Digital Aural Broadcast: Modulation, Transmission & Performance Analysis

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Abstract—Radio broadcasting technology has grown rapidly over the last few years due to ever increasing exaction for as high quality sound services with additional data transmission in mobile environment. In order to achieve this, association of European Broadcasting Union (EBU), the European Telecommunications Standards Institute (ETSI) and International Telecommunications Union (ITU-R) grow a completely new digital radio broadcasting technology called the Eureka-147 Digital Aural Broadcasting (DAB) system which promotes the overall broadcasting performance by delivering near CD quality sound and data services in mobile receivers along with efficient use of the available radio frequency hue cycle. Digital Aural Broadcasting (DAB) system developed within the Eureka 147 Project is a new digital radio technology for broadcasting radio stations that accommodates high-quality audio and data services to both fixed and mobile receivers. The system uses COFDM technology that combines the effect of multipath fading & ISI and makes it spectrally more active compared with avial AM/FM systems. In this project we will show the performance analysis of Eureka-147 DAB i.e digital aural broadcasting system. DAB transmission mode-II is achieve first and then extended successfully to other modes. A frame-based processing will be used in this study. Performance studies for AWGN, Rayleigh and Rician channels will be attended. For all studies BER will be used as performance basis. In this project we will 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.

Keywords—aural, accommodate, exaction, hue cycle, supersede.

I. INTRODUCTION

Radio broadcasting is one of the most widespread electronic assemblage media comprising of hundreds of programme providers, thousands of HF transmitters and billions of radio receivers worldwide. Since the broadcasting commence in the early 1920s, the market was widely enveloped by the AM services. Today with the conceive 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 Aural Broadcasting (DAB) has the capability to supersede the avial AM and FM aural 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 suitable for mobile receivers and provides very high forbearance against multipath reception and inter symbol interference (ISI). It allows use of single hue cycle (frequency) networks (SFN) for high frequency efficiency. In several countries in Europe and overseas, broadcasting organizations, network providers and receiver fabricators are already implementing digital broadcasting services using the DAB system. Emotive 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].

II. THE SIMULATION MODEL

A. Introduction

This section describes the detailed method for prototyping of the DAB transmission and receiving system factually in conformance with the ETSI DAB standard [1] as shown in in previous chapter. Transmission mode –II has been used in the simulation so all the standard parameters of this mode has been chosen. After flourish 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 complementary environment that allow programmer to achieve computationally accelerate tasks faster than any other programming languages. It is well suitable for design and investigation of complete digital communication systems. All the simulation work has been evolved in the baseband transmission and frame based processing is used. The simulation results are present only for transmitted mode –II. Before simulating the essential DAB system some basic simulation has been presented for achievement of BPSK, QPSK and QAM in AWGN & Rayleigh fading channels. Therefore the after effect from this simulation are first shown in chapter 4. The DAB system was accomplished and simulated without MPEG audio coding, conglomerating, time interleaving, and ADC/DAC and up/down converter. DAB Simulation Model

Figure 3.1 presents the unimpaired catalog of the DAB system prototype which was simulated in MATLAB environment.

Figure 1: Complete block diagram of DAB system for Simulation [1] [4].
B. Information source

As we have seen in the block diagram of digital aural broadcasting system that information source is the first block in the transmitter region. The work of the information source is that it generate the random binary data bit sequence for FIC and MSC. So we can calculate the data for one transmission frame. Which is given by the below formula:

\[
\text{DATA-bits} = \text{FIC-DATA} + \text{MSC-DATA}
\]

As we know that for mode-II that each transmission frame has 16 OFDM symbols.

Each FIC contains 3 FIBS and for the transmission of 24ms frame there is only one CIF is present. For transmission mode-II the total number of sub carriers are 384. Thus the QPSK mapping is done by the below formula:

\[
\text{FIC\_DATA} = \text{No. of OFDM symbols} \times \text{bits/OFDM symbol} \Rightarrow 3 \times 768 = 2304 \text{ bits.}
\]

\[
\text{MSC\_DATA} = \text{No. of OFDM symbols} \times \text{bits/OFDM symbol} \Rightarrow 72 \times 768 = 55296 \text{ bits.}
\]

By using equation (3.1) we can calculate the total data bits for each transmission frame which is 57600 bits. There is a MATLAB function which can be used to generate the random data bits for each transmission which is known as “randint”. There is another version of this MATLAB function which can be used to generate the M X M matrix of random binary numbers of “0” and “1” with equal probability. These version of MATLAB function is known as “randint (m)”.

