IOSR Journal of Electronics and Communication Engineering (IOSR-JECE)
e-ISSN: 2278-2834,p- ISSN: 2278-8735.Volume 10, Issue 4, Ver. III (Jul - Aug .2015), PP 56-59
www.iosrjournals.org

Efficient Performance analysis of OFDM based DAB systems Using Reed Solomon coding technique

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Abstract: *DAB (digital audio broadcasting) is a radio system which is depends on EUREKA-147 project. This project was designed by the Members of European Broadcasting Union (EBU). DAB System depends on wireless communication. DAB channel estimation was designed by the OFDM technology that reduces co-channel interference and multipath fading. In this paper we have analyzed bit error rate performance v/s signal to noise ratio (BER v/s SNR).for DAB modes we done block coding with the help of MATLAB Simulink tool. In this paper we use 16-QAM modulation technique and reed-Solomon convolution coding.*
Keywords: *DAB, OFDM, co-channel interference, multipath fading, BER, SNR, MATLAB and 16-QAM.*

I. Introduction

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

In this paper we proposed a reed-Solomon based channel coding technique for improve performance of DAB system in different transmission channel.

In this paper we consider 16-QAM modulation technique and OFDM signal generator. For the DAB block coding we use IFFT (inverse fast Fourier transform), FEC (forward error correction), and frequency interleaving, fading or non-fading channels. Here DAB transmitter sides we use convolution encoding (reed-Solomon) so must be required Viterbi decoder at receiver side.

II. Overview Of Dab System

Digital Audio Broadcasting (DAB) is a new digital radio system that delivers radio services from the studio to the receiver. DAB is intended to deliver very high quality digital audio programmers and data services to fixed, mobile and portable receivers which can use simple whip antennas.

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

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

(A) Source coding:

Source coding of DAB specified MPEG-2 audio layer encoding. Audio coding system codes only audio signal components that the ear will hear. 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. DAB to use spectrum more efficiently and delivering high quality sound to the listeners and receiver. All type coding is used for data encryption, data decryption and data compression according to the sender and receiver.

(B) Channel coding:

Channel coding is a second element of DAB system which is used in modulation and demodulation. Channel coding is dependent on different type of coding like BCH coding, Huffman coding, reed Solomon coading, shannfano coding. In this thesis the channel coding of DAB depends on the "**Reed–Solomon**" coding.

(C) Reed–Solomon coding:

Reed–Solomon code was introduced by **Irving S. Reed** and **Gustavo Solomon** in 1960.These codes is also known as error correcting codes. This code is satisfied to the non-binary cyclic error-correcting codes. This code is able to detect multiple symbol combination up to'‘t’ error symbol and correct multiple symbol combination up to ‘t/2’ error symbol. Reed–Solomon codes are suitable as multiple-burst bit-error correcting codes

(D) DAB transmission frame

The main service channel (MSC) is a logical channel which has all information about the data. This channel is divided into sub channel which are individually coded with error protection, the fast information channel (FIC) is used to access the information rapidly by a receiver. It is a non-time-interleaved data channel that is used to highly protect for the data. FIC is made up by the (FIBs) fast information block. A synchronous channel (SC) is used to comprise two symbols. One is null symbol and another one is phase reference symbol. In null symbol rf signal is transmitted and phase reference symbol has a predetermined modulation. This channel is decoded the received DAB signal.

(E) COFDM (coded orthogonal frequency division multiplexing)

COFDM is coded orthogonal frequency division multiplexing that is used in OFDM method. COFDM is an error correction coding techniques by which remove the signal error. Signal is communicate without any error from one point to another point. The term of coded is used to reduce fading in the signal. COFDM is the best technique of OFDM by which we get best signal without any error, without interference and without fading. The signal strength is increase by the COFDM technology. OFDM have many types like VOFDM (vector OFDM), FOFDM (flash OFDM) etc.

(F) Modes of DAB

The digital audio broadcasting system consist four modes by which data is transmit and receive from one point to another point. The four modes is mod I, mod II, mod III and mod IV. All modes have unique parameters to design a signal which is transmitted. In this paper we consider only two modes, mode I and mode II for communication. The detail of DAB system parameter for all transmission modes is shown in the table. Each mode of DAB system having particular set of parameter and these parameters of modes are fixed from EUREKA-147 project. This table consist many parameters for DAB modes like bandwidth, duration, framing, application etc. by these parameters we get all knowledge about transmitting and receiving signal.

