channel estimator for DAB system.

Chapter 5: This chapter gives the results of the simulation and also gives the performance analysis of DAB system

Chapter 6: 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.

2.1 DAB: An Introduction

The digital radio system DAB (Digital Audio Broadcasting) is a very innovative and universal multimedia broadcast system. DAB is a digital method of delivering radio services from the studio to the receiver. It promises to provide sound of compact-disk quality, nearly free from multipath distortion or other transmission interferences. Moreover very well suited for mobile reception and provide very high robustness against multipath reception. Besides high-quality digital audio services (mono, two-channel or multichannel stereophonic), DAB is able to transmit program-associated data and a multiplex of other data services (e.g. travel and traffic information, still and moving pictures, etc.) [1].

DAB works very differently from conventional broadcasting systems. Most of the system components such as perceptual audio coding, channel coding and modulation, multiplex management or data transmission protocols are new solutions and typically not so familiar to the expert in existing analogue or digital broadcast systems.

Broadly speaking, in AM and FM radio, the original audio signals are broadcast and then reproduced-together with any interference signals-by receivers. In DAB, the original audio coding signals are converted from the very beginning. The conversion to digital form and the all important steps in reducing the number of bits that must be transmitted, is achieved by a type of perceptual coding, that relies on masking properties of human hearing system.

The DAB bit-rate reduction based on these hearing system function ranges from approximately 5:1 to 10:1, depending on the particular algorithm. They are “lossy” data reduction methods in that unnecessary and imperceptible information is removed and lost. The original full data can never be precisely reproduced, but the transmitted data is closely enough to original data to avoid perceptible differences.

Additional bits in the coded bit stream are used to tell the receiver whether or not they have received all the data bits needed to reconstruct the original digital signal.

When the codes bit stream impressed upon a radio station's transmitted digital signal, any extraneous interference is not reproduced [1]-[3]

2.2 History of DAB

Radio broadcasting is one of the most widespread electronic mass media. Since the beginning of broadcasting in the early 1920s, the market has been widely covered by the AM and FM emission standards. These broadcasting services have achieved the technological and operational maturity such that there was hundreds of programs provider, thousands of HF transmitter and billions of receiver word wide. They offer a large diversity of speech and audio programs to the listeners.

Today we live in a world of digital communication systems and services. These analog standards are now failing to provide the quality of services to the listeners they expect. Now the essential parts of the production processes in radio houses was to change to digital ones in recent times. The world was changing from analog to digital techniques to produce the optimum performance, with acceptable cost to a large consumer market. There are now many mass-storage digital medium to offer superior sound quality like CD's, hard disks, digital compact cassettes, Mini-Disks, DVDs etc or streaming and download formats (such as MP3) for distribution via Internet. Consequently, broadcast transmission systems now tend to change from conventional analogue transmission to digital. The first steps in the introduction of digital broadcasting services were taken by the systems NICAM 728 (Near Instantaneously Companded Audio Multiplex, developed by the BBC for stereo television sound in the VHF/UHF bands), DSR (Digital Satellite Radio, which was already shut down), or ADR (Astra Digital Radio), but none were suited to replace the existing conventional services completely, especially for mobile reception. For that reason, the universal digital multimedia broadcasting system Eureka 147 DAB was developed and is now being introduced worldwide [1][2].

The first digital sound broadcasting systems were developed in the early 1980s for satellite delivery providing CD-like audio quality.

Primarily they broadcast in 10 to 12 GHz bands, employed very little sound data compression. They were not suitable for mobile reception. Thus, it was not possible to serve all users, especially not for mobile users. These well-established FM radio were not able to provide satellite delivery, namely ‘local services’. As a result terrestrial digital sound broadcasting was come into consideration. Therefore, towards the end of 1986 a consortium of 19 organizations from France, Germany, The Netherlands and the United Kingdom had signed a co-operation agreement and applied for notification as a Eureka project. The Eureka 147 project was initiated through the collaboration between IRT (Institut für Rundfunktechnik) and CCETT (Commun d’Etudes de Te´ le´diffusion et Te´ le´communications), both of which undertake research on behalf of broadcasting organization in their respective countries.

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

- High quality audio comparable to that of the CD;
- Suitable for mobile reception in a car, even at high speeds;
- Efficient use of frequency spectrum;
- Transmission capacity for ancillary data;
- Low transmitting power;
- Terrestrial, cable and satellite delivery options;
- Easy-to-operate receivers;

In the beginning, DAB system was created by the conjunction of two advanced digital techniques, audio bit-rate reduction, pioneered by IRT, and RF transmission using a technique known as COFDM, pioneered by CCETT. The BBC contribution has been diverse, including the third major component in the system, the dynamic, flexible multiplex and system control mechanism. The BBC contribution also involved research into many aspects of audio, data and RF parts of the system, as well as the major role in determining the final system parameters and drafting the written specification for the system.

As the system has evolved, parameters such as bandwidth of the RF signal have been changed. The first system approach considered at least 16 stereo programs of CD audio quality plus ancillary data to be transmitted in the 7 MHz bandwidth of a television channel. This definitely cannot be achieved by simply transmitting the combined net bit rates of 16 CD-like program channels, which are around 1.4 Mbps each, over the TV channel. So a high degree of audio data compression without any perceptible loss of audio quality was mandatory. Data rates below 200 kbps per stereo channel had to be achieved. In Canada experiments with the COFDM system revealed that substantial performance degradation begins around 1.3MHz and lower. So, a reasonable bandwidth for a DAB channel or ‘DAB block’ was defined as 1.5 MHz. Starting at 7 MHz to fill a continental television channel, changes have been made to 3.5 MHz and then to 1.5 MHz in order to fit 4 DAB signal, plus guard bands into such a television channel, the final specification correspond to 1.537 MHz. During this evolution, extensive field test taken extensive field tests of the system were undertaken by Research Department using experimental transmitter in London (Crystal Palace) and Birmingham and latterly a mini-network of low-power transmitter at existing UHF television transmitter site in Surrey, followed by a London-wide network. Finally, the DAB system completed in autumn 1994 [1][2][8].

2.3 Effect of Multipath reflection

Analog radio networks are able to provide good quality of radio broadcasting services for fixed reception. But for mobile reception suffers loss of broadcast quality. For example, FM reception is badly affected by shadowing (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) and AM system is badly affected by seasonal propagation variation that cause fading and loss of signal. These problems occur due to multipath reflections. The multipath effect is shown in figure 2.1

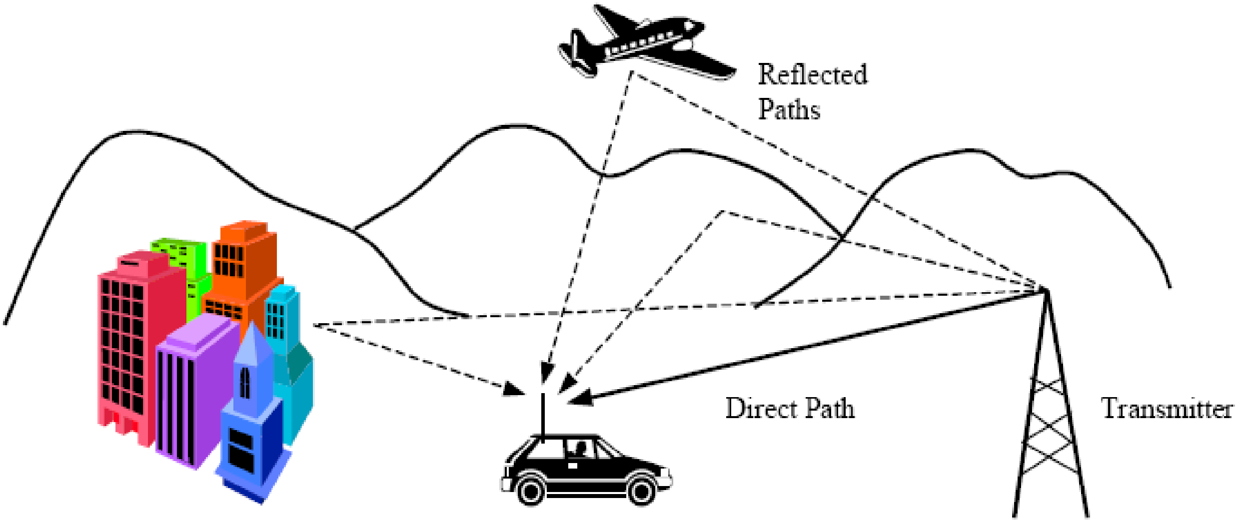

Figure 2. 1 Multipath Effect

To solve these problems and provide audio broadcasting of compact disk quality Eureka project was developed. DAB eliminates interference and problem of multipath reflection by using a non-directional whip antenna [2][4][5].

2.4 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. This includes:

1. Rugged, reliable delivery to fixed, portable and mobile receivers for interference free reception with just a simple, non-directional whip antenna.

2. **Spectrum efficient:** This means that it will be possible to increase the number of radio stations, without congesting the radio waves. As more efficient audio coding (compression) methods are introduced, it will be possible to carry even more radio programs with no degradation to existing services, and without needing to modify existing receivers.
3. **Echoes:** DAB system is able to use "passive echoes" such that they add in a constructive manner to the direct signals already received.

The Eureka system is also able to use “active echoes” constructively – i.e. delayed signals generated by other co-channel transmitters.

4. **Single Frequency Networks (SFNs):** SFN enables all transmitters covering a particular area with same set of sound programs to operate on the same radiofrequency channel. Although the signals emitted by the various transmitters are received with different time delays, the receiver automatically selects the stronger signal without interference from overlapping zones. This eliminates the problem of having to return a receiver at frequent intervals such as in car, and allow efficient use of spectrum.

5. High-quality digital audio.

6. **Flexibility and choice:** The DAB signal is essentially a highly flexible over-air 'data-pipe' that can broadcast a wide range of service types, from purely audio-based to multimedia. Typically, each multiplex can broadcast five or more high-quality audio services plus a number of extra services. The multiplex can be dynamically reconfigured to introduce additional services, temporary or permanent, either with time-of-day or on a day-to-day basis.

7. Rather than searching wavebands, users can select all available stations or preferred formats from a simple text menu.

8. Added-value system features that will allow enhancements to existing radio services such as text, graphics and still-pictures, and provide the opportunity to introduce innovative new services, such as multimedia radio-with-pictures and broadcast web-sites.

2.5 Working of DAB

In this section we will discuss about the principle working of DAB transmitter according to the ETSI standard [6].

Eureka 147 digitally combines multiple audio channels, and the combined signal is interleaved in both frequency and time across a wide broadcast band. This approach provides spectrum- and power-efficient transmission and reliable reception, even over a multipath fading channel. The Conceptual Block Diagram for DAB transmitter is given in the figure 2.2. Each block is labeled in order to indicate the function it performed.

