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Performance Of Bit Error Rate Of Coded OFDM For Digital Audio Broadcasting By Utilizing Convolutional Systems Under Different Channels

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Abstract: **In this paper we exhibit an investigation of Bit Error Rate (BER), for Digital Audio Broadcasting (DAB) framework, utilizing Coded OFDM with various channel coding plans. Examination is done for convolutional coded and turbo coded information in an Additive White Gaussian Channel (AWGN) in light of various imperative lengths and code generator polynomials utilized for coding. A relative study on the computational multifaceted nature is likewise done by applying an sound flag and measuring the information preparing time per outline, on PCs with various processor speeds. It is demonstrated that a coding increase of roughly 6 dB is accomplished utilizing turbo coding when contrasted with convolution coding, at an expense of higher computational multifaceted nature.**

Keywords– **DAB; OFDM; Convolutional Codes; Turbo Codes;**

I. INTRODUCTION

The prerequisite of versatility while associated with system is energizing the development of remote correspondence. The routine simple transmission systems don't perform well in versatile environment, since appropriate procedures to moderate the impacts of multipath spread instigated blurring have not been produced for these frameworks. Orthogonal Frequency Division Multiplexing (OFDM) is one such strategy to battle the impact of multipath blurring, recurrence specific blurring and Intersymbol Obstruction (ISI) [1]. OFDM Diminishes the measure of equipment execution since multiplexing and sifting operations can be performed by utilizing the Fast Fourier Transform (FFT). This wipes out the need numerous oscillators at the transmitter and synchronizing circles at the beneficiary. Because of the cyclic augmentation of sign period into a gatekeeper interim, OFDM framework is reasonable for Single Frequency Networks (SFN) [5]. In this paper an OFDM application standard called Computerized Audio Broadcasting (DAB) framework model is executed in Matlab/Simulink environment. The execution of this framework over a channel irritated by AWGN clamor is considered. Coded Orthogonal Frequency Division Multiplexing (COFDM) procedure is considered in which convolutional codes and turbo codes are utilized what's more, figured the subsequent piece mistake rates (BER). The variety in BER is dissected taking into account diverse coding parameters. A sound sign is transmitted and information preparing time per edge is measured and thought about for diverse channel coding plans.

Orthogonal frequency division multiplexing (OFDM) is an attractive technique for combating the effects of delay spread in high-speed wireless data transmission [1]. By transmitting multiple data streams in parallel over different low-rate sub channels, intersymbol interference (ISI) can be reduced significantly. If each subchannel is narrow enough such that the multipath fading can be characterized as flat, the need for equalization can be eliminated. The popularity of OFDM is evident by its use in recent standards for digital audio and video broadcasting (DAB/DVB), asymmetric digital subscriber line (ADSL), and wireless local area network (WLAN) [2] applications.

Orthogonal Frequency Division Multiplexing (OFDM) is special form of multi-carrier transmission technique in which a single high rate data stream is divided into multiple low rate data streams. These data streams are then modulated using subcarriers which are orthogonal to each other. In this way the symbol rate on each sub channel is greatly reduced, and hence the effect of inter symbol interference (ISI) due to channel dispersion in time caused by multipath delay spread is reduced. Guard interval can also be inserted between OFDM symbols to reduce ISI further. The orthogonality between subcarriers can be maintained, even though the signal passes through a time-dispersive channel by cyclically extending the OFDM symbols into guard interval. In an OFDM transmission system, each subcarrier is attenuated individually under the frequency-selective and fast fading channel. The channel performance may be highly fluctuating across the subcarriers and varies from symbol to symbol [3].

However, in OFDM, each subcarrier is attenuated individually under the frequency-selective and fast fading channel. The channel performance may be highly fluctuating across the subcarriers and varies from symbol to symbol .If the same fixed transmission scheme is employed for all OFDM subcarriers; the error probability is dominated by the OFDM subcarriers with highest attenuation resulting in a poor performance. Therefore, in case of frequency selective fading the error probability decreases very slowly with increasing average signal-to-noise ratio (SNR) [4].

