Modeling and Performance prediction of Eureka-147 OFDM based DAB system

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Abstract*— The new digital radio system DAB (Digital Audio Broadcasting), developed within the Eureka 147 project is a very innovative and universal multimedia broadcast system that has the potential to replace existing AM and FM audio broadcast services in many parts of the world in the near future. DAB employs coded OFDM technology that enables it for mobile reception and makes receivers highly robust against channel multipath fading effect. In this paper, we have analyzed the bit error rate (BER) performance of DAB system conforming to the parameters established by the ETSI (EN 300 401) using frequency interleaving and Forward Error Correction (FEC) in different transmission channels. The results shows DAB to be suitable radio broadcasting technology for high performance in mobile environment.

Keywords- ***DAB, OFDM, Multipath effect, frequency interleaving, FEC.***

I. Introduction

Digital Audio Broadcasting (DAB) was developed within the European Eureka-147 standard [1] to provide CD quality audio programmes (mono, two-channel or multichannel stereophonic) along with ancillary data transmission (e.g. travel and traffic information, still and moving pictures, etc.) to fixed, portable and mobile receivers using simple whip antennas [2]. In 1995, ETSI (European Telecommunications Standards Institute) adopted DAB as the only European standard for digital radio. The reception quality of analog AM/FM systems on portable radio is badly affected by Multipath fading (reflections from aircraft, vehicles, buildings, etc.) and shadowing [3]. These systems also suffer from interference from equipment, vehicles and other radio stations. DAB uses coded orthogonal frequency division multiplexing (COFDM) technology to combat the effects of Multipath fading and inter symbol interference (ISI).

Additionally the VHF frequency band available for the sound broadcasting throughout the world has either saturated or fast approaching saturation. There is a need for more spectrally efficient broadcasting technology apart from conventional analog systems. Since DAB uses OFDM technology therefore the system can operate in single frequency networks (SFNs) providing the efficient usage of available radio frequency spectrum.

In this paper we developed a DAB base-band transmission system based on ETSI standard [1]. DAB transmission mode-II is implemented. A frame based processing is used in this work. Bit error rate (BER) has been considered as the performance index in all analysis. The design consists of energy dispersal scrambler, QPSK symbol mapping, convolutional encoder (FEC), D-QPSK modulator with frequency interleaving and OFDM signal generator (IFFT) in the transmitter side and in the receiver corresponding inverse operations is carried out along with fine time synchronization [4] using phase reference symbol.

Following this introduction the remaining part of the paper is organized as follows. Section II presents the DAB system standard. In Section III, the details of the modeling and simulation of the system using MATLAB is presented. Then, simulation results have been discussed in Section IV. Finally, Section V provides the conclusion.

II. SYSTEM SPECIFICATIONS

The working principle of the DAB system is illustrated in conceptual block diagram shown in Fig. 1. At the input of the system the analog signals such as audio and data services are MPEG layer-II encoded and then scrambled. In order to ensure proper energy dispersal in the transmitted signal, individual inputs of the energy dispersal scramblers is scrambled by modulo-2 addition with a pseudo-random binary sequence (PRBS), prior to convolutional coding [1]. The scrambled bit stream is then subjected to forward error correction (FEC) employing punctured convolutional codes with code rates in the range 0.25-0.88. The coded bit-stream is then time interleaved and multiplexed with other programs to form Main Service Channel (MSC) in the main service multiplexer.

The output of the multiplexer is then combined with service information in the Fast Information Channel (FIC) to form the DAB frame. Then after QPSK mapping with frequency interleaving of each subcarriers in the frame, $\pi/4$ shifted differential QPSK modulation is performed. Then the output of FIC and MSC symbol generator along with the Phase Reference Symbol (PRS) which is a dedicated pilot symbol generated by block named synchronization symbol generator is passed to OFDM signal generator. This block is the heart of the DAB system. Finally, the addition of Null symbol to the OFDM signal completes the final DAB Frame structure for transmission.