C. Convolutional Encoder

As we have seen from the DAB block diagram that output data stream of information source is the input of the convolution encoder. In this channel coding is based on the punctured convolutional FEC. As we know that the punctured convolutional forward error correction allows both equal and unequal error protection. In dab system we have used the convolutional encoder which has constraint length 7. It also have the octal kind of generator polynomials (133, 27, 171, 145). The expand rate of this encoder is 1/R which means that each input bit is protected or reserved by the 4 bits.

We are not going to use any MATLAB code for this block. For hard decision decoding and soft decision decoding it uses the Viterbi algorithm. The output of convolutional encoder is coded bit stream having gross data rate of 230400 bps. Which has the mother rate 1/4.

1) Puncturing

Basically puncturing and encoding has the same meaning. It provide the error correction code with higher rate and less redundancy. Without increasing the system complexity it can increase the system flexibility.

a) Puncturing of FIC

The puncturing procedure for slow information channel is variable and for fast information channel is fixed. For the FIC or fast information channel it provides high protection level. According to the DAB standards there is 24 puncturing vectors. For audio bit rate of 32 kbit/s the convolutional encoder can be split into 24 consecutive blocks (128 bits). The last 3 blocks will be punctured according to the puncturing index PI=15 and the remaining 21 blocks will be punctured according to the puncturing index PI=16. This gives the mother rate of 1/3, now the puncturing vector of 24 bits can be given by:

\[
\text{VT} = (1100 1100 1100 1100 1100 1100)
\]

The output of 12 bits are called tail bits, all of the block combined together and the tail bits add to the last block. The output codeword is called the punctured codeword which has 2304 bits. Agering the zero padding bits will be added at the end of punctured codeword to complete the word length of 64 bits.

For VPI = 0, the bit will not be transmitted.

For VPI = 1, bit will be transmitted.

b) Puncturing of MSC

As we know that there are five kind of protection level for different audio bit rates. Which can be defined as PL1 PL2 PL3 PL4 PL5. So the puncturing of MSC is defined in these protection level. Table 1 shows the five protection levels at the rate of 32kbit/s.

TABLE I: Five protection levels for audio rate of 32 kbit/s [1].

<table>
<thead>
<tr>
<th>Audio bit rate (kbit/sec)</th>
<th>p</th>
<th>L1</th>
<th>L2</th>
<th>L3</th>
<th>L4</th>
<th>P11P12P13P14</th>
<th>Number of padding bits</th>
</tr>
</thead>
<tbody>
<tr>
<td>32</td>
<td>5</td>
<td>3</td>
<td>4</td>
<td>17</td>
<td>0</td>
<td>5</td>
<td>3 2</td>
</tr>
<tr>
<td>32</td>
<td>4</td>
<td>3</td>
<td>3</td>
<td>18</td>
<td>0</td>
<td>11</td>
<td>6 5</td>
</tr>
<tr>
<td>32</td>
<td>3</td>
<td>3</td>
<td>4</td>
<td>14</td>
<td>0</td>
<td>15</td>
<td>9 6 8</td>
</tr>
<tr>
<td>32</td>
<td>2</td>
<td>3</td>
<td>4</td>
<td>14</td>
<td>3</td>
<td>22</td>
<td>13 8 13</td>
</tr>
<tr>
<td>32</td>
<td>1</td>
<td>3</td>
<td>5</td>
<td>13</td>
<td>3</td>
<td>24</td>
<td>17 12 17</td>
</tr>
</tbody>
</table>

2) Concatenated coding

The error free channel can be formed by this concatenated coding method. In this coding we can use the inner code with outer code or convolutional code with the block code. For different transmission channel we can improve the performance in the form of bit error rate i.e BER by the use of concatenated coding techniques. The basic block diagram of concatenated coding technique is shown in below figure.

In the above block diagram of concatenated coding the place of inner code the DAB system uses soft decision convolutional Viterbi decoded code not the block code. Similarly on the place of outer code it uses the hard decision
block code which is the reed Solomon with 8-bit symbols. This DAB system also uses the linear block code, cyclic code, hamming code and BCH code on the place of outer encoder. For the linear block code, cyclic code and hamming code the MATLAB function “encode” is used and for BCH codes the MATLAB function “bchenc” is used. For the code word length n=511 is used and for message length k=502 is used which gives the error correction capability which is t=8.

