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BER Analysis of Digital Broadcasting System Through AWGN and Rayleigh Channels

P.Ravi Kumar, V.Venkata Rao., M.Srinivasa Rao.

Abstract* — Radio broadcasting technology in this era of compact disc is expected to deliver high quality audio programmes in mobile environment. The Eureka-147 Digital Audio Broadcasting (DAB) system with coded OFDM technology accomplish this demand by making receivers highly robust against effects of multipath fading environment. In this paper, we have analysed the performance of DAB system conforming to the parameters established by the ETSI (EN 300 401) using time and frequency interleaving, concatenated Bose-Chaudhuri-Hocquenghem coding and convolutional coding method in different transmission channels. The results show that concatenated channel coding improves the system performance compared to convolutional coding.

Keywords- DAB, OFDM, Multipath effect, concatenated coding.

I. INTRODUCTION

Digital Audio Broadcasting (DAB) was developed within the European Eureka-147 standard [1] to provide CD quality audio programmes 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.

Earlier work focused more on the effect ofprotectionlevels in diverse transmission channels, design and implementation of DAB channel decoder (physical layer) on FPGA hardware [5, 13, 14].

In this paper we propose a BCH coding based concatenated channel coding technique for improved performance of DAB system in different transmission channels.

In this paper we developed a DAB base-band transmission system based on Eureka-147 standard [1]. 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. 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 analysis has been carried out with simulation studies under MATLAB environment.

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

II. INTRODUCTION TO DAB SYSTEM

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 theFast Information Channel (FIC) to form the DAB frame. Then after QPSK mapping with frequency interleaving of each subcarriers in the frame, π/4 shifted differential QPSK modulation is performed. Then the output of FIC and MSCsymbol generator along with the Phase Reference Symbol (PRS) which is a dedicated pilot symbolgenerated by block named synchronizationsymbol generatoris 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. The Eureka-147 DAB system consists of three main elements. These are Source coding, Channel coding, multiplexing and transmission frame and COFDM. These technical aspects make the system very reliable, multi- programme, providing robust reception on mobile receivers using simple antenna.

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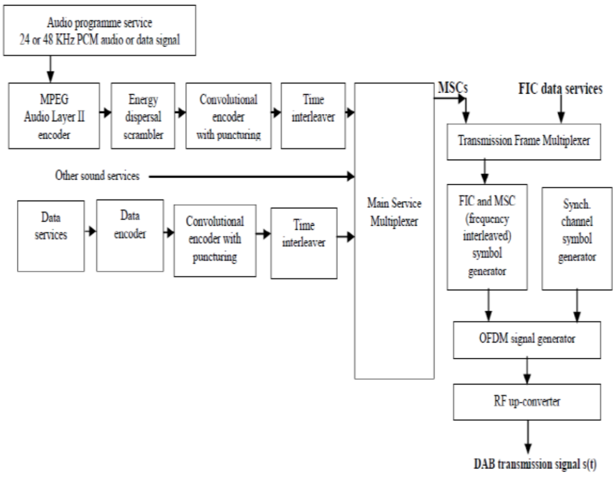
* Correspondence Author (s)
P.Ravi Kumar, M.Tech. Scholar, EC, NEC, Gumtur (A.P.), India
Dr. V.Venkata Rao., Professor & Head, NEC, Gumtur (A.P.), India
Mr. M.Srinivasa Rao., Asst. Professor, NEC, Gumtur (A.P.), India.



A. 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 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 anddelivering high quality sound to the listeners.

B. Channel coding, multiplexing and transmission frame.

Channel coding is based on punctured convolutional forward-error-correction (FEC) which allows both equal and Unequal Error Protection (UEP), matched to bit error sensitivity characteristics [1]. 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 fordifferent decoders [2]. Basic idea of RCPC channel coding is to generate first the mother code. The daughter codes will be generated byomitting certainredundancy 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 TF 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 PRSsymbol. 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 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 InformationBlock). 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

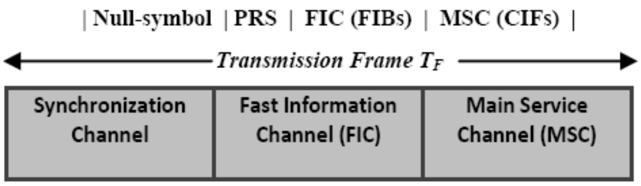


Figure 2. DAB transmission signal frame structure.