This problem can be mitigated if different modulation schemes are employed for the individual OFDM subcarriers. Unlike adaptive serial systems, which employ the same set of parameters for all data symbols in a transmission frame, adaptive OFDM schemes have to be adapted to the SNR of the individual subcarriers. This will substantially improve the performance and data throughput of an OFDM system. For example if the subcarriers that will exhibit high bit error probabilities in the OFDM symbol to be transmitted can be identified and excluded from data transmission, the overall BER can be improved in exchange for a slight loss of system throughput OFDM also has some drawbacks. Because OFDM divides a given spectral allotment into many narrow subcarriers each with inherently small carrier spacing, it is sensitive to carrier frequency errors. which may be caused by Doppler shift in the channel, or by the difference between the transmitter and receiver local oscillator frequencies. This frequency offset introduces inter-carrier interference in the OFDM symbol. This paper describes an efficient way to improve the BER performance of an AOFDM system by introducing convolutional coding in the system. Efficiency of the resultant system is verified via simulation and analysis.

II. SYSTEM MODEL

OFDM with forward error correction methods is most suitable scheme to transmit information efficiently BER using the Convolutional code in the presence of the fading channel is shown is explained briefly in the review paper [8]. A type of Convolutional code, called turbo codes that enable the reduction of errors in noisy channels without the need to increase the signal power. A turbo code consists three distinct part namely encoder, interleaver and decoder .The performance of turbo code depend on the design & implementation of all the three part. A typical turbo encoder uses parallel concatenated convolutional codes (PCCC) in which data bits are coded by two or more recursive systematic convolutional (RSC) coders separated by an interleaver. Turbo code uses two same block of decoders, the decisions from one component decoder are passed as input to another decoder and this process is iteratively done for several times to get more reliable decisions. The high bit error correction power of turbo code originates from the interleaving at the encoder and iterative decoding using extrinsic information at the decoder [8] Turbo coded AOFDM (TC AOFDM) system combines the good features of AOFDM with that of turbo code. Using the iterative property of turbo codes, a large coding gain is achieved with respect to an un-coded system.

A. A Simplified DAB Block Diagram

A general block diagram of the Digital Audio Broadcasting transmission system is shown in Fig. 1. The simple sign is encoded and connected to channel encoder. After channel coding the bit streams are QPSK mapped. The information is then gone to OFDM generator. The high information rate bit stream is separated into "N" parallel information surges of low information rate and separately regulated on to orthogonal subcarriers which is

acknowledged utilizing IFFT calculation. Orthogonality of the subcarriers accomplishes zero Inter Image Interference, hypothetically [1]. At long last, the OFDM image is given cyclic prefix and the finished Touch outline structure is transmitted through an AWGN channel.

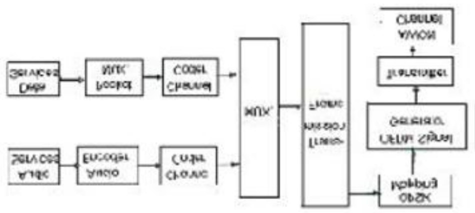

Figure 1. DAB transmitter – Block Diagram.

B. DAB Transmission Modes

DAB system has four transmission modes, each with its own set of parameters, shown in Table-I [12]. In this paper Transmission Mode-I is selected for simulation.

III. CHANNEL CODING

A. Convolutional Encoding & Viterbi Decoding

A convolutional encoder comprises of a M-stage shift register with "k" inputs, endorsed associations with "n" modulo-2 adders and multiplexer that serializes the yields of the adders. Here the encoder chose has k=1, ie; the info succession touches base on a solitary information line. Subsequently the code rate is given by r = 1/n. In an encoder with a M stage shift enroll, the memory of the coder approaches M message bits and K = (M+1) movements are required before a message bit that has entered the movement register can at long last exit. This parameter K is alluded to as the imperative length of the encoder. The channel coding utilized for standard DAB comprises of code rate ½, memory 6, convolutional code with code generator polynomials 133 and 171 in octal organization [2]. For Spot lower code rates give better execution. Subsequently in this work, encoder with code rate=w̄ is chosen. One such convolutional encoder is appeared in Fig. 2. The quantity of registers=6. Consequently the imperative length K=7. Generator Polynomials are 171, 133 and 115 in octal organization. Reproduction is done for different estimations of limitation length and generator polynomials, which are given in Table-II.