D. Data Mapping

Data mapping is the another part of the simulation model of DAB system which has 3 different task. Which is as follows:

1) Block partitioning

Block partitioning is the transmission mode dependent operation whose work is to divide the convolutional code bits into the blocks of data by its input which is associated to OFDM symbols. According to this the convolutional codeword (57600) is divided into 75 consecutive blocks (768 bits). This procedure is shown in following figure.

![Figure 3: Method of Block partitioning](image)

2) QPSK mapping

This QPSK mapping sets the serial bit stream according to QPSK modulation scheme. The QPSK modulation scheme or mapping is shown according to DAB standards:

$$Q_l,n = \frac{1}{\sqrt{2}}[(1-2b_l,n)+j(b_1n+k)]$$

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

For one OFDM symbols each data block of size 768 bits is mapped onto the another 384 complex coefficients. The first 384 bits represent the real part of QPSK symbols and the another 384 bits represent the imaginary part of QPSK symbols which encoded 2 bits/symbol. So the output of this block is complex which consists 75 blocks of 384 bits. Below figure represent the QPSK constellation plot contaminated with AWGN noise with 20 dB SNR. We can also see that symbols are distorted.

![Figure 3 QPSK constellation diagram and QPSK constellation diagram with AWGN noise.](image)

Frequency interleaving Basically frequency interleaving is used to remove the fading effect. It also balance deep fades that occur in wireless channel by spreading data bits over sub carrier channel.

Frequency interleaving characterize the connection 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-arranged according to the following relation:

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

where F =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 which can be obtained from the following relation: $\pi(0) = 0$; for i = 1, 2, …..511. $dn = \pi(i)$ (excluding 256. The frequency interleaving law between QPSK symbols and carrier index is defined as $k = F(n) = dn - 256$

The interleaving law is illustrated in Table below:

<table>
<thead>
<tr>
<th>i</th>
<th>$\pi(i)$</th>
<th>$dn$</th>
<th>$F(n)$</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>0</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>1</td>
<td>127</td>
<td>127</td>
<td>0</td>
</tr>
<tr>
<td>2</td>
<td>242</td>
<td>142</td>
<td>-83</td>
</tr>
<tr>
<td>3</td>
<td>201</td>
<td>202</td>
<td>-99</td>
</tr>
<tr>
<td>4</td>
<td>180</td>
<td>180</td>
<td>-78</td>
</tr>
<tr>
<td>5</td>
<td>410</td>
<td>140</td>
<td>-105</td>
</tr>
<tr>
<td>6</td>
<td>484</td>
<td>256</td>
<td>-35</td>
</tr>
<tr>
<td>7</td>
<td>307</td>
<td>207</td>
<td>141</td>
</tr>
<tr>
<td>8</td>
<td>168</td>
<td>168</td>
<td>65</td>
</tr>
<tr>
<td>9</td>
<td>283</td>
<td>283</td>
<td>7</td>
</tr>
<tr>
<td>10</td>
<td>474</td>
<td>474</td>
<td>-111</td>
</tr>
<tr>
<td>11</td>
<td>145</td>
<td>145</td>
<td>8</td>
</tr>
<tr>
<td>12</td>
<td>476</td>
<td>476</td>
<td>-111</td>
</tr>
<tr>
<td>13</td>
<td>171</td>
<td>171</td>
<td>0</td>
</tr>
<tr>
<td>14</td>
<td>392</td>
<td>392</td>
<td>8</td>
</tr>
<tr>
<td>15</td>
<td>492</td>
<td>492</td>
<td>8</td>
</tr>
<tr>
<td>16</td>
<td>80</td>
<td>80</td>
<td>11</td>
</tr>
<tr>
<td>17</td>
<td>113</td>
<td>113</td>
<td>12</td>
</tr>
<tr>
<td>18</td>
<td>450</td>
<td>450</td>
<td>0</td>
</tr>
</tbody>
</table>

![Table 2: Frequency Interleaving Rule for Transmission Mode-II](image)

E. Phase Reference Symbol Generator

The first OFDM symbol in the transmission frame except the null symbol is known as the phase reference symbol. It helps in the synchronization of receiver. The detection of this symbol can be used for frame synchronization. In the transmission frame it can also be used as reference for differential modulation.