Transmission mode	I	II	III	IV
Number of carriers (K)	1536	384	192	768
Number of OFDM Symbol/frame (L)	76	76	153	76
Transmission frame Duration (T_F)	196 608T 96ms	49 152T 24ms	49 152T 24ms	98 304T 48ms
Null symbol Duration (T_{NULL})	2656T ~1,297ms	664 T ~324us	345T ~168us	1328T ~648us
Total symbol Duration (T_S)	2552T ~1,297ms	638T ~312us	319T ~156us	1276T ~623us
Useful symbol Duration (T_U)	2048T 1ms	512T 250us	256T 125us	1024T 500us
Guard interval Duration	504T ~246us	126T ~62us	63T ~31us	212T ~123us
Carrier frequency separation	1 KHz	4 KHz	8 KHz	2 KHz
Application	Terrestrial transmission	LAN services	Satellite communication	L- band Frequency

Table 1: DAB modes parameter

According to the table each mode have same signal bandwidth that is 1.536 MH_Z, 2 bits per carrier per symbol and sampling frequency is 2.048 MH_Z it may be seen from table that number of sub-carrier is decrease and number of sub-carrier spacing is increase with respect to the transmission modes.

III. Simulation Result And Discussion

In this section we have presented simulation result of DAB system along with the bit error rate and signal to noise ratio. This simulation result is analysis for either fading channel or non-fading channel. The results are shown for DAB mode I and DAB mode 2as the parameter as per the DAB standard. The previous work define (0.45 - 0.30) with respect to bit error rate but my work and paper define signal to noise ratio with respect to bit error rate (0.44 - 0.14) for transmission mode I and (0.41 - 0.11) for transmission mode II in DAB system.