A. Channel coding, muliplexing and transmission frame

Channel coding is based on punctured convolutional forward-error-correction (FEC) which allows both equal and

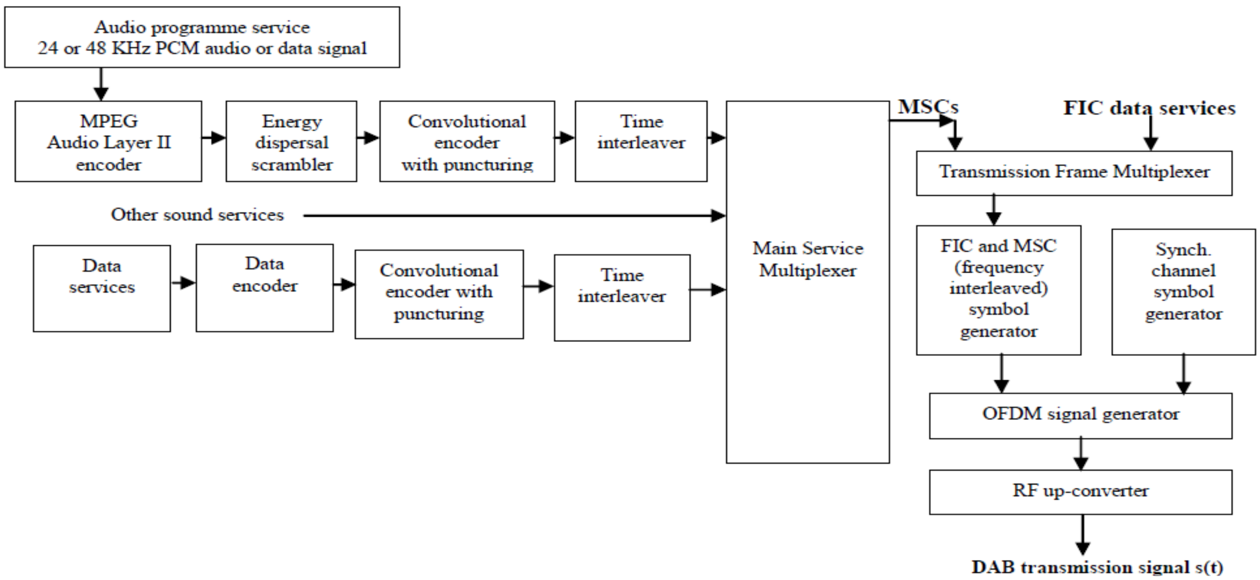

Figure 1. Complete DAB transmitter block diagram [1].

Unequal Error Protection (UEP), matched to bit error sensitivity characteristics [1]. Using rate compatible punctured convolutional (RCPC) codes, it is possible to use codes of different redundancy in the transmitted data stream in order to provide ruggedness against transmission distortions, without the need for different decoders [2]. Basic idea of RCPC channel coding is to generate first the mother code. The daughter codes will be generated by omitting certain redundancy bits, the process known as puncturing.

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

The DAB signal frame has the structure shown in Fig. 2 that helps in efficient receiver synchronization. The period T_F of each DAB transmission frame is of 24 ms or an integer multiple of it. According to system standard the first symbol is the Synchronization channel consisting of Null symbol during which no information is transmitted and the PRS symbol. The null symbol is used to estimate rough frame timing and PRS is the dedicated pilot symbol having predetermined modulation for fine time synchronization. The

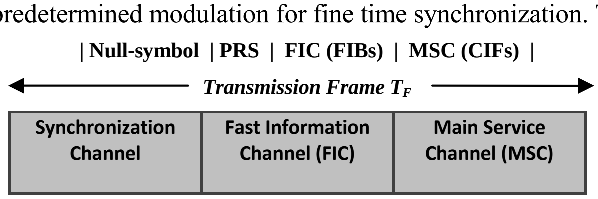

Figure 2. DAB transmission signal frame structure.

next symbol is the FIC channel which carries Multiplex Configuration Information (MCI). It has fixed symbols which are known to the receivers to decode any of the sub-channels instantly. The FIC is made up of FIBs (Fast Information Block). The FIBs contains 256 bits. The FIC data is a non-time- interleaved channel with fixed equal error protection [1] code rate (1/3).

The last symbol is the MSC channel that carry audio and data service component. It forms the main payload of the DAB frame. 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 (EEP or UEP). Each sub- channel may carry one or more service components. The MSC is made up of CIFs. The CIF contains 55296 bits. The smallest addressable unit of the CIF is the Capacity Unit (CU), comprising 64 bits. Therefore, the CIF contains 864 CUs.

B. *COFDM*

DAB uses COFDM technology that makes it resistant to Multipath fading effects and inter symbol interference (ISI). OFDM is derived from the fact that the high serial bit stream data is transmitted over large (parallel) number sub-carriers (obtained by dividing the available bandwidth), each of a different frequency and these carriers are orthogonal to each other. OFDM converts frequency selective fading channel into N flat fading channels, where N is the number of sub-carriers. Othogonality is maintained by keeping the carrier spacing multiple of 1/Ts by using Fourier transform methods, where Ts is the symbol duration. Since channel coding is applied prior to OFDM symbol generation which accounts for the term ‘coded’ in COFDM.