The phase reference symbol can be expressed as:

$$Z_{nk} = e^{j\Psi k}$$

for $-K/2 < k < 0$ and $0 < k < K/2$

$$= 0$$

for $k=0$

Where

$$\Psi_k = \frac{1}{2}(h_{nk} + n)$$
The value of parameter $i,k,n$ are given according to the function of carrier index $k$ for all the DAB transmission modes. The values of the parameter $h_{i,j}$ is set according to the function of its indices $i$ and $j$. The simulated output for real part of the phase reference symbol which looks like a noise signal is shown in following figure.

![Real part of the Phase reference symbol and Phase reference symbol constellation diagram](image)

Figure 5: Real part of the Phase reference symbol and Phase reference symbol constellation diagram

**F. Differential Modulation**

In mobile communications the multipath effect can humiliate the phase of the carriers. The solution to this difficulty is to transmit the information as the difference between the phases of two symbols. This is achieved by this block which performs $\pi/4$ shifted differential QPSK modulation. According to this modulation scheme there is no absolute phase reference between symbols, each symbol is referenced only against the previous symbol, which simplifies the decoder. Apart from being bandwidth efficient the additional $\pi/4$ phase shift resolves phase vagueness of ordinary D-QPSK. Differential modulation is applied to QPSK symbols on each carrier which is by:

$$z_{l,k} = z_{l-1,k} \cdot y_{l,k} \ldots \text{where } l=2,3,4,\ldots,L \text{ and } -K/2 \leq k \leq K/2$$

$z=$ complex D-QPSK symbol and $y =$ input QPSK symbol.

**G. OFDM Symbol Generator**

OFDM symbol generator is the heart of the DAB system. The OFDM technology makes it robust against the multipath fading environment which delivers high quality audio services. The OFDM symbol generator can be explained in following sub-sections:

Zero padding

If the number of carrier has power of 2 then IFFT/FFT algorithm works efficiently. The D-QPSK symbol output has length of 384. The FFT length for transmission mode-II is 512 so zero padding is compulsory to make it power of 2. The zero padding adds 128 zeros to each D-QPSK symbol block to work with 512 FFT length which illustrated in below figure.

![D-QPSK symbol block before and after zero padding and rearrangement](image)

Figure 6: D-QPSK symbol block before and after zero padding and rearrangement.

1) **IFFT operation**

As we know that OFDM symbol generator is the heart of DAB technology in similar way the IFFT operation is the heart of OFDM technology. It can perform 512 IFFT operation for each block after the zero padding. Thus the frequency domain changed into the time domain sample. For this operation the MATLAB function “ifft” is used.

2) **Guard time insertion**

For the inter symbol interference this block provides the OFDM symbol resistant. It takes copy of last 126 samples which is equals to guard interval 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 which is equivalent to OFDM symbol duration $T_S$.

**H. Null Symbol Generator and final DAB frame**

The last block of DAB transmitter is null symbol generator by adding this the final DAB frame structure is completed. It has the duration of $T_{NULL}$ which is equivalent to 664 samples. So we can say that the 664 zeros are generated by using “zeros” which is a MATLAB function and added at the beginning of the frame. During the transmission of or the working of NULL symbol no other information is transmitted.

![Simulated DAB frame in time domain](image)

Figure 7: Simulated DAB frame in time domain.

**I. Channel**

As we know that channel is the physical transmission medium by which the final DAB signal is passed for BER performance analysis. The final DAB signal is generated by the transmission section. This can be achieved by selecting AWGN channel, Rayleigh fading channel or Rician channel. This channel can be selected from the channel library of MATLAB communication toolbox. The mobile radio channel is characterized by time variance and frequency selectivity. The Doppler frequency shift is the most important parameter for mobile channels, because the 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]:

\[ F_d = \left( \frac{v}{c} \right) f_0 = \left( \frac{1}{1080} \right) f_s/MHz \] Hz

Where \( f_0 \) = transmission frequency and \( v \) = speed of the vehicle. The following Figure shows the received VHF signal level for a car moving at a speed of 192 km/hr as a function of time for a carrier frequency of 225 MHz.