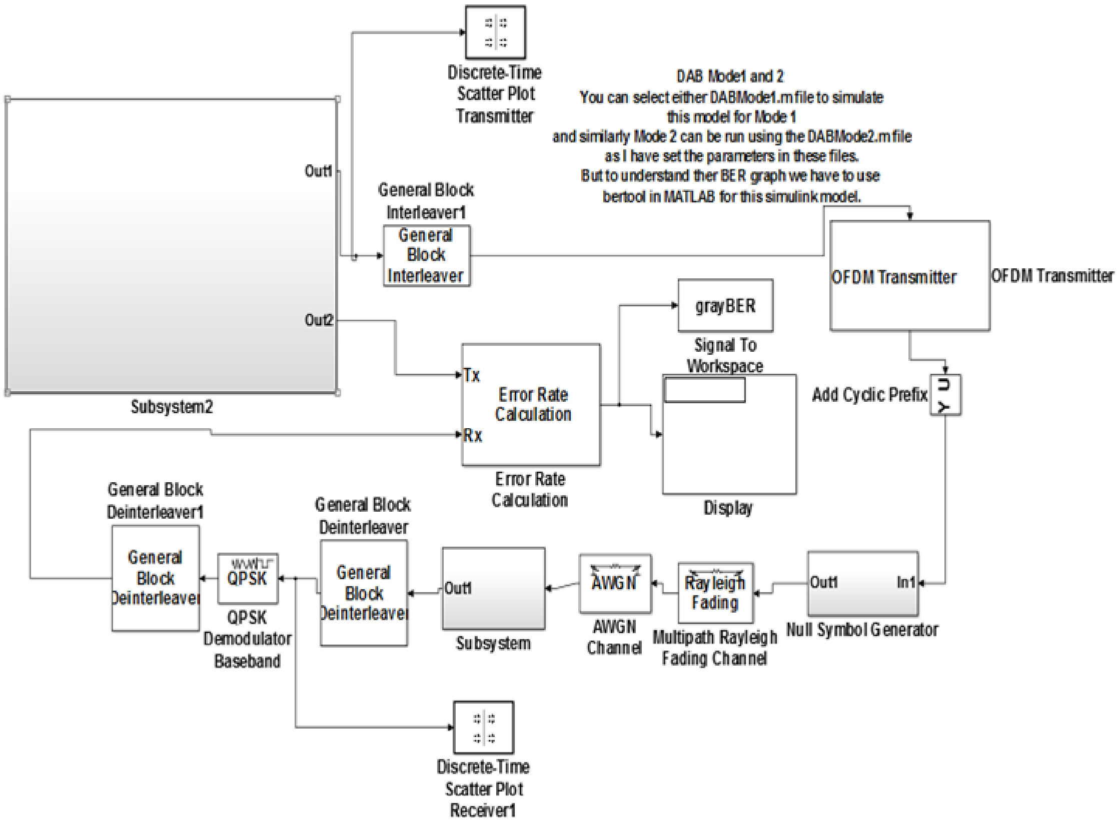

Fig 1: DAB Block Diagram

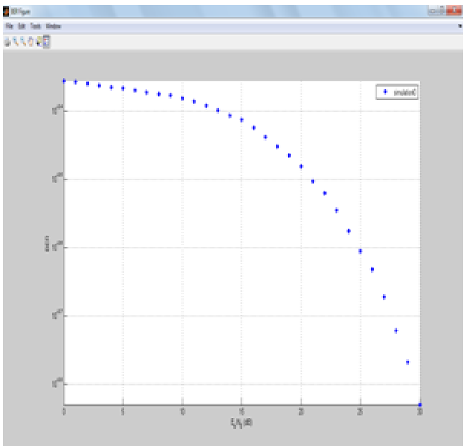

Fig 2: BER V/S SNR Graph for Mode I

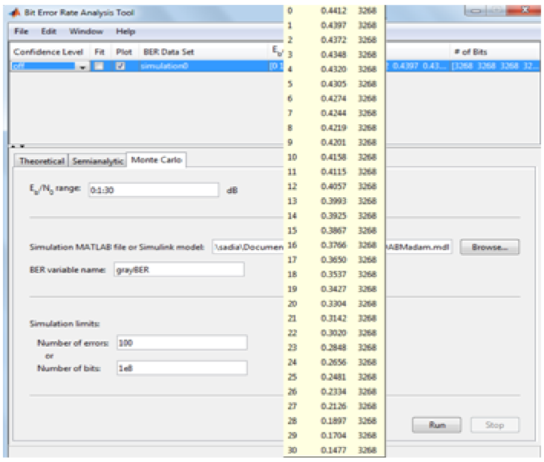

Fig 3: BER V/S SNR Numeric Values

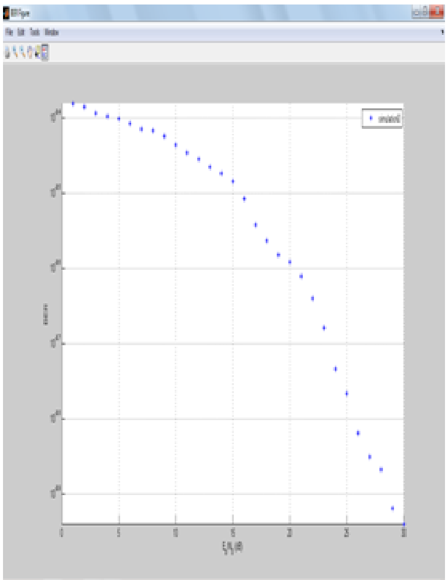

Fig 4: BER V/S SNR Graph for Mode I

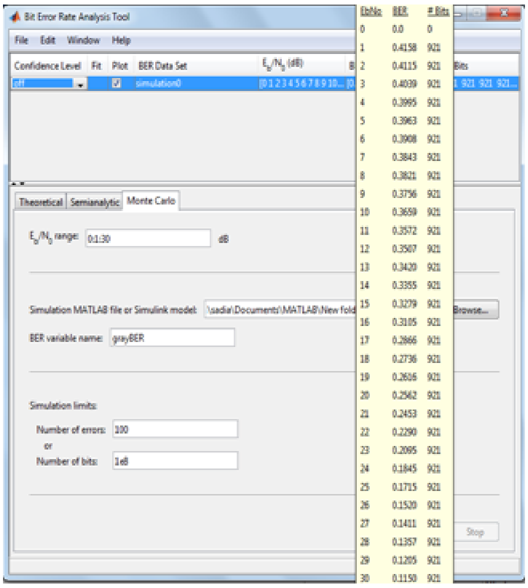

Fig 5: BER V/S SNR Numeric Values

IV. Conclusion

The reason behind the lower complexity and less execution time is that, we have used Simulink and some part we have designed in MATLAB. Use of Simulink instead of MATLAB reduces the code length and therefore complexity also. In this paper we used with AWGN channel use Rayleigh fading channel and then analysis the result. After the simulation I achieve my motive that if DAB signal have high signal to noise ratio then we get low bit error rate of the same communication signal. The digital audio broadcasting system does not provide high quality radio signal in wide areas.

The limitation of this research is that this system technology is also not capable to transmit and receive the multi data services. In this the radio signal is also affected by the multipath propagation like building, aircraft, vehicles any other obstacles etc. this system signal also suffers from inter symbol interferences and fading.

The future works for this research is as a perfect working hardware for this DAB System can be designed in future by the implementation work. A complete DAB system can be designed for all transmission modes with the same techniques and parameters we have used for our thesis.

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