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Bit Error Rate Analysis of Coded OFDM for Digital Audio Broadcasting System, Employing Parallel Concatenated Convolutional Turbo Codes

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Abstract

In this paper we present a study of Bit Error Rate (BER), for Digital Audio Broadcasting (DAB) system, employing Coded OFDM with different channel coding schemes. Analysis is carried out for convolutional coded and turbo coded data in an Additive White Gaussian Channel (AWGN) based on different constraint lengths and code generator polynomials used for coding. A comparative study on the computational complexity is also done by applying an audio signal and measuring the data processing time per frame, on computers with different processor speeds. It is shown that a coding gain of approximately 6 dB is achieved using turbo coding when compared to convolutional coding, at a cost of higher computational complexity.

Keywords - ***DAB; OFDM; Convolutional Codes; Turbo Codes.***

INTRODUCTION

The requirement of mobility while connected to network is fueling the growth of wireless communication. The conventional analog transmission techniques do not perform well in mobile environment, since suitable techniques to mitigate the effects of multipath propagation induced fading have not been developed for these systems. Orthogonal Frequency Division Multiplexing (OFDM) is one such technique to combat the effect of multipath fading, frequency selective fading and Inter symbol Interference (ISI) [1]. OFDM decreases the amount of hardware implementation since multiplexing and filtering operations can be performed by employing the Fast Fourier Transform (FFT). This eliminates the need to have multiple oscillators at the transmitter and synchronizing loops at the receiver. Due to the cyclic extension of signal period into a guard interval, OFDM system is suitable for Single Frequency Networks (SFN) [5].

In this paper an OFDM application standard called Digital Audio Broadcasting (DAB) system model is implemented in Matlab/Simulink environment. The performance of this system over a channel perturbed by AWGN noise is studied. Coded Orthogonal Frequency Division Multiplexing (COFDM) technique is studied in which convolutional codes and turbo codes are employed and computed the resulting bit error rates (BER). The variation in BER is analyzed based on different coding parameters. An audio signal is transmitted and data processing time per frame is measured and compared for different channel coding schemes.

SYSTEM MODEL OF DAB USING CODED OFDM

Simplified DAB Block Diagram

A general block diagram of the Digital Audio Broadcasting transmission system is shown in Fig. 1. The analog signal is encoded and applied to channel encoder. After channel coding the bit streams are QPSK mapped. The data is then passed to OFDM generator. The high data rate bit stream is divided into 'N' parallel

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data streams of low data rate and individually modulated on to orthogonal subcarriers which is realized using IFFT algorithm. Orthogonality of the subcarriers helps to achieve zero Inter Symbol Interference, theoretically [1]. Finally, the OFDM symbol is provided with cyclic prefix and the completed DAB frame structure is transmitted through an AWGN channel.

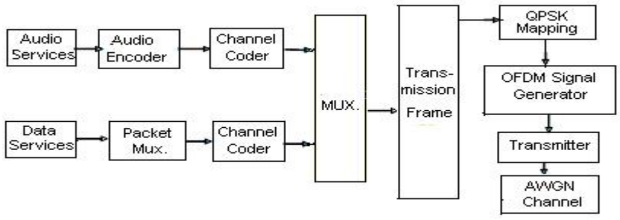

Figure 1. DAB transmitter – Block Diagram.

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.

TABLE I. DAB TRANSMISSION MODES

Trans - mission Mode	No. of Sub-carriers	Sub carrier spacing	FFT Length	Maximum Radio Frequency
TM I	1536	1 KHz	2048	§ 375 MHz
TM II	384	4 KHz	512	§ 1.5 GHz
TM III	192	8 KHz	256	§ 3 GHz
TM IV	768	2 KHz	1024	§ 750 MHz

Coded OFDM Block Diagram

Coded OFDM, or COFDM, is a term used for a system in which the error control coding and OFDM modulation processes work closely together.

An important step in a COFDM system is to interleave and code the bits prior to the IFFT. This step serves the purpose of taking adjacent bits in the source data and spreading them out across multiple subcarriers.

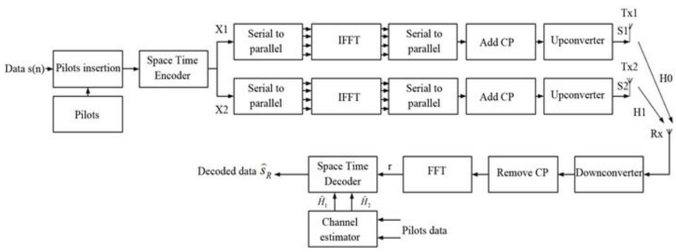

Figure 2. Coded OFDM – Block Diagram

Idealized system model OFDM

This section describes a simple idealized OFDM system model suitable for a time-invariant AWGN channel.

Transmitter

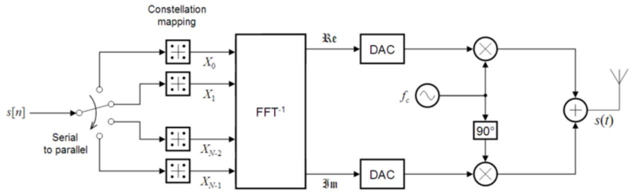

Figure 3. OFDM transmitter – Block Diagram

An OFDM carrier signal is the sum of a number of orthogonal sub-carriers, with baseband data on each sub-carrier being independently modulated commonly using some type of quadrature amplitude modulation (QAM) or phase-shift keying(PSK). This composite baseband signal is typically used to modulate a main RF carrier.