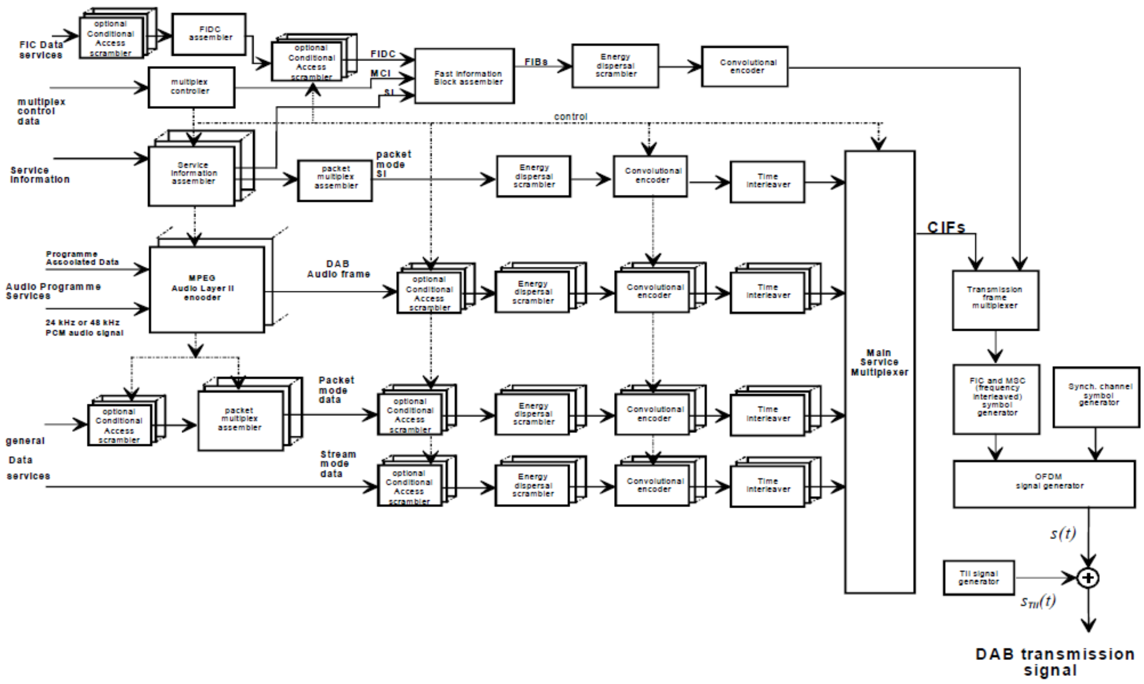

Figure 2. 2 Transmitter Block Diagram

The block diagram follows the following steps:

a. Audio data as well as other data is individually encoded with channel coders, then error protected and time interleaved

b. A multiplexer combines many different services to create a main service channel (MSC).

c. The multiplexer output is frequency interleaved and synchronization symbols are added

d. Coded Orthogonal Frequency Division Multiplexing (COFDM) with quadrature phase shift keying modulation is employed for each carrier to create an ensemble DAB signal.

In this project work we study two elements in detail. These are:

- Multiplexing and Transport Mechanism
- COFDM Modulation

2.6 Multiplexing and Transport Mechanism

The data from individual services such as audio, data are to be initially encoded at individual level, error protected and time interleaved. The output services are combined into a single data stream ready for transmission. The process of combining data stream is known as multiplexing and the resulting data stream is called multiplex. In DAB several programs are multiplexed into a so-called ensemble with a bandwidth of 1.536 MHz

The transmission frame consists of sequence of three groups of OFDM symbols: Synchronization channel symbol, Fast Information Channel (FIC) symbols, and Main Service Channel symbol. The synchronization channel symbol comprises the null symbol and the phase reference symbol.

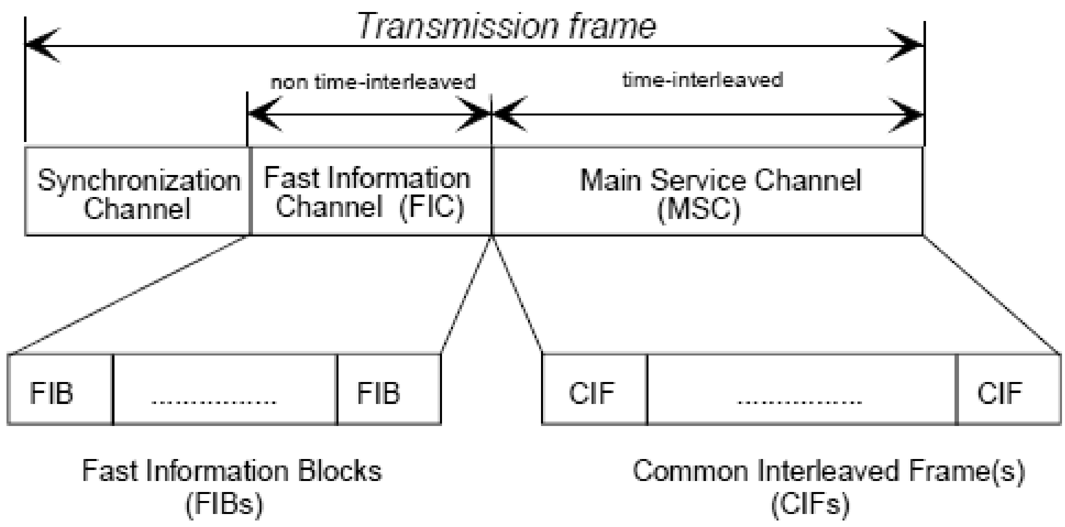

Figure 2. 3 Transmission Frame

The description of three channels is given blow:

1. **Main Service Channel (MSC):** 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 organization of the sub-channels and service components is called the multiplex configuration.

2. **Fast Information Channel (FIC):** used for rapid access of information by a receiver. It is used to send the Multiplex Configuration Information (MCI) and optionally Service Information and data service. It is a non-time-interleaved data channel with fixed equal error protection.

3. **Synchronization channel:** used internally within the transmission system for basic demodulator functions, such as:
   - Transmission frame synchronization
   - Automatic frequency control
   - Channel state estimation
   - Transmitter identification.

Each channel supplies data from different sources and these data are provided to form a transmission frame. Both the organization and length of a transmission frame depend on the transmission mode. The Fast Information Block (FIB) and the Common Interleaved Frame (CIF) are introduced in order to provide transmission mode independent data transport packages associated with the FIC and MSC respectively.

2.7 COFDM

2.7.1 Introduction

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. OFDM is multi-carrier system. Data is transmitted at a low symbol rate using many narrow-band carriers rather than at a higher rate using single wide-band carrier. Carriers are arranged to be mutually orthogonal, so each carrier has its peak amplitude, in the frequency domain, where all other have a zero-crossing [9][10].

2.7.2 Basic Principle of OFDM

OFDM is a parallel data transmission system, in which several sequential streams of data are transmitted. This parallel approach spreads the frequency selective fading over many symbols. This effectively randomizes the burst error caused by fading or impulse interference so that, instead of several adjacent symbols being completely destroyed, many symbols are only slightly destroyed. This allows successful reconstruction of a majority of them even without forward error correction (FEC).

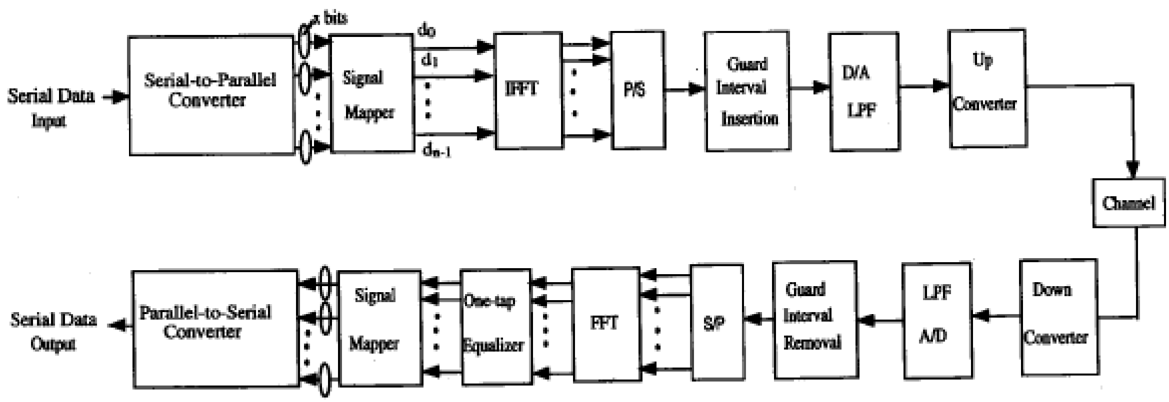

Figure 2. 4 FFT-based OFDM System

The incoming serial data is first converted from serial to parallel and grouped into x bits each to form a complex number. The number x determines the signal constellation of the corresponding subcarrier, such as 16 QAM or 32 QAM. The complex numbers are modulated in a baseband fashion by the inverse FFT and converted back to serial data for transmission. A guard interval is inserted between symbols to avoid inter-symbol interference (ISI) caused by multipath distortion. The discrete symbols are converted to analog and low-pass filtered for RF up-conversion. The receiver performs the inverse process of the transmitter.

Consider a data sequence $(d_0, d_1, d_2 \ldots d_{N-1})$, where d_n is a complex symbol.

The complex symbol d_n can be expressed as:

$$d_n = a_n + jb_n \tag{2.1}$$

Where

$$a_n = \cos \Phi_n \,, b_n = \sin \Phi_n and\ \Phi\ is\ the\ phase$$

The waveform of an individual sub-carrier at frequency n$f_0$can be defined as:

$$\begin{aligned} x_n(t) &= \sqrt{a_n^2 + b_n^2}\cos\left(2\pi(f_c + nf_o)t + \Phi_n\right) \\ &= a_n\cos(2\pi(f_c + nf_o))t - b_n\sin(2\pi(f_c + nf_o))t \end{aligned} \tag{2.2}$$

Where,

$$f_0 = 1/T \qquad \text{and} \qquad \Phi_n = tan^{-1}\frac{b_n}{a_n}$$

f_c is the central frequency of the signal.

When this is summed over all N sub-carriers, the generated OFDM signal is:

$$\mathrm{x(t)} = \sum_{n=0}^{N-1}[a_n \cos\{2\pi(f_c + nf_o)t\} - b_n \sin\{2\pi(f_c + nf_o)\}\, t] \tag{2.3}$$

Equation can be written as:

$$\mathrm{y(t)} = \mathrm{Re}\{\sum_{n=0}^{N-1} d_n \exp(j2\pi(f_c{+}nf_o)t)\}$$

$$= \mathrm{Re}\left\{\left[\underbrace{\textstyle\sum_{n=0}^{N-1} d_n \exp\left(\frac{j2\pi kn}{N}\right)}_{IDFT}\right]\exp\left(j2\pi f_c t\right)\right\} \tag{2.4}$$

Where,

$$t = \frac{k}{Nf_o}$$

The DFT transformation (to modulate and demodulate) require N^2 complex product. In order to work with real time system, both transmitter and receiver can be implemented using efficient FFT techniques which reduces the number of operations from N^2 in DFT down to about NlogN [4][5][9]]10].

2.8 DAB Transmission Signal

As we discussed previously, the transmitted frame consist of synchronization channel, FIC and MSC. Each transmission frame is divided into a sequence of OFDM symbols, each made up of a fixed number of carriers. The number of OFDM symbols in a transmission frame depends on the transmission mode.

As per ETSI standard [6], the first two symbols of the transmission frame are made up of a synchronization channel regardless to the transmission mode. The first symbol of the transmission frame should be a Null symbol of duration T_{NULL} and the remaining to be made of OFDM symbols of duration Ts. Each of these OFDM symbols is equally spaced with carrier spacing 1/Tu.

The defined DAB transmission signal s(t) is given by:

$$s(t) = Re\left\{e^{2j\pi f_c t} \sum_{m=-\infty}^{\infty} \sum_{l=0}^{L} \sum_{k=-K/2}^{K/2} z_{m,l,k} \times g_{k,l}(t - mT_F - T_{NULL} - (l-1)T_s)\right\} \tag{2.5}$$

With,

$$g_{k,l}(t) = \begin{cases} 0, \; for\; l = 0 \\ e^{2j\pi k(t-\Delta)/T_U}. Rect\left(\frac{t}{T_s}\right), \; for\; l = 1,2,\dots,L \end{cases} \tag{2.6}$$

And

$$T_s = T_U + \Delta \tag{2.7}$$

Where,

L	is the number of OFDM symbols per transmission frame (the Null symbol being excluded);
K	is the number of transmitted carriers;
TF	is the transmission frame duration;
TNULL	is the Null symbol duration;
Ts	is the duration of the OFDM symbols of indices l=1,2,3,……., L;
TU	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;
fc	is the central frequency of the signal.