C. DAB Transmission modes and system parameters

The Eureka 147 DAB [1] system has four transmission modes of operation named as mode-I, mode-II, mode-III, and mode-IV, each having its particular set of parameters. The use of these transmission modes depends on the network configuration and operating frequencies. This makes the DAB system operate over a wide range of frequencies from 30 MHz to 3 GHz.

The details of DAB system parameters for all the four transmission modes is shown in Table-I. All the four DAB modes have same signal bandwidth of 1.536 MHz, 2 bits per carrier per symbol (D-QPSK modulation) and sampling frequency of 2.048 MHz It may be seen from Table I that Transmission mode-II has 384 sub-carriers at 4 KHz spacing. OFDM symbol length (Ts) is 312 µs. If channel impulse response is < 62 µs then there will be no ISI.

D. Source coding

Source coding employs MUSICAM (Masking Pattern Universal Sub-band Integrated Coding And Multiplexing) audio coding that uses the principle of Psycho acoustical masking as specified for MPEG-2 Audio Layer-II encoding. This exploits the knowledge of the properties of human sound perception, particularly, the spectral and temporal masking effects of the ear. Principle of MUSICAM audio coding system is that it codes only audio signal components that the ear will hear, and discards any audio component that, according to the Psycho acoustical model, the ear will not perceive [7]. This technique allows a bit rate reduction 768 Kbit/s down to about 100 Kbit/s per mono channel, while preserving the subjective quality of the digital audio signal. This allows DAB to use spectrum more efficiently and delivering high quality sound to the listeners.

III. THE SIMULATION MODEL

Fig. 3 presents the complete block diagram of the DAB system which was modeled and simulated by us in MATLAB environment. The main objective of this simulation study is to evaluate the BER performance of the DAB system using coded OFDM technique. The simulation parameters are obtained from Table I for transmission mode-II. A frame based processing is used in this simulation model. The system model was exposed to AWGN channel, Rayleigh fading channel and Rice channel for performance analysis. The important blocks of the simulation model is discussed in detail as follows:

A. Energy dispersal scrambler

In order to ensure appropriate energy dispersal in the transmitted signal, the individual inputs of the energy dispersal scramblers shown in Fig 3. shall be scrambled by a modulo-2 addition with a pseudo-random binary sequence (PRBS), prior to convolutional encoding. The PRBS shall be defined as the output of the feedback shift register and shall use a polynomial of degree 9, defined by:

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

TABLE I. SYSTEM PARAMETERS FOR THE FOUR DAB MODES

System Parameter	Mode -I	Mode -II	Mode -III	Mode -IV
No. of sub-carriers	**1536**	**384**	**192**	**768**
OFDM symbols/frame	**76**	**76**	**153**	**76**
Transmission frame duration	**196608 T**	**49152T 24 ms**	**49152 T 24 ms**	**98304 T 48 ms**
Null-symbol duration	**2656 T 1297 ms**	**664 T 324 µs**	**345 T 168 µs**	**1328 T 648 µs**
OFDM symbol duration	**2552 T 1246 ms**	**638 T 312 µs**	**319 T 156 µs**	**1276 T 623 µs**
Inverse of carrier spacing	**2048 T 1 ms**	**512 T 250 µs**	**256 T 125 µs**	**1024 T 500 µs**
Guard interval	**504 T 246 µs**	**126 T 62 µs**	**63 T 31 µs**	**252 T 123 µs**
Max. RF	**375 MHz**	**1.5GHz**	**3 GHz**	**750MHz**
Sub-carrier spacing	**1 kHz**	**4 kHz**	**8 kHz**	**2 kHz**
FFT length	**2048**	**512**	**256**	**1024**

B. Synchronization

Fine time synchronization or symbol timing synchronization [4] is performed by calculating the Channel Impulse Response (CIR) based on the actually received time frequency PRS and the specified PRS stored in the receiver. Multiplication of received PRS with complex conjugate of PRS at the receiver results in cancellation of the phase modulation of each carrier. The phase reference symbol can be converted to impulse signal or CIR can be obtained by an IFFT operation of the resultant product as illustrated in following formula:

$$CIR = IFFT\ \{Received\ PRS \bullet PRS^*\} \qquad (2)$$

C. Viterbi decoding

To minimize the transmission errors due to channel impairments the DAB system in the transmitter employed powerful rate compatible punctured convolutional code (RCPC) with constraint length 7 and mother code rate of 1/4 for channel coding. For decoding these codes the Viterbi algorithm [11] will be used, which offers best performance according to the maximum likelihood criteria. The input to the Viterbi decoder will be hard-decided bits that are ‘0’ or ‘1’, which is referred to as a hard decision. Computational requirements or complexity of Viterbi decoder grow exponentially as a function of the constraint length (L), so it is usually limited in practice to constraint length of L = 9 or less.