![Figure 8: Time variance due to multipath channel [2].](image)

Below Table 3 shows the Doppler frequencies for different vehicle speed and carrier frequency.

**TABLE 3: DOPPLER FREQUENCIES FOR DIFFERENT VEHICLE SPEEDS [2]**

<table>
<thead>
<tr>
<th>( F_d ) max</th>
<th>( V = 48 ) km/hr</th>
<th>( V = 96 ) km/hr</th>
<th>( V = 192 ) km/hr</th>
</tr>
</thead>
<tbody>
<tr>
<td>( f_0 = 225 ) MHz</td>
<td>10 Hz</td>
<td>20 Hz</td>
<td>40 Hz</td>
</tr>
<tr>
<td>( f_0 = 900 ) MHz</td>
<td>40 Hz</td>
<td>80 Hz</td>
<td>160 Hz</td>
</tr>
<tr>
<td>( f_0 = 1500 ) MHz</td>
<td>67 Hz</td>
<td>133 Hz</td>
<td>267 Hz</td>
</tr>
</tbody>
</table>

J. Spectrum Characteristics

The total PSD is the sum of power spectral densities of each carrier. The bandwidth of DAB signal is 1.536 MHz, therefore any signal component outside the nominal bandwidth can be removed by suitable filtering. Below figures shows the simulated theoretical DAB signal spectrum for all the four transmission modes.

![Figure 9: DAB signal spectrum for TM-I; DAB signal spectrum for TM-II; DAB signal spectrum for TM-III; DAB signal spectrum for TM-IV.](image)

The practical values of centre frequency \( F_c \) may be evaluated from above figures that all the above four transmission modes have a bandwidth of 1.536 MHz.

SNR (signal-to-noise ratio) was taken to be 15 dB. The bandwidth of the signal can be represented in baseband mode. According to figure (3.1) after the addition of null symbol the final DAB frame was used to obtain the spectrum of the transmitted signal and the output of the channel before synchronization was used to obtain the received signal spectrum.

![Figure 10: Simulated transmitted signal spectrum; Simulated received signal spectrum in AWGN channel; Simulated received signal spectrum in Rayleigh fading channel; Simulated received signal spectrum in Rician channel.](image)

According to Figure 10 the received signal spectrum in AWGN channel has approximately the same power level as transmitted signal. Figure 10 represent the received DAB signal spectrum (mode-II) in Rayleigh fading channel. As shown in Figure 10 above the power level of received signal spectrum in Rayleigh fading channel is 2 dB which is less than the transmitted signal. Figure represents the received DAB signal spectrum (mode-II) in Rician channel. As shown in Figure 10 the power level of received signal spectrum in Rician channel is 5 dB which is less than the transmitted signal. So network gain is less in the case of Rician channel.

K. Receiver Synchronization

The DAB receiver will be designed exactly opposite way of the task performed for the transmitter. In order to produce the original information at the receiver side all digital communication systems require proper synchronization. The synchronization block is used to locate accurately for each DAB frame, so that the demodulation can be performed frame by frame or symbol by symbol. In DAB system the Null symbol and the phase reference symbol used for synchronization purpose. Below figure 11 illustrates the process of receiver synchronization.

![Figure 11: Block diagram of Symbol and Frame synchronization [13] [4].](image)

1) Fine time synchronization

It is also known as symbol timing synchronization. This type of synchronization is performed by calculating the Channel Impulse Response (CIR) which is based on the actually received time frequency phase reference symbol (PRS) and the specified PRS stored in the receiver. To estimate the channel impulse response, training Sequences (PRS in case of DAB system) are used. This means that a part or the whole transmitted signal is known from the receiver. As the receiver aware of the signal it is supposed to be observed, it can
It is based on the phase reference symbol. The phase reference symbol is the dedicated pilot symbol in each DAB transmission frame. Since the modulation each carrier is known, multiplication of received PRS with complex conjugate the imaginary or the real part 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 which can be obtained by an IFFT operation of the resultant product as explained in following formula:

\[
CIR = \text{IFFT}(\text{Received PRS} \cdot \text{PRS}^*)
\]

Where PRS* = complex conjugate of the phase reference symbol. The peak of the impulse signal which is obtained from the equation will give location of the beginning of the PRS as 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 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 eliminate and data rearrangement has to be done. The resulting sample block is of size 384 same as PRS. Now sample block is multiplied by the complex conjugate of the PRS which is known at the receiver and then transformed into impulse signal in time by performing IFFT operation on the product.