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

2.9 DAB Modes and system parameters

The Eureka 147 DAB system has four transmission modes of operation named as mode-I, mode-II, mode- III, and mode-IV, each having its particular set of parameters. The 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.

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.

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. 1 DAB TRANSMISSION FRAME COMPOSITION

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 and width of 1.536 MHz.

Parameter	Transmission mode -I	Transmission mode -II	Transmission mode -III	Transmission mode -IV
K	1536	384	192	768
L	76	76	153	76
TF	196608 T 96 ms	49152 T 24 ms	49152 T 24 ms	98304 T 48 ms
TNULL	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
TU	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 Carrier spacing	375 MHz 1 kHz	1.5 GHz 4 kHz	3 GHz 8 kHz	750 MHz 2 kHz
FFT length	2048	512	256	1024

Table 2. 2 SYSTEM PARAMETER OF FOUR DAB TRANSMISSION MODE

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). 23
Transmission mode –III	is available for satellite broadcasting below 3 GHz (UHF L-and).
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

3.1 DAB Transmitter Simulation Model

The DAB transmitter is based on ETSI standard [6]. DAB transmission mode II is used for simulation. All the work developed in the simulation model follows this mode standard parameter. DAB mode II is suitable for local area terrestrial broadcasting. The simulation is developed in base-band transmission. The RF section for both receiver and transmitter was not studied in this work. All the simulation work developed under MTLAB environment. The block diagram of the transmitter is given below:

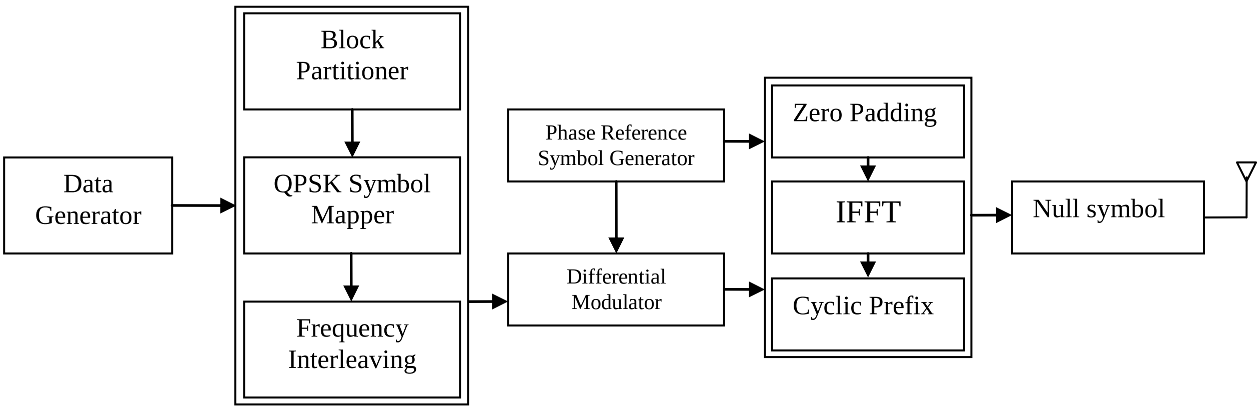

Figure 3. 1 Transmitter Block Diagram

3.2 Data Generator Block

The first block in the DAB Transmitter simulation model, it generates random binary data bit sequence for FIC and MSC. The total random bits generated for one transmission frame excluding synchronization channel, are given by:

$$\text{Total bit} = \text{FIC bit} + \text{MSC bit}$$

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. 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_bit = No. of OFDM symbols*bits/OFDM symbol

= 3*768

= 2304 bits.

msc_bit = No. of OFDM symbols*bits/OFDM symbol

= 72*768

= 55296 bits.

The total data bits for each transmission frame are 57600 bits.

3.3 Data Mapper Block

The Data Mapper Block consist of three sub-blocks: block partitioner, QPSK symbol mapper block, and frequency interleaving block. This block divides the generated bits from Data generator block, into data blocks and maps the QPSK bit in each data block. These mapped bits further undergoes for frequency interleaving. This Data mapper block converts the symbol into phase.

3.3.1 Block Partitioner

This block is responsible for partitioning the serial data array into the parallel data block. Each Data block associated to OFDM symbols. This operation is transmission mode dependent. The serial data bit (57600) partitioned into 75 consecutive blocks of 768 bits to be transmitted as OFDM symbols of index l=2,3,4,……76 for MODE II. As shown in figure 3.2:

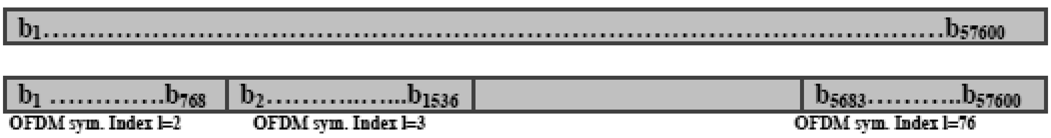

Figure 3. 2 Block Partition

3.3.2 QPSK Mapping

In each data block, each OFDM symbol is mapped into QPSK symbol constellation. According to the DAB Standard, defined as:

$$q_{l,n} = \frac{1}{\sqrt{2}}\left[\left(1 - 2p_{l,n}\right) + j\left(1 - 2p_{l,n+k}\right)\right] \text{ for n = 0,1,2,…,K-1} \qquad (3.1)$$

Where,

l = 2,3,4…,76

K is the number of carrier used

Each data block of 768 bits are mapped to the 384 complex QPSK symbols. The first bit in each bit-pair $p_{l,n}$ is used to generate I-component and the second bit $p_{l,n+k}$ is used to generate the Q-component of the generated symbol stream.

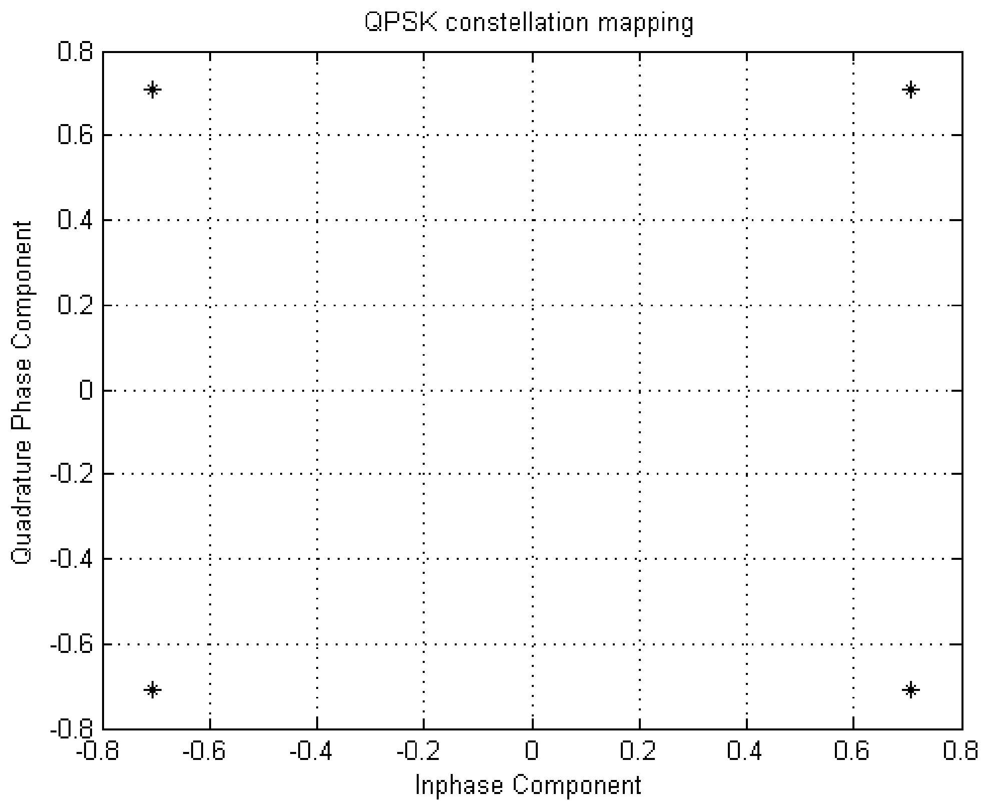

Figure 3. 3 QPSK Constellation Mapping

3.4 Frequency Interleaving Block

Frequency interleaving is used to eliminate the effects of frequency 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 defines the correspondence between the index n of the QPSK symbols $q_{l,n}$ and the carrier index k (- K/2 ≤ k < 0 and 0 < k ≤ K/2).

The QPSK symbols shall be re-ordered according to the following relation:

$$y_{l,k} = q_{l,n} \, for \; l = 2,3,4, \ldots, L$$

With k = F(n), where F is a function defined in table 3.1 for transmission mode-II.

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

$$\prod(i) = \prod(i-1) + 127 \; (mod \; 512) and \prod(0) = 0; \qquad (3.2)$$

For i = 1,2,…,511.

$$d_n = \prod(i) \;\; (excluding \; 256)$$

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

$$k = F(n) = d_n - 256 \qquad (3.3)$$

The interleaving rule is illustrated in Table 3.1 given blow:

i	π (i)	dn	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

Table 3. 1 interleaving rule for transmission mode II

3.5 Phase Reference Symbol

$$z_k = \begin{cases} e^{j\varphi_k}, \ for \frac{-K}{2} \le k < 0 \ and \ 0 < k \le \frac{K}{2} \\ 0, \ for \ k = 0 \end{cases} \tag{3.4}$$

The value of φ_k will be obtained from the following expression:

$$\varphi_k = \frac{\pi}{2}\left(h_{i,k-k'} + n\right) \tag{3.5}$$

Where the indices i,k' and the parameter n are specified as the function of carrier index k. The values of the parameter $h_{i,j}$ as a function of i and j are obtained from the

Table 3.2 given blow

k in the range of min	max	*k'*	*i*	*n*
-192	-161	-192	0	2
-160	-129	-160	1	3
-128	-97	-128	2	2
-96	-65	-96	3	2
-64	-33	-64	0	1
-32	-1	-32	1	2

k in the range of min	max	*k'*	*i*	*n*
1	32	1	2	0
33	64	33	1	2
65	96	65	0	2
97	128	97	3	1
129	160	129	2	0
161	192	161	1	3

Table 3. 2 Relation between the indices i, k' and n and the carrier index k for transmission mode II

Relation between the indices i, k' and n and the carrier index k for transmission mode-II.