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 convolution ally 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

C. COFDM

DAB uses COFDM technology that makes it resistant to Multipath fading effects and inters 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 channelling to 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.

D. 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. Othogonality between sub-carriers is maintained if sinusoidshave ʃʃinteger number of cycles in Ts given by (1) below.

$$\int_0^{ts} \sin(2\pi f c t)\sin(2\pi 2 f c t)\, dt = 0 \qquad (1)$$

Where is the sub-carrier frequencyOthogonality in frequency domain is kept by taking FFT of 4 ms segment (mode-II) which is equivalent to convolving spectrum with sinc and sampling at multiples of 4 KHz.

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Mode-II can be designed to operate as a SFN where all the transmitter radiate over same frequency identical OFDM symbols at the same time. Optimal spacing between transmitters can be evaluated from the guard time interval Tg and velocity of light c as given by (2)

$$s = T_g \times c \qquad (2)$$

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

Equation (2) gives transmitter spacing of 18.6 Km for mode-I.

IV. THE SIMULATION MODEL

Fig. 3 presents the complete block diagram of the DAB system which was modelled 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 concatenated coding 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 Rician channel for performance analysis. The important blocks of the simulation model is discussed in detail as follows:

A. Concatenated channel coding.

The virtually error free channel can be achieved by concatenated coding usingconvolutional code as„inner code" together with a BCH code as „outer code" given in Fig. 4. This technique improves the BER performance of the finite field called Galois array with a particularly chosen generator polynomial. These are cyclic codes.

The inner coding is based on punctured convolutional forward-error-correction (FEC) which allows both Equal and Unequal Error Protection. We have used convolutional encoder with constraint length 7 with octal forms of generator polynomials as 133, 171, 145 and 133, respectively [1].

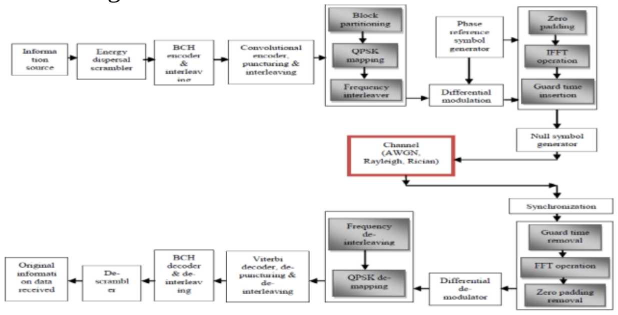
The mother code has a rate R =1/4 which means that each input bit is protected by four bits. Thus expand rate is 1/R.

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 form

$$\text{CIR} = \text{IFFT } \{\text{Received PRS} \bullet \text{PRS}^*\} \qquad (3)$$

Where PRS* is the complex conjugate of the phase reference symbol. The peak of the impulse signal obtained from (4) will give position of the start of the PRS compared to set threshold (T) providing correct symbol timing as well as frame timing.



To minimise 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 constraintlength (L), so it is usually limited in practice to constraint length of L = 9 or less.

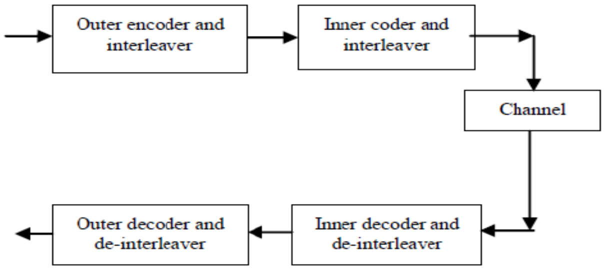


Figure 4. Block diagram of concatenated coding.