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

The threshold level can be determined by observing the magnitude 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 illustrated the phase reference impulse symbol in presence and absence of noise.

From the above figure it can be observed that the highest peak is located at sample index 791. As we know 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, so that the sum of null symbol and guard interval samples equals 790, therefore the peak is located exactly at the beginning point of useful phase reference symbol. The useful OFDM symbol duration TU does not include the guard time interval.

2) Coarse time synchronization

It gives the irregular frame timing by envelope detection of the received signal that means detecting the null symbol by comparing average signal power during null symbol period TNULL with a set threshold. From the received signal, a data block of size 664 which is equal to TNULL samples is taken to calculate the average signal power. In coarse time synchronization we can use the NULL symbol detection method. According to this method the average transmitted signal power is greater than the half of average signal power, the NULL symbol has been detected. This method is not applicable on low SNR. Because high noise power provide incorrect frame timing. Thus phase reference symbol detection is well suitable for correct symbol timing and frame timing.

L OFDM Symbol Demodulator

This block demodulates the OFDM symbols from the synchronized DAB frame. The demodulation process is done by removing the null symbol and the phase reference symbol. The OFDM symbol demodulator is explained in the following sub-sections.

1) Guard time removal

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

2) FFT operation

After guard time removal the OFDM symbols has a length of 512 is equal to the FFT length. This block performs the FFT operation on each OFDM symbol obtained from guard time removal. Thus we get back frequency domain samples after FFT operation.

3) Zero padding removal

The zero padding removes the 128 zeros that was padded to each D-QPSK symbol block in the transmitter and after that it
reread the data in proper form which is illustrated in the Figure 3.24.

M. Differential Demodulation

It 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.

\[ y_{l,k} = z_{l,k} \times z_{l,k}^* \]  

where \( y_{l,k} \) is the output of this block, \( z_{l,k} \) is the received D-QPSK symbol from the OFDM symbol demodulator and \( z_{l,k}^* \) is the complex conjugate of the differential phase reference.

N. Data De-mapping

After the differential demodulation the received QPSK symbols transformed back into bits by data de-mapping. Data de-mapping further divided into two sub-blocks which as follows.

1) Frequency de-interleaving

This operation is performed at the transmitter to abolish the transmission disturbance such as selective fade. The frequency de-interleaving performs the opposite operation of frequency interleaving by re-arranging bits to get QPSK symbol block identical with the output of QPSK mapping.

2) QPSK de-mapping

This sub-block changes the received complex QPSK symbols from the frequency de-interleave output into bits. When the real part of the complex QPSK symbol is negative, the decoded bit is “1” and when it is positive the bit is “0”. This rule is also applicable to imaginary part of the complex QPSK symbol. The decoded bits should be set 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.

O. Viterbi Decoder

To decrease the transmission errors due to channel impairments the DAB system at the transmitter employed powerful rate compatible punctured convolutional code (RCPC) with constraint length 7 and mother code rate of 1/4 for channel coding. This mother code was punctured with different puncturing vectors to obtain a wide range of possible code rates to acclimatize to the channel characteristics. For decoding these codes the Viterbi algorithm will be used, which offers best performance according to the most likelihood criteria. The input to the Viterbi decoder will be hard-decided bits i.e “0” or “1”, which is also known as a hard decision. No MATLAB code has been written for this block instead the function „vitdec” is used for Viterbi decoding purpose.

1) De-puncturing

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

2) Concatenated decoding

As explained in section concatenated coding of convolutional codes, the concept of concatenated coding has been applied for improving the BER performance of the DAB system. This was achieved 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 MATLAB function “bchdec” for BCH code.

III. SIMULATION RESULTS AND DISCUSSION

A. Introduction

This chapter presents simulation results for AWGN channel, Rayleigh fading channel and Rice channel along with the bit error rate (BER) analysis.

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 analysed. Before simulating the DAB system some basic simulations had been conducted for BPSK, QPSK and QAM modulation in AWGN & Rayleigh fading channels.

B. 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.
As can be seen from Figure 14 and 15 that both theoretical and practical BER are in good agreement with each other, respectively for BPSK and QPSK modulation in AWGN and Rayleigh fading channel.