Generated Phase Reference symbol waveform and its constellation are shown below

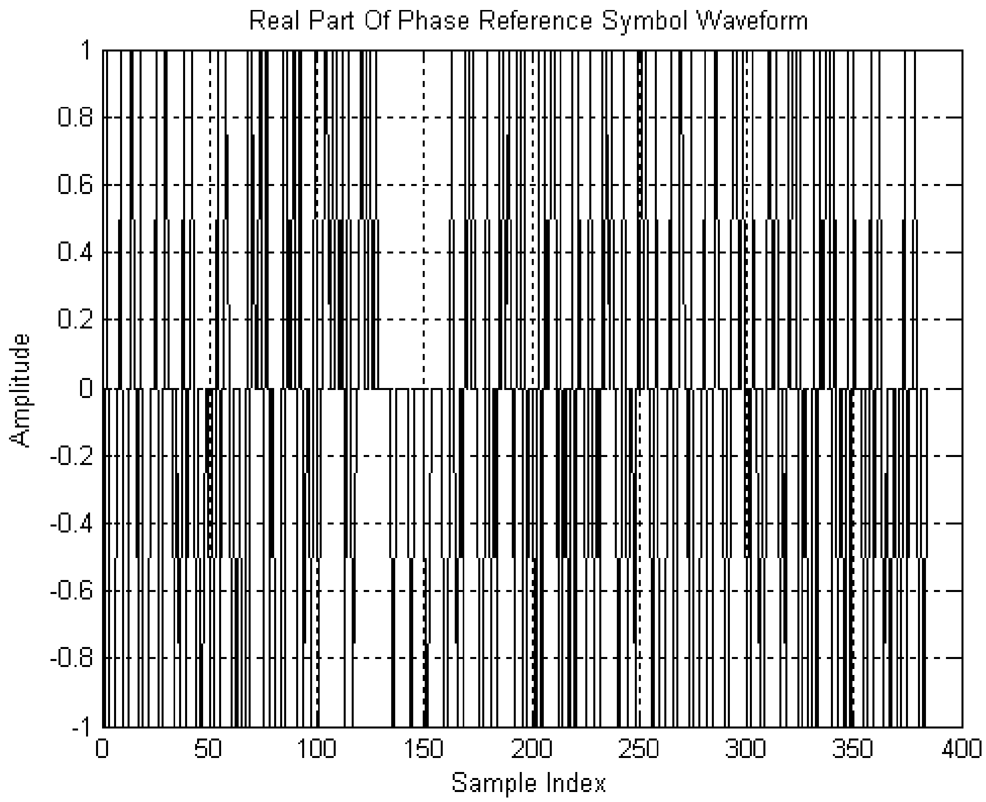

Figure 3. 4 Real Part of the Phase Reference Symbol

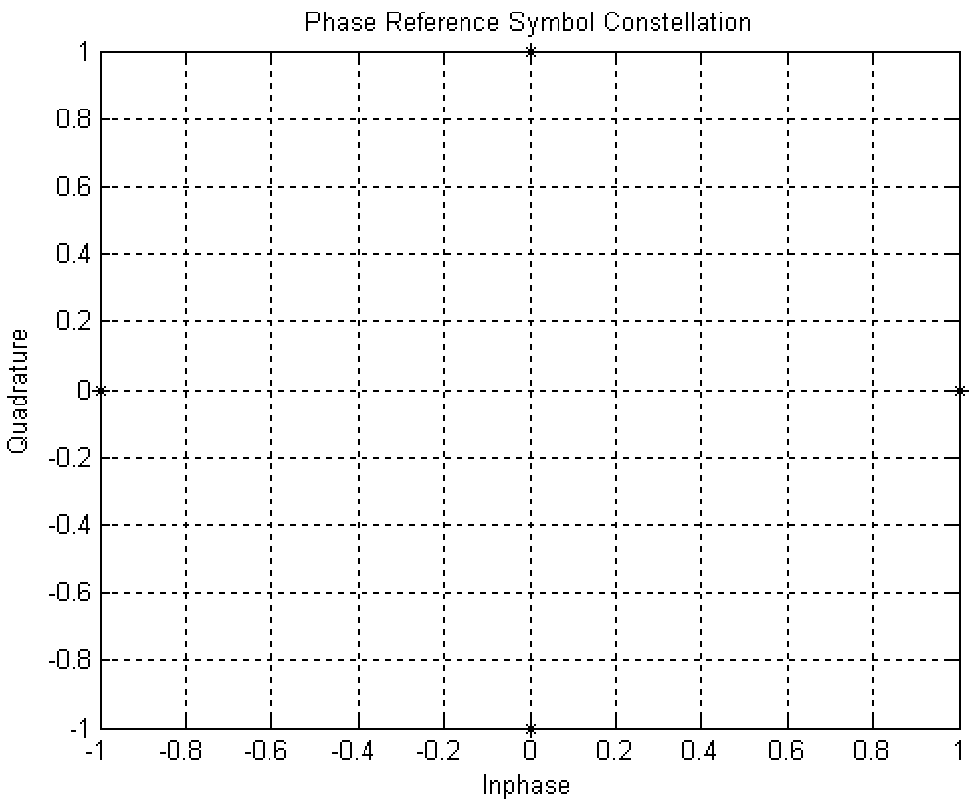

Figure 3. 5 Phase Reference Constellation

3.6 Differential Modulator

In wireless communications multipath spreading occurs, which causes degradation in signal and cause an offset in the phase of the carriers. So to avoid this, differential modulation is used. In this the information is sent through the difference between the phases of two successive symbols. This simplifies synchronization and timing recovery.

π/4 shift D-QPSK modulation is applied to QPSK symbols on each carrier and is defined by:

$$z_{l,k} = z_{l-1,k} \times y_{l,k} \tag{3.6}$$

For l = 2,3,…,L

and $\frac{-K}{2} \leq k \leq \frac{K}{2}$

where,

z	complex D-QPSK symbols, i.e. output from the differential modulator
y	input QPSK symbol block
l	OFDM symbol index
k	carrier index

3.7 OFDM Symbol Generator

This block is the main block of the DAB transmitter. This block transforms the frequency domain samples of the each DQPSK block into a time domain samples. The output from DQPSK block is fed to the input of the OFDM block.

The DQPSK block contains 75 DQPSK modulated OFDM symbols and one phase reference symbol is added at the start of frame to achieve the total number of ODFM symbols desired for one frame.

3.8 Zero Padding

The FFT/IFFT algorithm has a good performance if the number of carriers is an integer power of 2. So for efficient use of FFT/IFFT algorithm we make the each OFDM symbols length 512 from 384 by appending 128 zeros. This is the one task of this block.

Second task of this block is to rearrange the lower frequency to the first part of the OFDM symbols and the upper frequency in the last part of the OFDM.

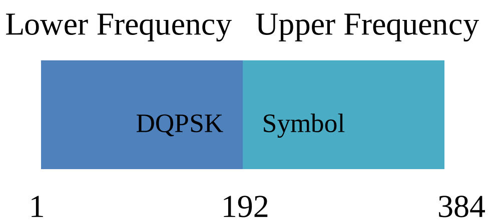

Figure 3. 6 DQPSK symbol block before zero padding and rearrangement

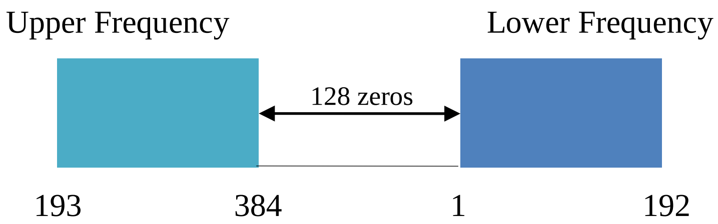

Figure 3. 7 DQPSK symbol block after zero padding and rearrangement

3.8.1 IFFT

In this block the transformation from the frequency domain to time domain is performed. The frequency domain symbol in each DQPSK symbols block including the phase reference symbol is transformed to time domain symbol by performing inverse Fourier transformation.

3.8.2 Cyclic prefix

This block adds the cyclic prefix at the start of each symbol. To perform this we append the copy of last 126 samples at the start of symbols. So now the total length of each symbol become 638 (126+512).

3.9 Null Symbol Generator

Null symbol is the first symbol in synchronization channel and in transmission frame as well. The length of null symbol T_{NULL} is equivalent to 664 samples in Mode II. To ensure that no main signal is transmitted during the period of TNULL, this block appends the 664 null samples at the start of the frame. This block is also responsible to convert the parallel data to serial data.

3.10 Frame Structure

The serial stream of data consisting TNULL at the start of the frame along with Phase Reference Symbol and 75 D-QPSK modulated symbol makes the complete frame. The structure of a frame in time domain is shown in figure 3.8 shown below.

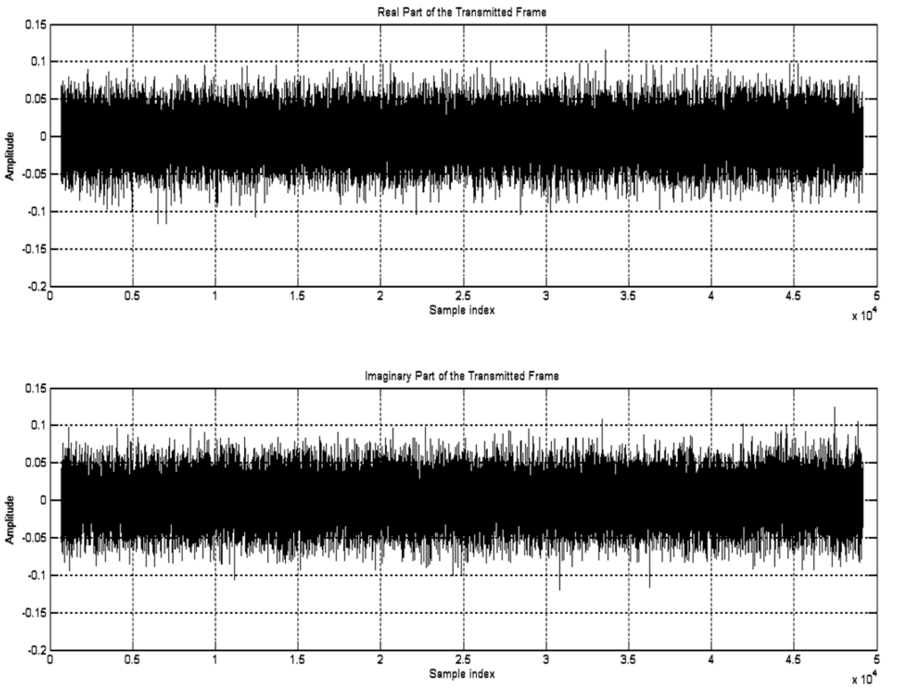

Figure 3. 8 Transmission Frame

4.1 Channel

A fixed receiver and a mobile receiver is has been simulated. Additive White Gaussian noise channel and Rayleigh Fading Channel is used for the performance study of the receiver.

4.2 Receiver

ETSI DAB standard provides only the transmission standard. Receiver design completely based on the receiver designer and manufacturers; however the receiver should be able to work with DAB signal.

The receiver design is the focus point of my work. In this I have designed the basic receiver with conventional synchronization method with a new QPSK classification scheme. In this classification a new continuous learning approach for SVM classifier is used. This new classification scheme will be discussed in different section. The basic block diagram is shown is Figure 4.1

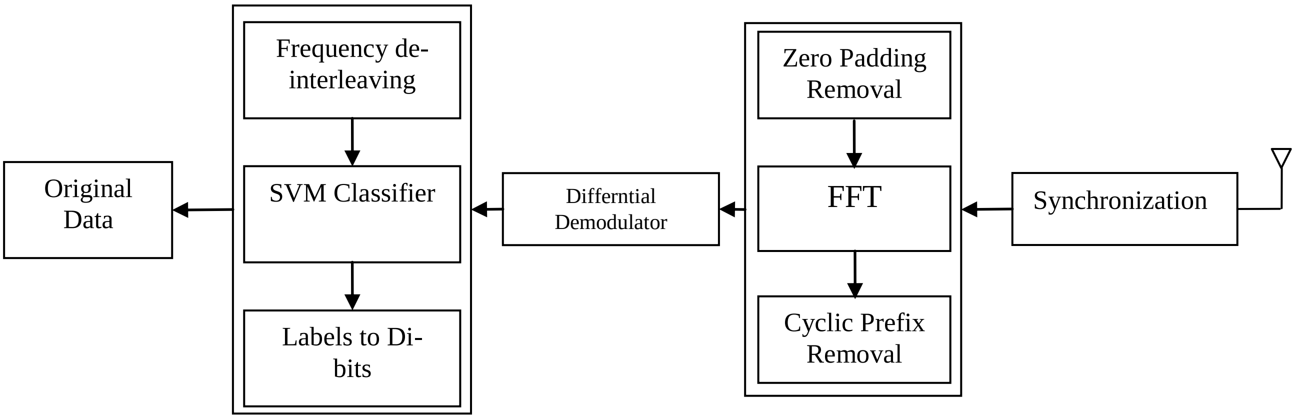

Figure 4. 1 Receiver Block Diagram

4.3 Synchronization

Synchronization is a challenging but very important issue in a digital communication system. We address two problems in the design of receiver. One problem is the unknown symbol arrival time and second problem is the mismatch of the oscillator in the transmitter and the receiver. These problems need to be eliminated, so the receiver need to know about the carrier phase, carrier frequency offset, symbol timing and frame timing for synchronization.