V. SIMULATION RESULTS AND DISCUSSION.

In this section we have presented the simulation results along with the bit error rate (BER) analysis for AWGN channel, 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].

First of all the correctness of the DAB simulation model given in Fig. 3 will be tested. Fig. 5 presents the system BER performance in AWGN channel. It can be seen from Fig. 5

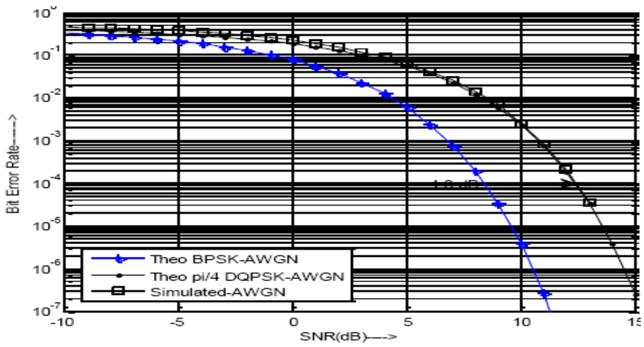


Figure 5. BER performance of DAB mode-II in AWGN channel.

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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.

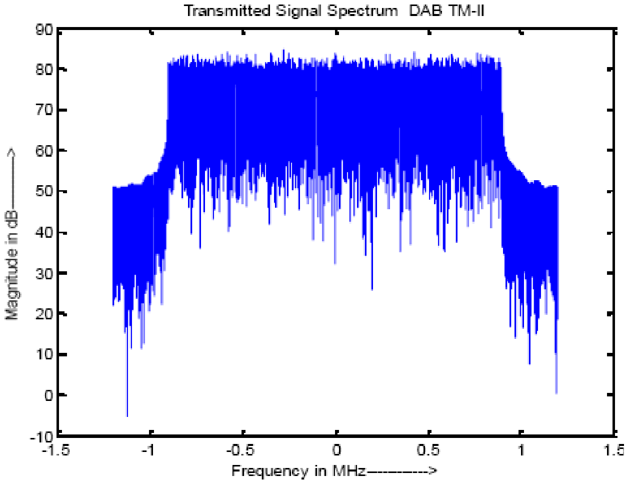

Figure 6. Transmitted signal spectrum.

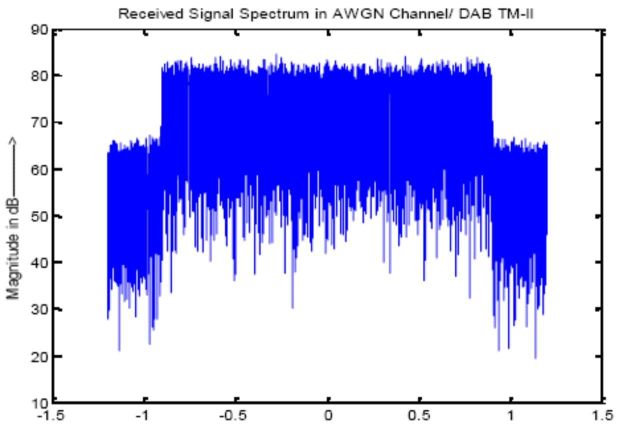

Figure 7. Received signal spectrum in AWGN channel.

The analysis of simulated DAB signal spectrum is analysednext. SNR (signal-to-noise) was taken to be 15 db. The bandwidth of the signal is represented in baseband mode. Fig. 6 presents the transmitted DAB signal spectrum and Fig. 7 the received DAB signal spectrum (mode-II) in AWGN channel. Fig. 7 reveals that received signal spectrum in AWGN channel has approximately the same power level as transmitted signal.

Fig. 8 shows the received DAB signal spectrum (mode- I) in Rayleigh fading channel. It can be seen that the power level of received signal spectrum in Rayleigh fading channel is 2 dB less than the transmitted signal. Fig. 9 presents the received DAB signal spectrum (mode-II) in Rician channel finally, the Fig. 9 shows that the power level of received signal spectrum in Rician channel is 5 dB less than the transmitted signal. Thus network gain is less in case of Rician channel.