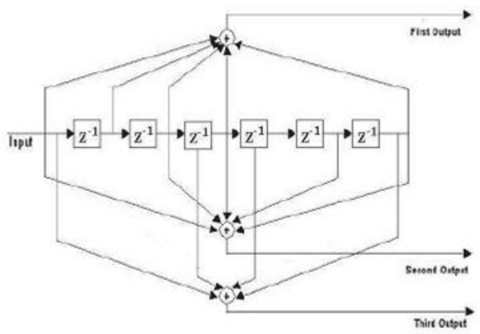

Figure 2. A rate w̄ convolutional encoder with constraint length, K=7.

B. Parallel Concatenated Convolutional Turbo Coding & Decoding

Parallel Concatenated Convolutional turbo code (PCC turbo code) comprises of two or more Recursive Systematic Convolutional (RSC) coders working in parallel [8]. The reason for interleaver is to offer each encoder an irregular variant of the data bringing about equality bits from each RSC that are free.

Transmission Mode	Mode I
No. of sub-carriers	1536
Transmission frame duration (F_t)	96 ms.
OFDM Symbols per Transmission Frame	76
Sample Time (T_s)	0.48828 µs
Frame length	196608 (or F_t/T_s)
FFT length	2048
Guard interval (Cyclic Prefix)	504
OFDM length	2552
Channel Coding schemes Used	Convolutional coding (rate ⅓), Turbo coding (rate ⅓)
Modulation	QPSK
Channel	AWGN

On the receiving side there are same number of decoders as on the encoder side, each working on the same information and an independent set of parity bits. In this work, to give same code rate to turbo encoder as on account of convolutional encoder, a parallel connection of two indistinguishable RSC encoders are utilized which gives a code rate of w̄. One such turbo encoder is appeared in Fig. 3, where the quantity of registers in each RSC encoder=2. Henceforth the limitation length K=3. Generator polynomials are 7 and 5 in octal arrangement. The number 7 indicates the input polynomial. ʌ is the arbitrary interleaver. Reenactment is completed for different estimations of requirement length, generator polynomials and input polynomials, which are given in Table-III.

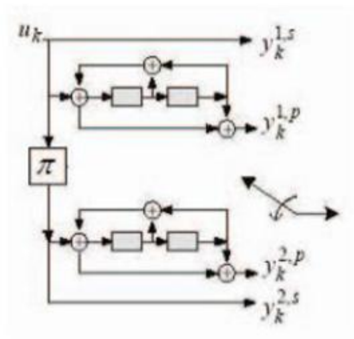

Figure 3. A rate w̄ turbo encoder with 2 parallel recursive systematic convolutional encoders, each with constraint length, K=3.

The inputs are data bits and called uk. The yields are code bits. Of these, the yield of first encoder, yk1, s is known as the precise piece, and it is the same as the information bit. The second yield bit, yk1, p is the principal equality bit which is recursive deliberate piece. An interleaver, indicated by ʌ, is put in the middle of the two encoders to guarantee that the information got by the second encoder is measurably autonomous. The third yield bit, yk2, p is the second equality bit which is additionally a recursive deliberate piece. The fourth yield yk2, s is deterministically reshuffling adaptation of yk1, s, which is not transmitted. For decoding, the Viterbi Algorithm is not suited to generate the A-Posteriori-Probability (APP) or soft decision output for each decoded bit. Here Maximum-A-Posteriori (MAP) algorithm is used for computing the metrics. Block diagram of turbo decoder is shown in Fig. 4.