In this section we study about the synchronization. The receiver simulation has been carried in base band, it does not include the RF section neither analog to digital conversion [4][11]-[13].

4.3.1 Symbol Timing Estimation

We provide the transmission signal to the correlator and the output of the corrlator is input to the one-sample differentiator in Fig: 4.2 make its output in the last T_g period of a symbol close to zero. Now the output has two different distributions with the same (zero) mean, but different variances in different parts of the OFDM signal. One is uniformly distributed between [-π, π] with zero mean and constant variance. The other is Gaussian distributed with zero mean and variance σ^2, which depends on the noise level. Therefore we can detect the OFDM symbol timing by detecting the abrupt change in variance.

The probability density functions of two distributions are:

$$p(f_1(x)) = \frac{1}{2\pi} \tag{4.1}$$

$$p\big(f_2(x)\big) = \frac{1}{\sigma\sqrt{2\pi}} \exp\{-\frac{y^2}{2\sigma^2}\} \tag{4.2}$$

Where the index $f_1$holds for the beginning Tu period of an OFDM symbol and $f_2$holds for the last Tg period of the symbol. A log likelihood ratio Si is defined to be

$$S_i = \ln \frac{f_1(x_i)}{f_2(x_i)}$$

$$= \ln\left[\frac{\sqrt{2\pi}}{\sigma} exp\left\{-\frac{{x_i}^2}{2\sigma^2}\right\}\right]$$

$$= C - \frac{{x_i}^2}{2\sigma^2} \qquad (4.3)$$

And a decision function can be defined as

$$S_1^M = \frac{1}{2\sigma^2}\sum_{i=1}^{M} x_i^2 \qquad (4.4)$$

Where, $S_1^M$is negatively related to the log-likelihood ratio for the observations from x_1 to x_M .

$$f_1 \; is \; choosen, if \; S_1^M \geq h$$

$$f_2 \; is \; choosen, if \; S_1^M < h$$

Where, h is the threshold that can be determined by the variances of the two distributions and sample size M. Sliding window average is used to implement the decision function and generate output signal. So the desired symbol timing can be calculated [4][12].

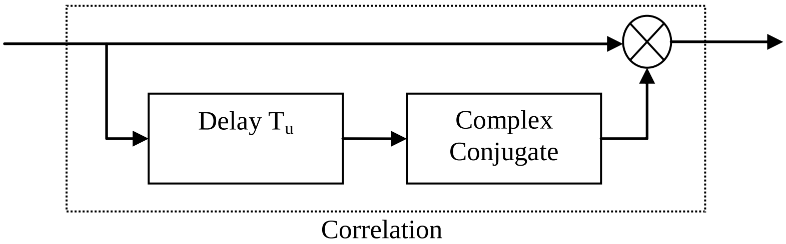

Figure 4. 2 Correlator Block

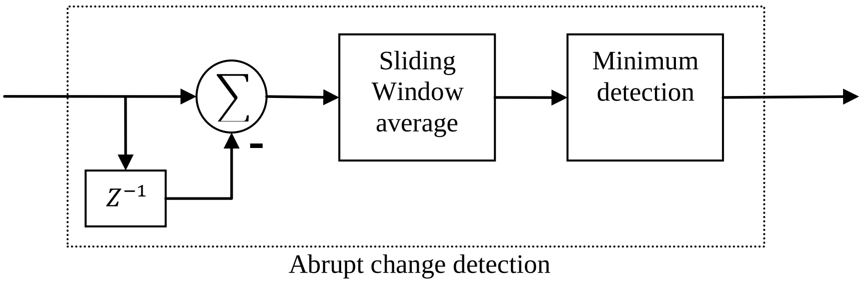

Figure 4. 3 Abrupt Phase Change Detector

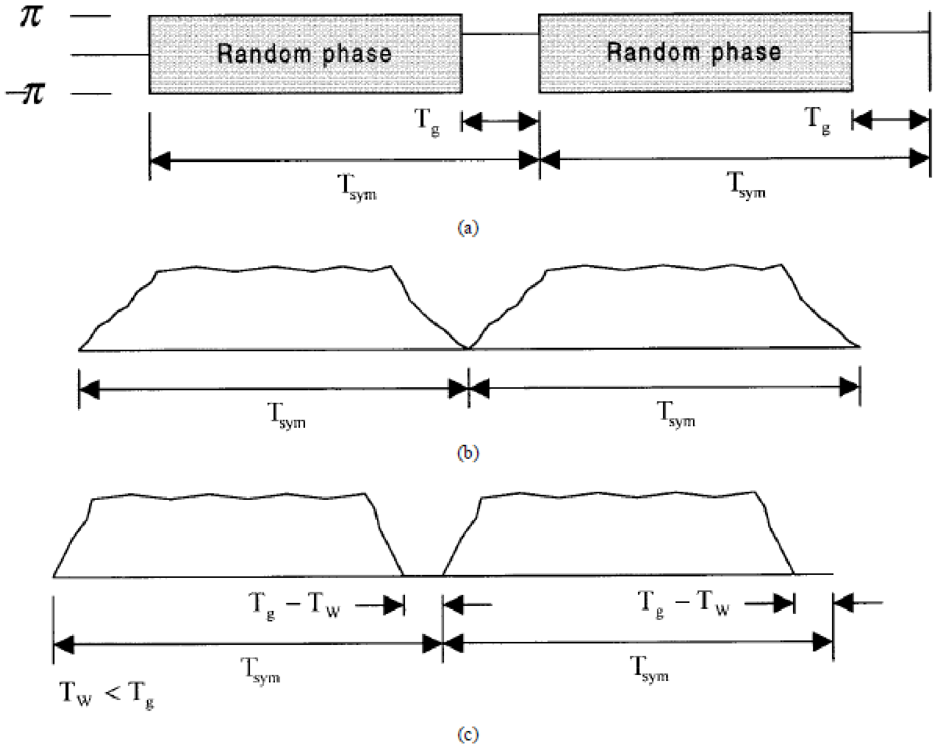

Figure 4. 4 (a) Variation in the phase of z(t) and the corresponding symbol timing. (b) Moving average output with window length Tw equal to Tg. (c) Moving average output with window length Tw less than Tg.

4.3.2 Frequency Offset Estimation

In OFDM symbols we add the guard interval, that is we add a copy of the last samples of period Tg length at the start of symbol. This property can be used to estimate frequency offset.

Consider when a received signal reaches the last Tg period of a symbol, a frequency offset exist.

We assume a frequency offset

$$\Delta f = (\delta_{in} + \delta_{fr}) f_{scs}$$

$$or\ \Delta f = \Delta f_{in} + \Delta f_{fr} \tag{4.5}$$

Where, $\delta_{in}$is an integer and $\delta_{fr}$is a fractional number less than one.

When the received signal r(t) is correlated with the shifted received frame by useful part of symbol duration Tu, the output of the correlator is given by:

z(t) = r(t))r*(t-Tu)

$$= a(t)exp\{j2\pi(\Delta f_{in} + \Delta f_{fr})t\} \times a^{*}(t - Tu)\exp\{-j2\pi(\Delta f_{in} + \Delta f_{fr})(t - Tu)\}$$

$$= C(t)\exp\left(j2\pi\Delta f_{fr}Tu\right) \qquad (4.6)$$

Where,

a(t) is the received symbol when frequency offset is zero and C(t) is defined as:

$$C(t) \equiv a(t)a^{*}(t-Tu) \approx a(t)a^{*}(t) = |a(t)|^{2} > 0 \qquad (4.7)$$

So we can estimate $fractional\ frequency\ offset\ \delta_{fr}$:

$$\delta_{fr} = \frac{\overline{\arg\left[z(nT_S)\right]}}{2\pi} \qquad (4.8)$$

4.3.3 Integral frequency offset estimation

In my simulation model, for detection of the start of the frame, I used Phase Reference symbol. If we correlate the received frame with phase reference signal, we get the peak at the start of the phase reference symbol in the transmitted frame. This peak location gives the information about the start of Phase Reference symbol, so we can roughly estimate the start of D-QPSK symbols.

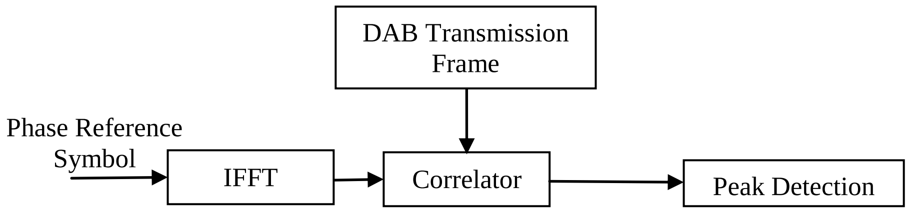

Figure 4. 5 Cross correlation of Phase Reference Symbol with Received DAB frame

4.4 OFDM Demodulator

This block demodulates the synchronized signal. Synchronized signal contains all the DQPSK modulated OFDM symbols, excluding phase reference symbol and null symbol.

This block do all the reverse operation as OFDM modulator did. This block consists of three sub-blocks, given as:

- Cyclic prefix removal
- FFT
- Zero padding removal

4.4.1 Cyclic prefix 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.

4.4.2 FFT

This block performs FFT operation to every OFDM symbol block. This transforms the OFDM symbol back to frequency domain.

4.4.3 Zero padding removal

This block remove the 128 padded zeros and re arrange the data.

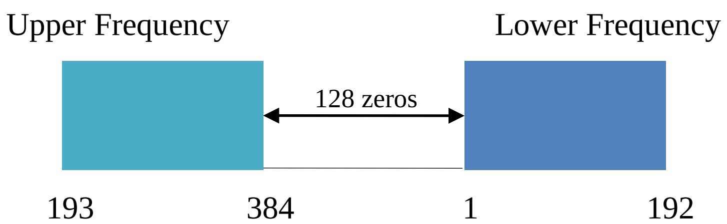

Figure 4. 6 After zero paading removal and data rearrangement

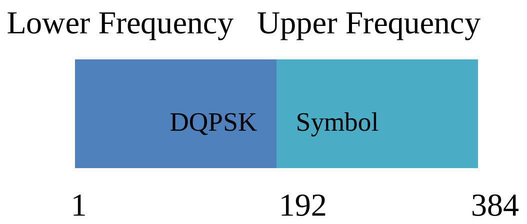

Figure 4. 7 After zero padding removal and data rearrangement

4.5 Differential demodulator

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 as:

$$y_{l,k} = z_{l,k} \times z_{l-1,k}^{*} \tag{4.9}$$

4.6 Continuous Learning Algorithm for QPSK Mapping

In DAB system once the DAB frame is synchronized, the received frame is fed to the OFDM demodulator and DQPSK demodulator. Finally the received frame fed to the Frequency interleaving and then to the SVM classifier.