IV. SIMULATION RESULTS AND DISCUSSION

In this section we have presented the simulation results along with the BER analysis for AWGN, Rayleigh fading channel and Rice channel. The results are shown for transmission mode-II and the simulation parameters are taken as per the DAB standard [1].

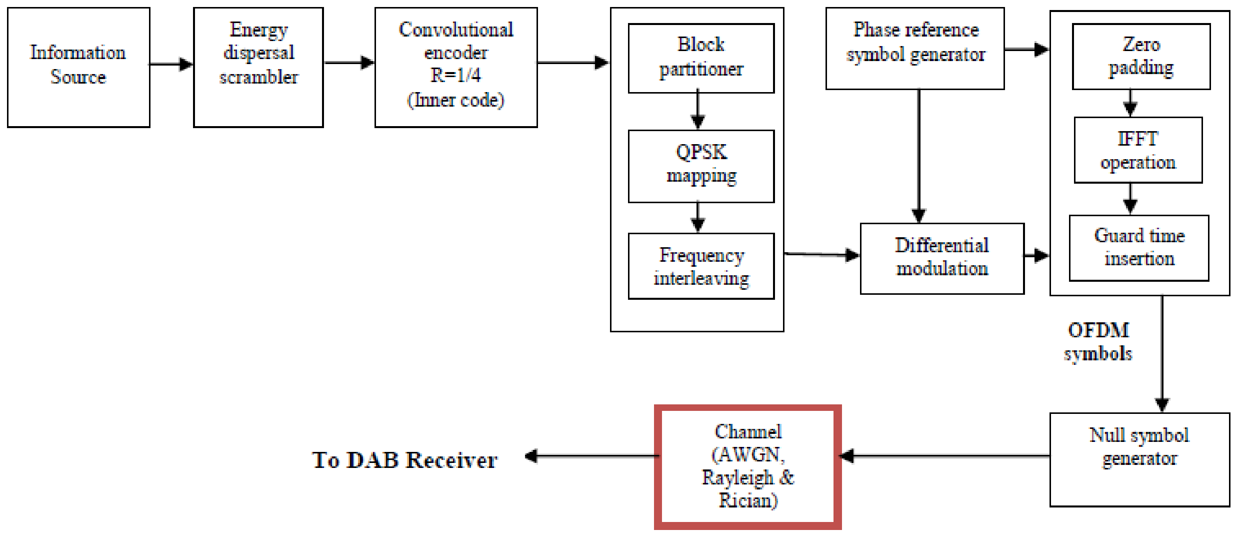

Figure 3. Block diagram of DAB system simulated.

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

Fig. 5 presents the transmitted DAB signal spectrum. The performance of DAB system with FEC coding is analyzed next. No puncturing was applied. Decoding was done with Viterbi algorithm. Fig. 6 presents the result for the DAB system with FEC coding in AWGN channel. From Fig. 6 it can be seen that the use of the channel coding improves the BER performance of the DAB system. It can be evaluated from figure that to achieve a BER of 10^{-4} coded DAB system without puncturing gives a coding gain of approximately 8 dB compared with the uncoded system.

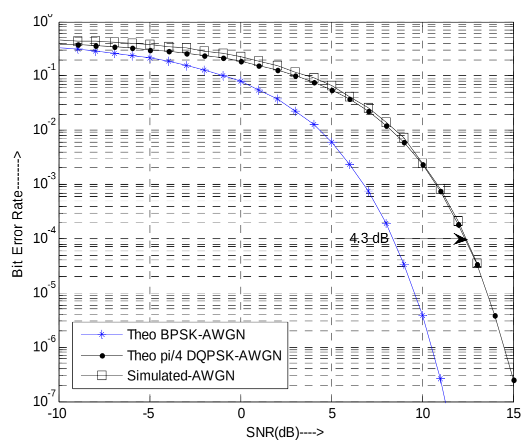

Figure 4. BER performance of DAB mode-II in AWGN channel.