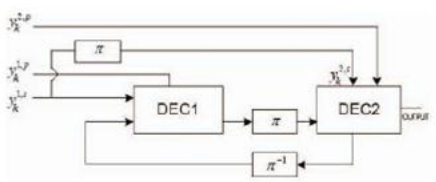

Figure 4. Turbo decoder – Block Diagram.

In Fig. 4, DEC1 and DEC2 are 2 APP decoders. ʌ and ʌ -1 are random interleaver and deinterleaver respectively [14]. The symbol vector sent for each time are described by yk=(yk1, s, yk1, p, yk2, p). The goal is to take these and make a guess about the transmitted vector and hence code bits which in turn decode uk, the information bit.

IV. SIMULATION MODEL

A. Simulation Parameters

The simulation parameters are shown in Table-II [1]. The different channel coding schemes and its parameters used for the analysis are given in Table-III. Even though, a complete DAB system consists of a multiplex of many information service channels, here, for the purpose of analysis, only a single audio signal is selected for transmission**.**

Channel Coding types	Constraint length	Code generator polynomials (Octal format)	Feedback Polynomial
Convolutional Coding	3	7, 6, 5	--
	4	15, 13, 11	--
	5	34, 27, 23	--
	6	71, 57, 47	--
	7	171, 133, 115	--
Turbo Coding	3	7, 5	7
	4	15, 13	15
	5	34, 27	34
	6	71, 57	71
	7	171, 133	171

TABLE III. Channel Coding Parameters

B. Simulation Waveforms

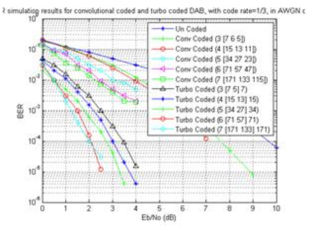

Fig5:BER simulation results for convolutional coded and turbo coded DAB, with code rate=1/3, in AWGN channel

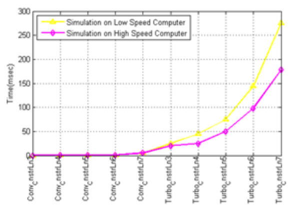

Fig6:Simulation on Low Speed Computer,Simulation on High Speed Computer along with Location at NorthWest direction

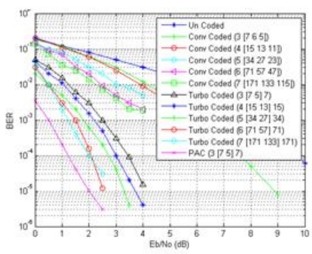

Fig7: BER simulation results for convolutional coded and turbo coded DAB, with code rate=1/3, in AWGN channel

V. CONCLUSION

Computerized Audio Broadcasting framework utilizing Coded OFDM is executed and contemplated over an AWGN channel. Bit-Error-Rate (BER) is measured and looked at by utilizing mistake adjusting codes like, Convolutional Code and Parallel Concatenated Convolutional Turbo Code. A decent BER for sound is thought to be 10-4. Utilizing turbo coding, it is almost accomplished with an Eb/No of 3 dB. A coding increase of about 6 dB is accomplished utilizing turbo coding, when contrasted with convolutional coding, at an expense of high computational many-sided quality. Likewise, reenactment is done on low speed and rapid PCs and edge preparing time is measured as a section of study on Quality of Service (QoS). It is demonstrated that, slightest complex turbo code obliges 3 to 4 times the preparing time taken by most astounding complex convolutional code. In this way coding increase is accomplished utilizing turbo code by trading off on computational time required.

The burdens of the customary codes like convolutional codes is that, with an end goal to approach the hypothetical point of confinement for Shannon's channel limit, we require to expand the imperative length of a convolutional code, which, thus, causes the computational intricacy of a most extreme probability decoder to increments exponentially. At last we achieve a point where unpredictability of the decoder is high to the point that it gets to be hard to figure it out physically furthermore there is no impressive diminishment in BER. Further coding addition is conceivable with turbo codes with sensible coding and deciphering multifaceted nature. In this paper, it is intended to legitimize these conclusions with reenactments. The channel chose just presents Gaussian clamor. In any case, issues confronted by blurring and multipath can be examined by picking other channel models. Rather than QPSK adjustment conspire, the same framework can be examined utilizing other tweak procedures like DQPSK also, QAM. Rather than parallel linked convolutional turbo codes, serial connected convolutional turbo code additionally can be executed and examined.