In this work SVM classifier is used for QPSK classification. In the transmitter side QPSK mapping was used, in receiving side, a new continuous algorithm was used for QPSK mapping and channel estimator for DAB system. For the first DAB symbol, hard decision is used for training of the classifier. Based on hard decision function classifier estimates the labels for the first symbol and for the second symbols the estimated labels of the first symbol is used for training. This process repeats for the remaining symbols. After one time training, the classifier learns continuously hence the name is given as “Continuous learning Algorithm”

4.7 SVM

Support vector machine (SVM) is one of a method for solving classification and function estimation problems. SVM is based on the structural risk minimization principle. SVM rely on preprocessing the data to represent patterns in high dimensions-typically much higher to the original feature space. The derivation of SVM is based on constructing an optimal separating hyperplane after nonlinearly mapping the input space into a higher dimensional space. The explicit construction of this mapping is avoided by the application of Mercer’s condition. Kernels that satisfy this condition and can be employed for SVM’s are polynomials, splines, radial basis functions, and multilayer perceptrons with one hidden layer. For classification problems the parameters which are related to these kernel functions are chosen so as to minimize an upper bound on the Vapnik–Chervonenkis (VC) dimension of the SVM.

The training of SVM's with Vapnik's epsilon insensitive loss function is done by quadratic programming. The number of hidden units in the SVM is determined by the number of support vector data, which corresponds to the number of nonzero coefficients in the solution vector to the QP problem [14]-[18].

4.7.1 Linearly Separable Data

Consider the binary classification task in which we have a set of training patterns $\{x_{l,i} = 1\ldots..n\}$ assigned to one of two classes ω_1 and ω_2, with corresponding labels $y_l = \pm 1$. Denote the liner discrimination function:

$$g(x) = \omega^T x + \omega_0 \tag{4.10}$$

With decision rule

$$\omega^T x + \omega_0 \begin{cases} > 0 \\ < 0 \end{cases} \Rightarrow x \in \begin{cases} \omega_1 \; with\ corresponding\ numeric\ value\ yi = +1 \\ \omega 2 \; with\ corresponding\ numeric\ value\ yi = -1 \end{cases}$$

Thus, all training points are correctly classified if

$$y_i(\omega^T x_i + \omega_0) > 0 \; for\ all\ i \tag{4.11}$$

There are many possible separating hyperplanes. The maximal margin classifier determines the hyperplane for which the margin – the distance to two parallel hyperplanes on each side of the hyperplane A that separate the data – is the largest. The assumption is that the larger the margin, the better the generalization error of the linear classifier defines by the separating hyperplane.

A variant of perceptron rule was to introduce a margin, b>0, and seek a solution so that

$$y_i(\omega^T x_i + \omega_0) \geq b \tag{4.12}$$

The perceptron algorithm yields a solution for which all points xi are at a distance greater than b/$|\omega|$ from the separating hyperplane. A scaling of b, ω_0, ω leaves this distance unaltered and the condition still satisfied. Therefore, without loss of generality, a value b = 1 may be taken, defining what are termed the canonical hypelplanes, H1 : $\omega^T x_i + \omega_0 = +1$ and H2 : $\omega^T x_i + \omega_0 = -1$, and we have

$$\omega^T x_i + \omega_0 \geq +1 \text{ for } y_i = +1$$
$$\omega^T x_i + \omega_0 \leq -1 \text{ for } y_i = -1$$

The distance between each of these two hyperplane and the separating hyperplane g(x)= 0, is 1/$|\omega|$ and is termed the margin.

Therefore maximizing the margin means that we seek a solution to minimize $|\omega|$ subject to the constraints.

$$C1: y_i(\omega^T x_i + \omega_0) \geq 1 \; i = 1 \ldots \ldots n \tag{4.13}$$

A standard approach to optimization problems with equality and inequality constraints is the Lagrange formalism which leads to the primal form of the objective function, Lp, given by

$$Lp = \frac{1}{2}\omega^T \omega - \sum_{i=1}^{n} \alpha_i (y_i(\omega^T x_i + \omega_0) - 1) \tag{4.14}$$

Where $\{\alpha_i, i = 1 \ldots \ldots n;\ \alpha_i \geq 0\}$ are the Lagrange multipliers. The primal parameters are ω and $\omega_0$and the number of parameters is p+1, where p is the dimensionality of the feature space.

The solution to the problem of minimizing $\omega^T \omega$ subject to constraints is equivalent to determining the saddelpoint of the function Lp, at which Lp is minimized with respect to ω and ω_0 and maximized with respect to the α_i. Differentiating Lp with respect to ω and ω_0 and equating to zero yields

$$\sum_{i=1}^{n} \alpha_i y_i = 0$$

$$\omega = \sum_{i=1}^{n} \alpha_i y_i x_i \qquad (4.15)$$

Substituting in eqn gives the dual form of the Lagrangian

$$Lp = \sum_{i=1}^{n} \alpha_i - \frac{1}{2}\sum_{i=1}^{n}\sum_{j=1}^{n} \alpha_i \alpha_j y_i y_j x_i^T x_j \qquad (4.16)$$

Which is maximized with respect to the α_i subject to

$$\alpha_i \geq 0 \ \sum_{i=1}^{n} \alpha_i y_i = 0 \qquad (4.17)$$

The importance of the dual form is that it express the optimization criterion as inner product of patterns, xi. This is a key concept and has importance consequence for nonlinear support vector machines [14]-[18].

4.7.2 Classification

Recasting the constrained optimization problem in its dual form enables numerical quadratic programming solver to be employed. Once the Lagrange multiplier, α_i, have been obtained, the value of ω_0 may be found from

$$\alpha_i(y_i(x_i^T \omega + \omega_0) - 1) = 0 \qquad (4.18)$$

Using any of the support vector (pattern for which $\alpha_i \neq 0$), or an average over all support vectors

$$n_{sv}\omega_0 - \omega^T \sum_{i \epsilon SV} x_i = \sum_{i \epsilon SV} y_i \qquad (4.19)$$

Where $n_{sv}$is the number of support vectors and the summation are over the set of support vectors, SV. The solution for ω used in the above is given by

$$\omega = \sum_{i \epsilon SV} \alpha_i y_i x_i \qquad (4.20)$$

Since α_i= 0 for other patterns. Thus, the support vectors define the separating hyperplane.

A new pattern, x, is classified according to the sign of

$$\omega^T x + \omega_0$$

Substituting for ω and ω_0 gives the linear discriminant: assign x to ω_1 if

$$\sum_{i\epsilon SV} \alpha_i y_i x_i^T - \frac{1}{n_{SV}} \sum_{i\epsilon SV} \sum_{j\epsilon SV} \alpha_i y_i x_i^T x_j + \frac{1}{n_{SV}} \sum_{i\epsilon SV} y_i > 0 \qquad (4.21)$$

4.7.3 Linearly non-separable data

In many real world problems there will be no linear boundary separating the classes and the problem of searching for an optimal separating hypelplane is meaningless. Even if we were to use sophisticated feature vectors, Φ(x), to transform the data to a high-dimensional feature space in which classes are linearly separable, this would lead to an over-fitting of the data and hence poor generalization ability. We shall return to nonlinear support vector machine

However we can extend the above idea to handle non-separable data by relaxing the constraints. We do this by introducing 'slack' variables $\xi_i, i = 1, \ldots \ldots, n$, into

$$\omega^T x_i + \omega_0 \geq +1 - \xi_i \text{ for } y_i = +1$$
$$\omega^T x_i + \omega_0 \leq -1 + \xi_i \text{ for } y_i = -1$$
$$\xi_i \geq 0 \; i = 1 \ldots \ldots n$$

For a point to be misclassified by the separating hyperplane, we must have $\xi_i > 1$

A convenient way to incorporate the additional cost due to non-separability is to introduce an extra cost term to the cost function by replacing $\omega^T \omega/2$ by $\frac{\omega^T \omega}{2} + C \sum_i \xi_i$ where C is a "regularization" parameter. The term $C \sum_i \xi_i$ can be thought of as meaning some amount of misclassification- the lower the value of C, the smaller the penalty for 'outliers' and a 'softer' margin. Other penalty terms are possible, for example, $C \sum_i {\xi_i}^2$

Thus we minimize

$$\frac{\omega^T\omega}{2} + C\sum_i \xi_i$$

Subject to the constraints, the primal form of the Lagrangian now becomes

$$Lp = \frac{1}{2}\omega^T\omega + C\sum_i \xi_i - \sum_{i=1}^{n}\alpha_i(y_i(\omega^T x_i + \omega_0) - 1 + \xi_i) - \sum_{i=1}^{n} r_i\xi_i \qquad (4.22)$$

Where $\alpha_i \geq 0$ and $r_i \geq 0$ are Lagrange multipliers: r_i are introduced to ensure positivity of ξ_i.

Differentiating with respect to ω and ω_0 still results in

$$\sum_{i=1}^{n} \alpha_i y_i = 0$$

$$\omega = \sum_{i=1}^{n} \alpha_i y_i x_i \qquad (4.23)$$

And differentiating with respect to ξ_i yields

$$C - \alpha_i - r_i = 0 \qquad (4.24)$$

Substituting the results into the primal form, gives the dual form of Lagrangian

$$L_D = \sum_{i=1}^{n} \alpha_i - \frac{1}{2}\sum_{i=1}^{n}\sum_{j=1}^{n} \alpha_i\alpha_j y_i y_j x_i^T x_j \qquad (4.25)$$

This is same form as the maximal margin classifier. This maximized with respect to the α_i subject to

$$\sum_{i=1}^{n} \alpha_i y_i = 0 \qquad (4.26)$$

$$0 \leq \alpha_i \leq C$$

Thus the only change to the maximization problem is the upper bound on the α_i
The Karush-Kuhn-Tucker complimentary conditions are

$$\alpha_i(y_i(\omega^T x_i + \omega_0) - 1 + \xi_i) = 0 \quad (4.27)$$

$$r_i \xi_i = (C - \alpha_i)\xi_i = 0 \quad (4.28)$$

Patterns for which $\alpha_i > 0$ are termed the support vectors. Those satisfying $0 < \alpha_i < C$ must have $\xi_i = 0$, that is, they lie on one of the canonical hyperplanes at a distance of $1/|\omega|$ from the separating hyperplane. Non-zero slack variables can only occur when $\alpha_i = C$. In this case, the point x_i are misclassified if $\xi_i > 1$. If $\xi_i < 1$, they are classified correctly, but lie closer to the separating hyperplane than $1/|\omega|$. As in the separable case, the value of ω_0 is determined using the first condition above and any support vector or by summing over samples for which $0 < \alpha_i < C$. This gives

$$\omega_0 = \frac{1}{N_{\widetilde{SV}}}\left\{\sum_{i\epsilon\widetilde{SV}} y_i - \sum_{i\epsilon SV, j\epsilon\widetilde{SV}} \alpha_i y_i x_i^T x_j\right\} \quad (4.29)$$

Where SV is the set of support vectors with associated values of α_i satisfying $0 < \alpha_i \leq C$ and $\widetilde{SV}$ is the set of $N_{\widetilde{SV}}$ support vectors satisfying $0 < \alpha_i < C$.