Figure 15: BER performance for QPSK modulation in AWGN & Rayleigh fading channel

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

Figure 17: BER performance of 32-QAM in AWGN & Rayleigh fading channel

Figure 16 shows that for 16-QAM modulation in AWGN channel, practical 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 practical 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 practical 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 practical BER needs an additional transmitted signal power of 16 dB compared to theoretical BER. It is also concluded that as SNR increases, bit error rate reduces and the use of channel coding could develop the BER performance.

Figure 18: BER performance of OFDM using BPSK modulation in Rayleigh fading channel

Figure 19 shows that to attained 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 for this simulation.

C. Simulation Results for DAB mode-II in AWGN channel

After performing basic simulation results 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 obtained. First of all the precision of the DAB simulation model given in Figure 3.1 will be tested. The simulation parameters have been taken according to the DAB standard for the mode-II. The probability of bit error for π/4 D-QPSK is given by:

Theoretical BER for π/4 D-QPSK in AWGN: 1/2erfc(√0.5858x(Eb/No))

Figure 20 represents the system performance.
It can be seen from Figure 4.7 that both practical and theoretical BER plots are same and almost overlapped each other. This substantiated that the DAB system model simulated is perfectly implemented. The result also indicates that to attained a BER of 10^-4 theoretical π/4 D-QPSK needs an additional SNR of 4.3 dB in comparison with the theoretical BPSK.

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

From the above Figure 21 it can be evaluated that the highest peak is located at sample index of 791 and the worst SNR at -11 dB providing accurate fine time synchronization. From the simulation model 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 the total sum of null symbol and guard interval samples equals 790, therefore the peak is located exactly at the beginning point of useful phase reference symbol.

The performance of DAB system with FEC coding will be analyzed subsequently. 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 22 represents the result for the DAB system with FEC coding.

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

After verifying perfection in the BER performance using channel coding, the performance with and without frequency interleaving will be examined next. The performance results are shown in Figure 23.

Figure 3 clearly shows that interleaving is essential for the channel coding to function appropriately. Also this compensates any deep fades that may occur in the wireless channel. Interleaving extends the data bits over the sub-carriers. The wireless channel as a wideband channel and rarely come across a flat, consistent response across the entire spectrum. As deep fades influence more than one sub-carrier channel, a block data is affected. By spreading adjoining bit across the channels, the bits have been re-arranged in their proper order. The effect of fade is decreased.

The performance output for different coding rates is represented next. A scrupulous coding rate is implemented by a consequent puncturing vector. Figure 4.11 shows the result for DAB mode-II applied with different coding lengths 8/11, 8/12, 8/16, 8/24 and 8/32.
Figure 24 discloses 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 raises and good high rate convolutional codes are generated from low rate mother code.

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

From Figure 26 and 27 we can see 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 to 7 start to show improvement at BER of 10^-4. But for t=8 a coding gain of about 0.5 dB which can be 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 examined 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].

After analyzing 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 511 and Message length was 493 for error correcting capability of two and 511 and 439 for error correcting capability of eight as simulation parameters. Figure 26 and 27 presents the results for BCH coding.
been applied. Quantization levels is the important parameter for soft decision decoding. It can be seen that for different Quantization levels the performance fluctuate greatly. It is finally concluded that use of adaptive Quantization levels would give a superior performance.

According to the DAB standards there are five protection levels for encoding of the MSC but coding of FIC is fixed. For mobile reception protection level 1 is considered to be most efficient among other profiles. Therefore performance of DAB system using Protection level 1 is examined next. The performance results are presented in Figure 29.

According to the DAB standards there are five protection levels for encoding of the MSC but coding of FIC is fixed. For mobile reception protection level 1 is considered to be most efficient among other profiles. Therefore performance of DAB system using Protection level 1 is examined next. The performance results are presented in Figure 29.

Figure 29: BER performance using Protection level 1 in AWGN channel.

At the cost of higher SNR value the protection level 1 makes the system resistant to channel impairments which is shown in figure 29.

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

The performance analysis in Rayleigh fading channel will be considered after the investigation of BER performance of DAB mode-II in AWGN channel. Firstly the performance of the system using different coding rates was considered. Figure 4.17 represents the result for DAB mode-II applied with different coding lengths 8/11, 8/12, 8/16, 8/24 and 8/32.