VI. REFERENCES

[1] Wolfgang Hoeg, Thomas Lauterbach, “Digital Audio Broadcasting-Principles and Applications of Digital Radio”, John Wiley & Sons, Ltd. England, 2003.

[2] Henrik Schulze, Christian Luders, “Theory and Applications of OFDM and CDMA”, John Wiley & Sons, Ltd. England, 2005.

[3] Simon Haykin, “Communication Systems”, 4th Edition, John Wiley & Sons, Inc. England, 2001.

[4] Bernad Sklar, "Digital Communications– Fundamentals and Applications", 2nd Edition Pearson Education (Singapore) Pte. Ltd., 2001

[5] Zimmerman G., Rosenberg M. and Dostert S., "Theoretical Bit Error Rate for Uncoded and Coded Data Transmission in Digital Audio Broadcasting", Proc. IEEE International Conference on Communications, Vol. 1, pp. 297-301, June 1996.

[6] Thomas May and Hermann Rohling, "Turbo Decoding of Convolutional Codes in Differentially Modulated OFDM Transmission System", Proc. IEEE 49th Vehicular Technology Conference, Vol. 3, pp-1891-1895, July 1999.

[7] Hanjong Kim, "Turbo Coded Orthogonal Frequency Division Multiplexing for Digital Audio Broadcasting", Proc. IEEE International Conference on Communications, Vol. 1, pp. 420-424, 2000.

[8] A. Burr, "Turbo-Codes: The Ultimate Error Control Codes?", Electronics & Communication Engineering Journal, Vol. 13, pp. 155-165, August 2001.

[9] R. Maerkle and C-E. W. Sudenberg, "Turbo Codes for Hybrid In Band On Channel Digital Audio Broadcasting", Porc. IEEE Global Telecommunication Conference, Vol. 2, pp. 941-945, 2001.

[10] Yan Li, Sumei Sun, "Design of Bit-Interleaved Turbo Coded Modulation in OFDM-Based Wireless LAN Systems", Proc. 4th International Workshop on Mobile and Wireless Communications Network, pp. 549-553, 2002.

[11] Lukas M. Gaetzi and M. O. J. Hawkaford, "Performance Prediction of DAB Modulation and Transmission Using Matlab Modeling", Proc. IEEE International Symposium on Consumer Electronics", pp. 272-277, September 2004.

[12] Hector Uhalte Bilbao, "Digital Transmission System Simulation", Linkoping Institute of Technology, Master Thesis August 2004.

[13] Seong Chul Cho, Jin Up Kim, Kyu Tae Lee and Kyoung Rok Cho, "Convolutional Turbo Coded OFDM/TDDMobile Communication System for High Speed Multimedia Services", Proc. Advanced Industrial Conference on Telecommunications, pp. 244-248, July 2005.

[14] Ibrahim S. Raad and Mehmet Yakan "Implementation of A Turbo Codes Test Bed in the Simulink Environment", Proc. of the Eighth International Symposium on Signal Processing and its Applications, Vol. 2, pp. 847-850, August 2005.

[15] Jorge Ortin, Paloma Garcia., Fernando Gutierrez and Antonio Valdovinos, "Performance Analysis of Turbo Deoding Algorithms in Wireless OFDM Systems", IEEE Transactions on Consumer Electronics, Vol. 55 pp. 1149- 1154, 2009.

[16] Mohammadnia-Avval M., ChrisSnow C and Lutz Lampe, "Error-Rate Analysis for Bit-Loaded Coded MIMOOFDM", IEEE Transactions on Vehicular Technology, Vol. 59, pp. 2340-2351, 2010.