4.8 One Time Training of the SVM Classifier

In this work, SVM is used for QPSK classifier and as a channel estimator. After the DQPSK modulation, the output from DQPSK modulator is fed to the SVM classifier. The classifier needs one time training to predict the correct label as shown in Figure 4.8. For the first DAB symbol, classifier was trained by the hard decision function. The QPSK mapping was done by the rule given below:

If the in-phase > 0 and quadrature phase > 0, it mapps to label 1
If the in-phase < 0 and quadrature phase > 0, it mapps to label 2
If the in-phase < 0 and quadrature phase < 0, it mapps to label 3
If the in-phase > 0 and quadrature phase < 0, it mapps to label 4

The I-Q values in the labels are mapped as,
00 is mapped to label 1,
01 is mapped to label 2,

11 is mapped to label 3,

10 is mapped to label 4,

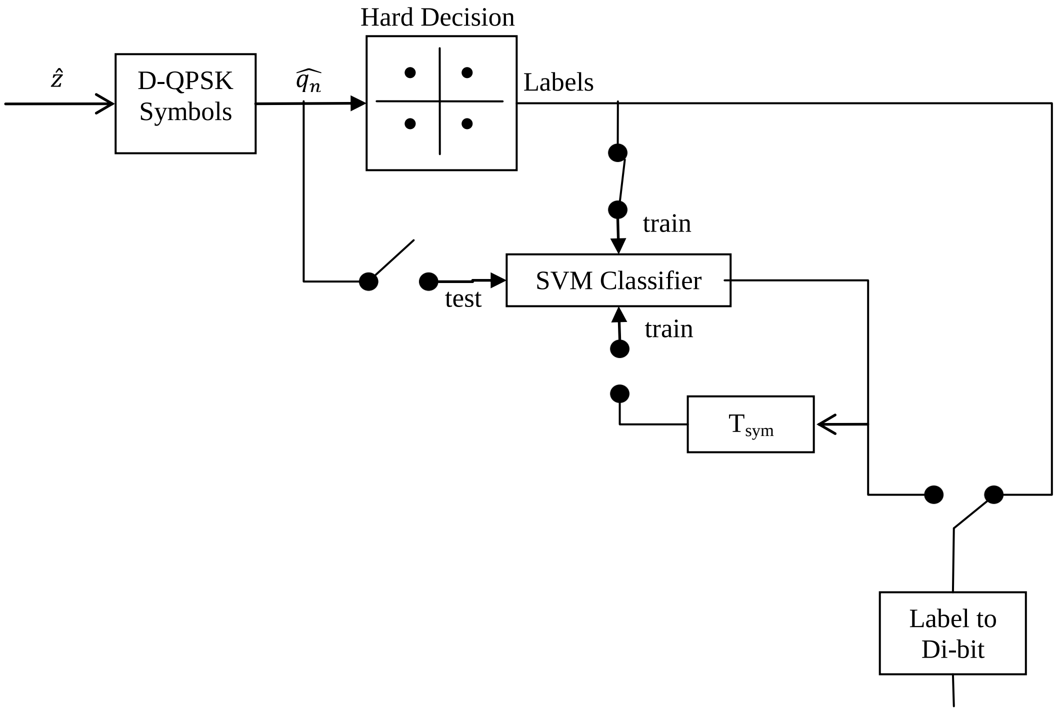

Figure 4. 8 One time training with hard decision

The output of the SVM classifier is fed to the QPSK de-mapper block, where the original signal is extracted from the labels predicted by the SVM classifier.

4.9 Continuous training and testing

Once the one time training of the classifier is done, the predicted labels of first DAB symbol is used for the training of the classifier for the second DAB symbol. The predicted labels of the second DAB symbols are used as training samples for third DAB symbols. This process repeated for the all DAB symbols of mode II. It is clear that after one time training the classifier continuously trains and tested, so the name is given “continuous learning algorithm”.

Now, the output of the SVM classifier fed to the DQPSK de-mapper. DQPSK de-mapper. So the recived information is obtained by the predicted labels after the DQPSK demapping.

The figure 4.9 shown below, gives the idea of continuous training and testing of the SVM classifier. The block label-to-d-bit is used as a DQPSK demapper.

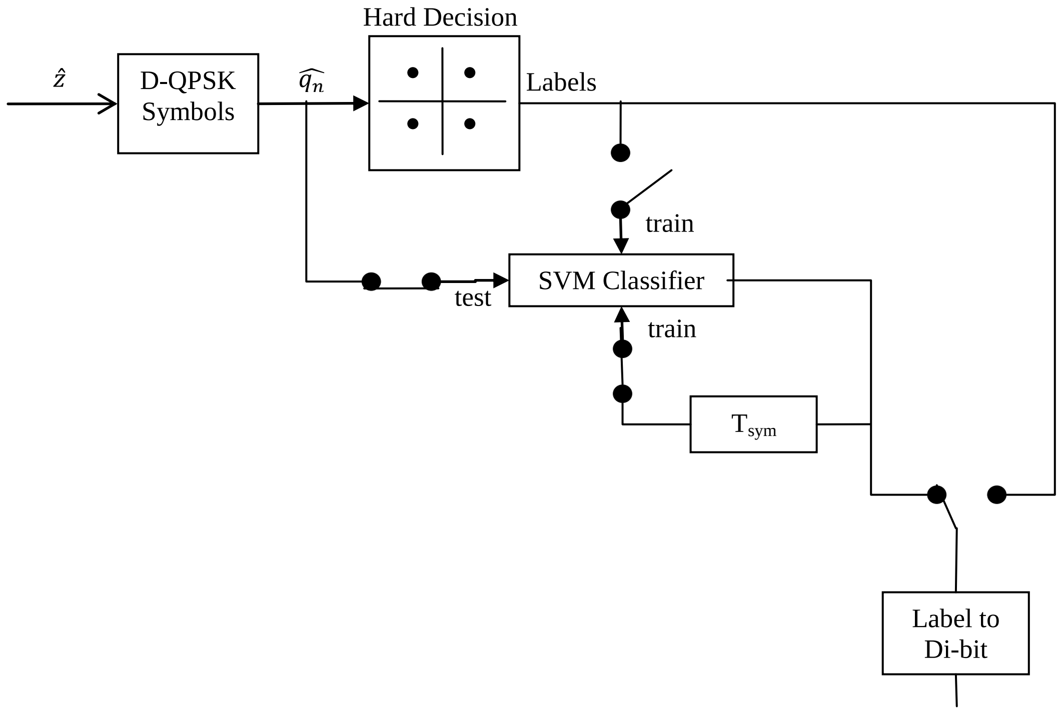

Figure 4. 9 Continuous training and testing

5.1 Introduction

This chapter presents the simulation results and performance analysis of the DAB system. In this work Bit Error Rate (BER) was used as a performance index.

All the simulation was carried out according to DAB standard defined by ETSI. Transmission mode II was used for simulation. These parameters have been presented in Table 2.2.

The simulation work was implemented using the Microsoft Windows Operating system and MATLAB 7.6.0 (R2008a).

In the simulation studies a DAB frame of length 57600 bits and number of bits in error is found at the receiver to calculate the BER for different values of SNR. Similar process is repeated for Rayleigh fading channels.

5.2 Simulation Result for AWGN channel

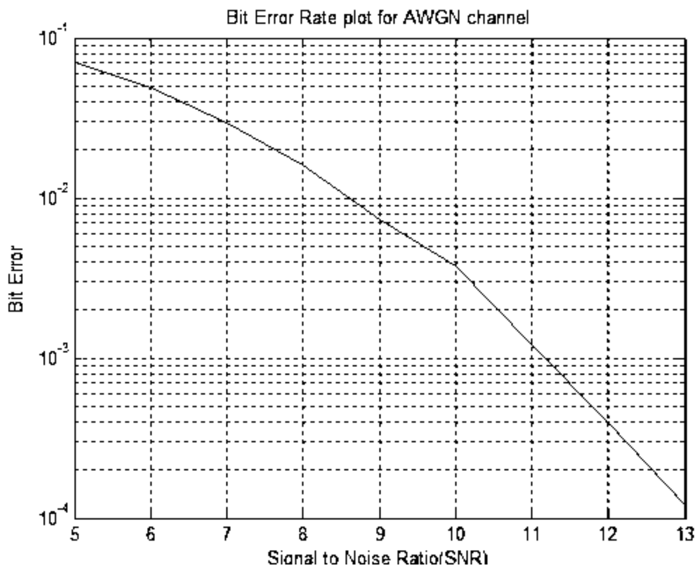

Figure 5. 1 Bit Error Rate Plot for AWGN Channel

BER performance of DAB transmitter for mode II with SVM classifier based receiver over AWGN channel is shown in figure 5.1. The figure indicates that, bit error reaching at the order of 10^{-4} at 12-13dB, which is an acceptable bit rate for DAB sytem.

5.3 Symbol Timing Synchronization

The symbol timing synchronization is carried out by cross-correlating the received frame with the phase reference symbol. In the simulated system, the received frame consists of 57600 bits including the the null symbol of length 664 bits and the phase reference symbol is of the length 638 bits including with the guard time. So at the start of the phase reference symbol in DAB received frame, peak will be detected. There is no delay for AWGN channel. So according to ETSI standard the phase reference symbol should start from the sample 665. In this simulation study, the peak was detectected exactly at sample 665. The detected peak at SNR -11dB is shown in the figure 5.2.

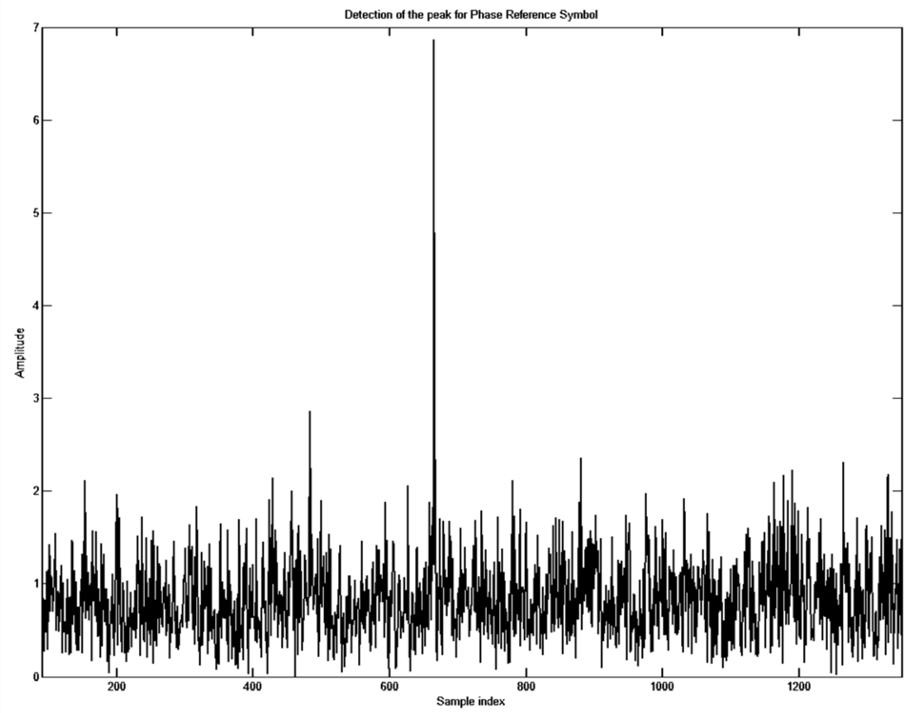

Figure 5. 2 Peak Detection at -11dB

After the symbol timing synchronization, the coarse synchronization has been simulated. Symbol timing estimation gives the idea of the start of the frame, so from the obtained simulated result, the phase reference signal is extracted from the received frame. Transmitted phase reference symbol having the length of 384 bits.After the guard time removal and zero padding removal, the extracted phase reference symbol from received frame is having the same length of 384 bits.