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

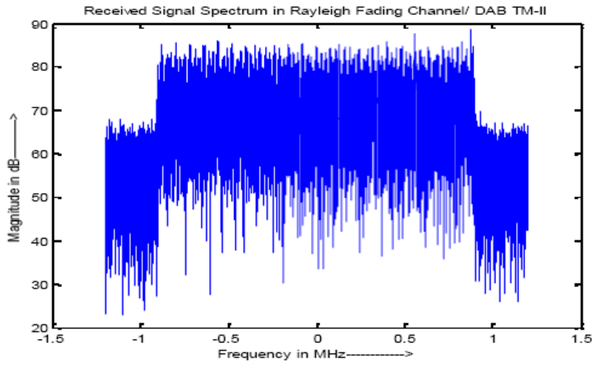

Figure 8. Received signal spectrum in Rayleigh fading channel.

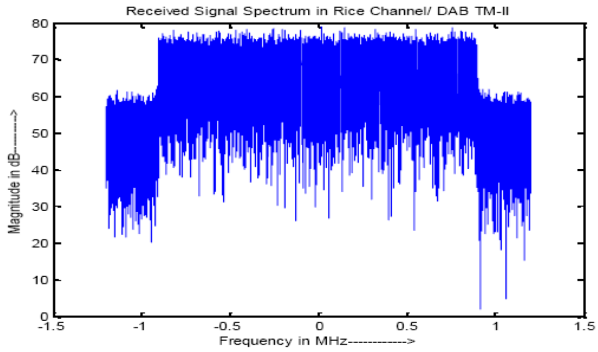

Figure 9. Received signal spectrum in Rician channel.

The performance analysis using concatenated channel coding technique is investigated next. Here we use BCH coding as the outer coding and convolutional coding as the inner coding. Code word length was taken as 511 and Message length to be 439 for error correcting capability of eight. Fig. 11 presents the results for BCH-FEC coding.

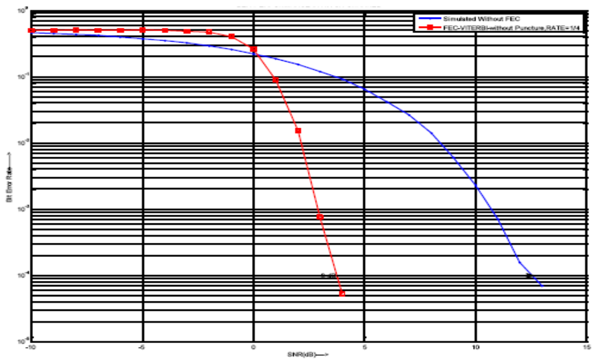

Figure 10. BER perfromance with and without FEC coding in AWGN channel.

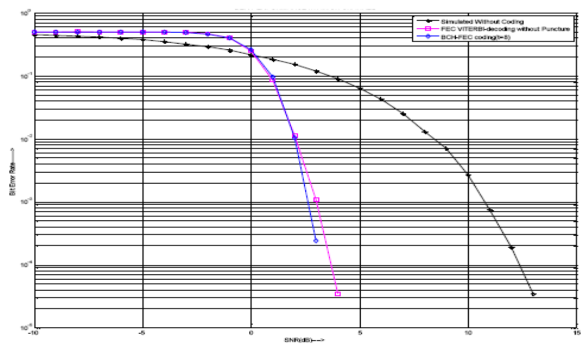

Figure 11. BER perfromance with concatenated coding in AWGN channel.

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Fig. 11 presents that concatenated coding (employing outer BCH coding and inner convolutional coding) improves the BER performance marginally compared with only convolutional coding. A coding gain of about 0.5 dB is observed in AWGN channel.