The performance analysis in Rayleigh fading channel will be investigated next. Fig. 7 presents the results for FEC coding in fading channel with Doppler frequency 40 Hz (i.e., v= 48 km/hr). Fig. 7 reveals that uncoded system in a fading channel is severely affected by multipath as seen by poor BER performance. But channel coding improves the

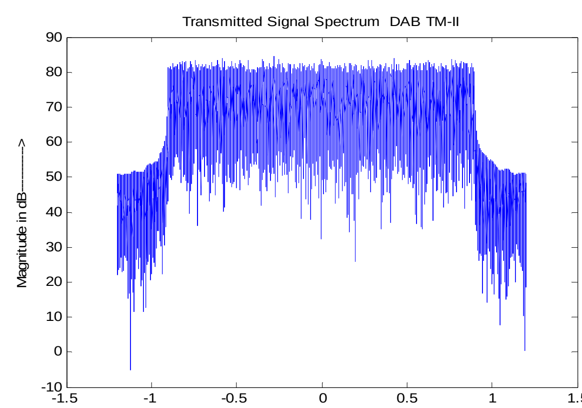

Figure 5. Transmitted signal spectrum.

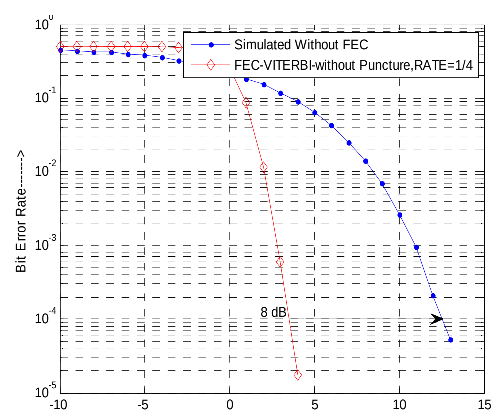

Figure 6. BER perfromance with and without FEC coding in AWGN channel.

performance of DAB system. It can be evaluated from figure that to achieve a BER of 10^{-4} in a fading channel requires a higher SNR value compared with AWGN channel.

Similarly the performance of the system in Rician channel will be investigated next. Fig. 8 presents the results for FEC coding in a rice channel. It can be evaluated from Fig. 8 that to achieve a BER of 10^{-4} coded DAB system without puncturing gives a coding gain of approximately 8 dB compared with the uncoded system.

V. PERFORMANCE ANALYSIS

The FEC channel coding provides an improved BER performance in different channels for OFDM-based DAB system. From the simulation results it is observed that FEC is practically well suited for channel coding giving a coding gain of about 8 dB in AWGN channel, 12 dB in Rayleigh fading channel and 8.2 dB in Rician channel.

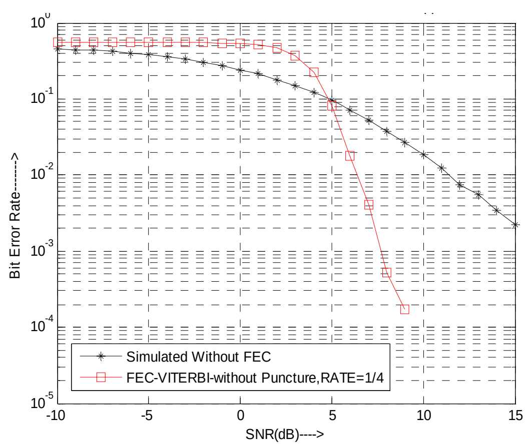

Figure 7. BER perfromance with and without FEC coding in Rayleigh channel with doppler 40 Hz.

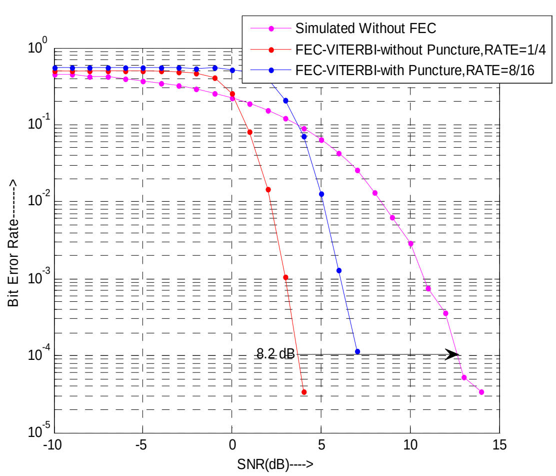

Figure 8. BER perfromance with and without FEC coding in Rice channel.

VI. CONCLUSION

A simulation based performance analysis of DAB system is described in this paper to evaluate the effectiveness of DAB system as a radio broadcasting technology in different transmission channels. Comparison was made between the performance of the uncoded system with convolutional coded system. The parameter of functional blocks can be changed to visualize the change in system performance. According to simulations, DAB appears to be suitable radio broadcasting technology for high performance in diverse transmission channels. With the present trend of wireless technologies becoming digital, DAB will become increasingly important in the future radio broadcasting.

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