Figure 30: BER performance with different coding rates in a fading channel.

It may be observed that for flat frequency Rayleigh channel without Doppler shift in Figure 4.17 utilizing punctured FEC with puncturing prototype PI=16 (code rate 8/24) needs SNR of 7 dB to get a BER of 10^-4. The equivalent system with Doppler shift of 20 Hz requires a SNR of 4.5 dB for the same BER performance. This explained that puncturing improves the system performance in different transmission channels.

Figure 31: BER performance with and without interleaving in a fading channel.

Figure 31 undoubtedly shows that interleaving is essential for the channel coding to function appropriately. Also this removes any deep fades that may occur in the wireless channel. Yet again in a fading channel it may be examined that the SNR required high for a given BER performance in comparison to AWGN channel. It is seen that to accomplish a given BER performance a higher SNR is needed for without interleaving in comparison to with interleaving. A coding gain of 0.5 dB with interleaving is calculated. For this the system was depicted 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 preset transmission frequency, f=900 MHz. The performance result is presented in below.

Figure 32: BER performance with and without puncture in a fading channel with Doppler 20 Hz.

Figure 32: BER performance with and without puncture in a fading channel with Doppler 20 Hz.
It may be examined that with Doppler frequency of 40 Hz, FEC with puncture (code rate 8/24) does not illustrate any increase in SNR for same BER performance of $10^{-4}$ compared to Doppler shift of 20 Hz.

It is observed that the performance of FEC without puncturing get worse 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 crucial for recovered performance.

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

In Figure 35 the system was depicted to frequency selective (three path) Rayleigh fading channel with Doppler shift of 100 Hz. It is examined that FEC without puncturing shows zero improvement in BER performance after SNR of 4 dB but FEC with Protection level 1 requires a SNR of 8 dB for a BER performance of $10^{-4}$. Therefore it specify that protection level 1 is well suitable for mobile reception.

The concatenated coding technique is considered here. For this outer coding utilized Block coding such as Linear, Cyclic and Hamming code and inner coding was achieved by convolutional codes. Codeword length was 511 and Message length was 502. The performance results are presented in Figure 4.23.

After analysing the BER performance using block coding methods, the performance analysis using BCH coding as the outer coding and convolutional coding as the inner coding. Codeword length was 511 and Message length 439 for error correcting capability of eight as simulation parameters. Figure 37 presents the results for BCH coding.

E. Simulation Results for DAB mode-II in Rician channel

Firstly we investigating the BER performance of DAB mode-II in Rayleigh fading channel, after that the performance analysis in Rician channel will be considered. The simulation parameters have been taken according to the DAB standard.

The performance of the system using different coding rates. 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 38: BER performance with different coding rates in a Rician channel.

Figure 38 exposes that as we increase the coding rate (or transmitting less redundancy) we require more transmitted signal power to get a better BER performance. It may be seen that in Ricien channel for FEC punctured with puncturing prototype PI=16 (code rate 8/24) gives 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.

Same as the performance of the system was examined with and without frequency interleaving. Figure 39 presents the performance result.

Figure 39: BER performance with and without interleaving in a Rician channel.

Figure 4.26 undoubtedly shows that interleaving is compulsory for the channel coding to function properly. It is seen that to accomplish 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 estimated.

Figure 40: BER performance with Protection level 1 in a Rician channel.

In Figure 40 the system was depicted flat- frequency fixed channel with Doppler shift of 100 Hz and k=1. It is examined that FEC without puncturing shows no improvement in BER performance after SNR of 3 dB but FEC with Protection level 1 requires a SNR of 8 dB for a BER performance of 10^-4. Therefore it shows that protection level 1 provides robustness against channel destructions.

According to the concatenated coding technique the outer coding employed Block coding such as Linear, Cyclic and Hamming code and inner coding was accomplished by convolutional codes. Codeword length is 511 and Message length to be 502. The performance results are presented in Figure 41.

Figure 41: 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 slightly compared with only convolutional coding. It may be observed that use of hamming coding as outer code achieves well compared with linear and cyclic code giving a coding gain of about 0.5 dB.

After analyzing 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 is 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 shows that concatenated coding (employing outer BCH coding and inner convolutional coding) recovers the BER performance marginally compared with only convolutional coding. It gives a coding gain of about 0.5 dB.

VI CONCLUSIONS

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.

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