After analysing the BER performance in AWGN channel, the performance analysis in Rayleigh fading channel was investigated next. Fig. 12presents the results forBCH-FEC coding in fading channel with Doppler frequency 40 Hz (i.e., v= 48 km/hr.). Fig. 12 reveals that concatenated coding (employing outer BCH coding and inner convolutional coding) improves the BER performance compared with only convolutional coding. It provides a coding gain of about 1 db.

Similarly the performance of the system with concatenated coding in Rician channel will be investigated next. Fig. 13 presents the results for BCH-FEC coding in a rice channel

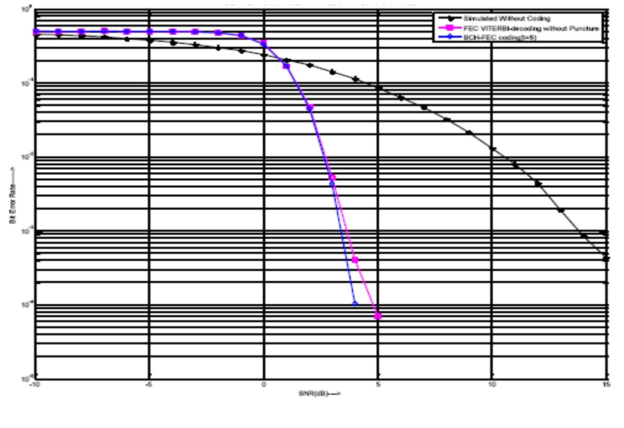

Figure 12. BER perfromance with concatenated coding in Rayleigh fading channel.

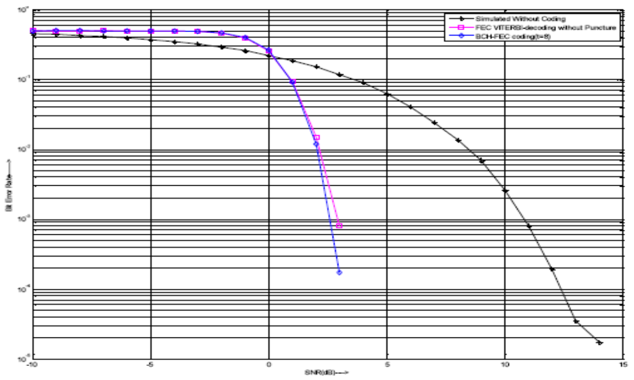

Figure 13. BER perfromance with concatenated coding in Rician channel.

Fig. 13 indicates that concatenated coding (employing outer BCH coding and inner convolutional coding) improves the BER performance marginally compared with only convolutional coding. It provides a coding gain of about 0.5 dB in Rice channel.

VI. PERFORMANCE ANALYSIS

The proposed concatenated 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. Results shows that system employing BCHcombined with FEC coding gave a coding gain of 0.5 dB in AWGN channel &Rice channel and coding g ain of 1 dB in fadingchannel for the toachieve a BER of 10^{-4}.

VII. CONCLUSION

A simulation based performance analysis of DAB system using concatenated coding technique 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 performances of the concatenated coding with only convolutional coding. 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.

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Mr. P.Ravi Kumar, was completed his B.TECH FROM NALANDA INSTITUTE OF ENGINEERING AND TECHNOLOGY IN 2010 and completed M.TECH From NARASARAOPET ENGINEERING COLLEGE IN 2012.He was interested in research in communication systems.

Dr. V.VENKATA RAO, was completed his PhD in the area of communication systemsand published so many papers in international and national journals and international and national conferences.

Mr. M.SRINIVASA RAO, was completed his B.TECH FROM LOYOLA INSTITUTE OF TECHNOLOGY MANAGEMENT IN 2007 and completed M.TECH From NALANDA INSTITUTE OF ENGINEERING AND TECHNOLOGY IN 2012.He was interested in research in VLSI SYSTEMS. He published a paper in an international conference paper named "DESIGN AND OPTIMISED IMPLEMENTATION OF a 1024 POINT PIPELINE FFT PROCESSOR". He was ready to going to do PhD in VLSI.

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