Now the extracted phase reference symbol from received frame is cross correlated with the transmitted received frame. The peak is obtained at the sample index of 384 shown in figure 5.3. So, this gives the idea that the frame is synchronized.

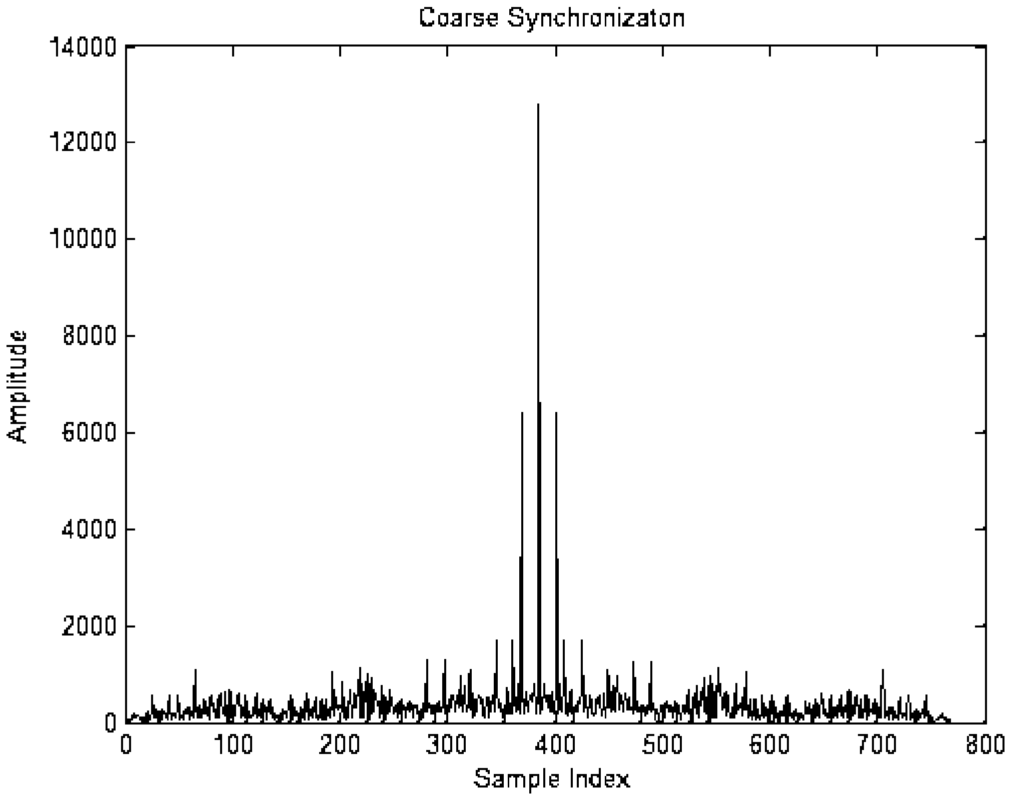

Figure 5. 3 Coarse Synchronization

5.4 Simulation Result for Rayleigh Fading Channel

As we discussed previously in section 4.8-4.9, SVM classification is used for QPSK mapping and more importantly SVM is used for channel estimator. The one time training of the SVM classifier for the first symbol is done by hard decision. Based on hard decision the first symbol of length 384 bit is mapped by the rule given blow:

If in-phase and quadrature phase > 0 it mapps to label 1.

If in-phase value <0 and quadrature phase value > 0 it mapps to label 2.

If in-phase value <0 and quadrature phase value < 0 it mapps to label 3.

If in-phase value <0 and quadrature phase value > 0 it mapps to label 4.

The original information obtained from these estimated labels.

The estimated labels from the SVM classifier are again fed to classifier as training samples for the next symbol. This process repeats for the 2 to 75 symbols. The figure 5.4 shown below is a bit error rate plot for the different Doppler shifts values in Rayleigh fading environment. Bit error rate plot is obtained by calculating the error in received bits. Total 57600 bits are transmitted in one DAB frame. From the figure 5.4, it can be concluded that SVM based classifier and channel estimator for DAB system will have a better BER performance for small Doppler shifts (Doppler shift < 70 i.e., 84 Kmph). Therefore, the present system can be of practical use in urban areas where the speed of the mobile is moderate i.e. 40-80 Kmph.

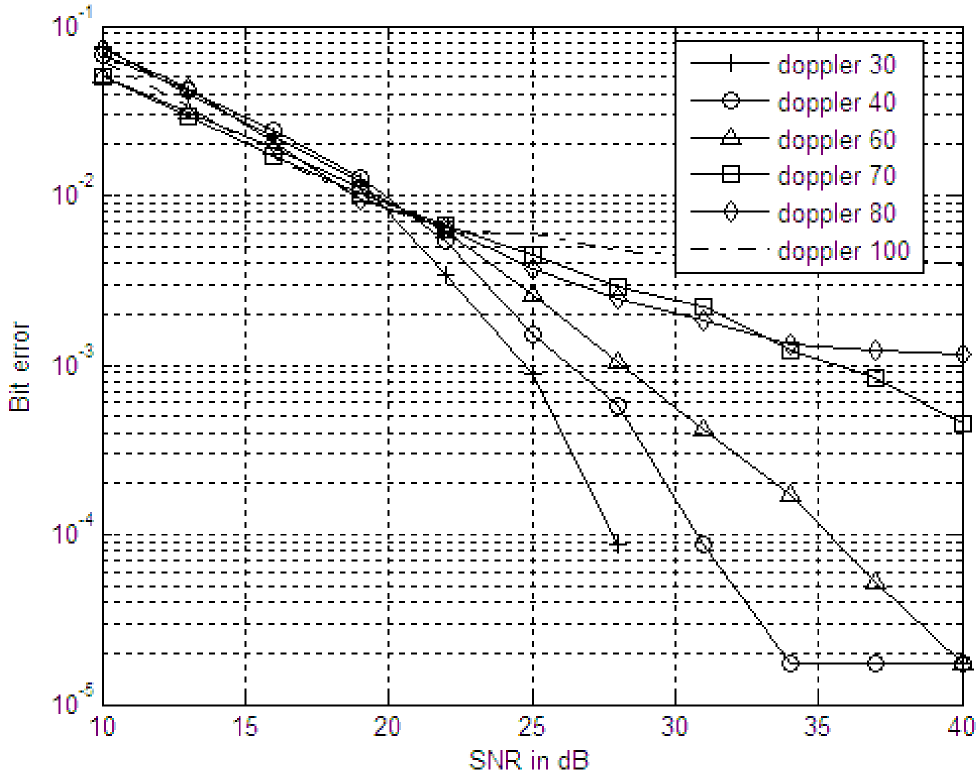

Figure 5. 4 Bit Error Rate plot for Rayleigh Fading for different Doppler Shift

5.5 Symbol Timing and Frequency Offset Estimation

As we discussed in previous section 4.3 conventional synchronization algorithm is used. The transmission signal is provided to the correlator and the output of the corrlator is input to the one-sample differentiator.

The sliding window average is taken to implement the decision function and generate output signal. So the desired symbol timing can be calculated. The window length of the guard interval Tg of 126 bits is taken. The peak will obtained at the start of each OFDM symbol shown in figure 5.5.

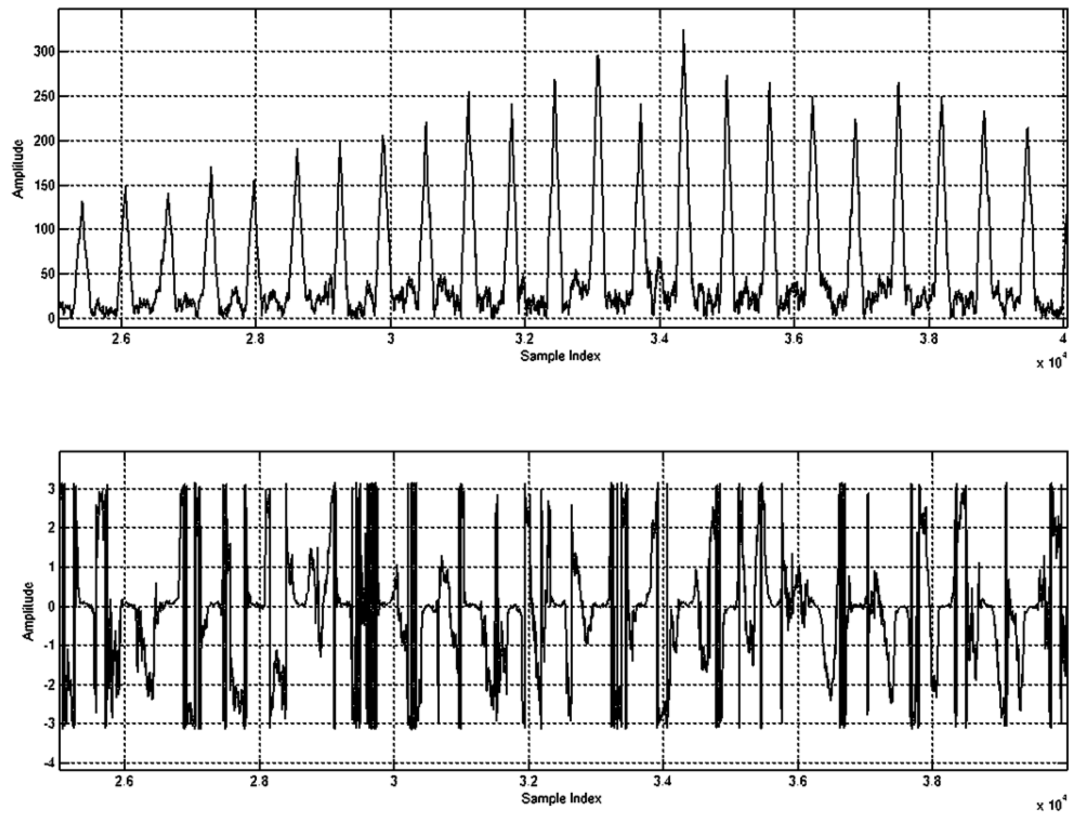

Figure 5. 5 Symbol Timing and Frequency Offset Calculation

As we discussed in section 4.3.2, in OFDM symbols the copy of the last sample of period Tg is added at the start of symbol. This property can be used to estimate frequency offset. So due to correlation, the peaks at the start of the each symbol is obtained the frequency offset is calculated by:

So we can estimate $fractional\ frequency\ offset\ \delta_{fr}$:

$$\delta_{fr} = \overline{\frac{\arg\left[z(nT_s)\right]}{2\pi}}$$

In this simulation model, flat fading is considered, so there is no offset present in the shown figure 5.5.

6.1 Conclusion Remark

This chapter gives the conclusion of the work done and also gives the some suggestions for future work

The goal of this thesis was to develop, implement and evaluate BER performance of the DAB system in MATLAB environment. 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 frequency interleaver, DQPSK modulator, and OFDM transmitter (IFFT) in the Transmitter side and in the Receiver, receiver Synchronization and a new QPSK classification algorithm is used. The transmitted signal was exposed to AWGN and Rayleigh channel. BER was used as performance criteria.

6.2 Future Work

In this system performance Analysis, I used AWGN and Rayleigh fading channel. So the system performance analysis can be done in the Rician fading channel environment.

A complete DAB signal generation should be implemented. The system should include the audio coding and channel coding part and able to generate a complete DAB signal.

In my work I used a mode II for DAB transmission; one can simulate the other mode for DAB transmission.

In the receiver part I used a hard decision for training the SVM, train the SVM classifier with better training so further improvement in BER is possible. For more accuracy we can implement the re-modulation for getting the correct training samples for SVM classifier.

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