S[n] is a serial stream of binary digits. By inverse multiplexing, these are first demultiplexed into N parallel streams, and each one mapped to a (possibly complex) symbol stream using some modulation constellation (QAM, PSK, etc.). Note that the constellations may be different, so some streams may carry a higher bit-rate than others.

An inverse FFT is computed on each set of symbols, giving a set of complex time-domain samples. These samples are then quadrature-mixed to passband in the standard way. The real and imaginary components are first converted to the analogue domain using digital-to-analogue converters (DACs); the analogue signals are then used to modulate cosine and sine waves at the carrier frequency, fc, respectively. These signals are then summed to give the transmission signal,

Receiver

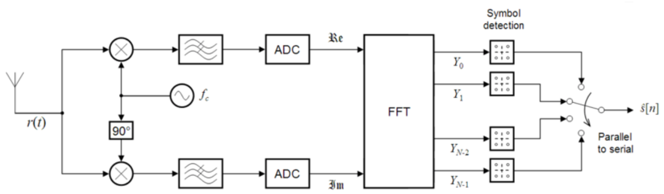

Figure 4. OFDM Receiver– Block Diagram

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The receiver picks up the signal r(t), which is then quadrature-mixed down to baseband using cosine and sine waves at the carrier frequency. This also creates signals centered on 2fc, so low-pass filters are used to reject these. The baseband signals are then sampled and digitised using analog-to-digital converters (ADCs), and a forward FFT is used to convert back to the frequency domain. This returns N parallel streams, each of which is converted to a binary stream using an appropriate symbol detector. These streams are then re-combined into a serial stream, S[n],which is an estimate of the original binary stream at the transmitter.

CHANNEL CODING

Convolutional Encoding & Viterbi Decoding

A convolutional encoder consists of an M-stage shift register with 'k' inputs, prescribed connections to 'n' modulo-2 adders and multiplexer that serializes the outputs of the adders. Here the encoder selected has k=1, ie; the input sequence arrives on a single input line. Hence the code rate is given by r = 1/n. In an encoder with an M-stage shift register, the memory of the coder equals M message bits and K = (M+1) shifts are required before a message bit that has entered the shift register can finally exit. This parameter K is referred to as the constraint length of the encoder.

The channel coding used for standard DAB consists of code rate ½, memory 6, convolutional code with code generator polynomials 133 and 171 in octal format [2]. For DAB lower code rates give better performance. Hence in this work, encoder with code rate=w̅ is selected. One such convolutional encoder is shown in Fig. 2. The number of registers=6. Hence the constraint length K=7. Generator Polynomials are 171, 133 and 115 in octal format. Simulation is carried out for various values of constraint length and generator polynomials, which are given in Table-III.

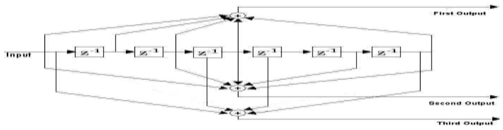

Figure 5. A rate w̅ convolutional encoder with constraint length, K=7.

The Viterbi algorithm operates by computing a metric for every possible path in the trellis [4]. The path with the lower metric is retained and the other path is discarded. This process is continued until the algorithm completes its forward search through the trellis and reaches the termination node, and makes a decision on maximum likelihood path. The sequence symbols associated with the path are then released to the destination as the decoded output.

Parallel Concatenated Convolutional Turbo Coding & Decoding

Parallel Concatenated Convolutional turbo code (PCC turbo code) consists of two or more Recursive Systematic Convolutional (RSC) coders working in parallel [8]. The purpose of interleaver is to offer each encoder a random version of the information resulting in parity bits from each RSC that are independent.

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 provide same code rate for turbo encoder as in the case of convolutional encoder, a parallel concatenation of two identical RSC encoders are used which gives a code rate of w̅. One such turbo encoder is shown in Fig. 3, where the number of registers in each RSC encoder=2. Hence the constraint length K=3. Generator polynomials are 7 and 5 in octal format. The number 7 denotes the feedback polynomial. ʌ is the random interleaver. Simulation is carried out for various values of constraint length, generator polynomials and feedback polynomials, which are given in Table-III.

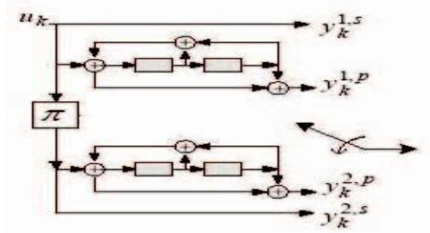

Figure 6. A rate w̅ turbo encoder with 2 parallel recursive systematic convolutional encoders, each with constraint length, K=3.

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The inputs are information bits and called u_k. The outputs are code bits. Of these, the output of first encoder, $y_k^{1,s}$ is called the systematic bit, and it is the same as the input bit. The second output bit, $y_k^{1,p}$ is the first parity bit which is recursive systematic bit. An interleaver, denoted by ᴧ, is placed in between the two encoders to ensure that the data received by the second encoder is statistically independent. The third output bit, $y_k^{2,p}$ is the second parity bit which is also a recursive systematic bit. The fourth output $y_k^{2,s}$ is deterministically reshuffle version of $y_k^{1,s}$, 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.

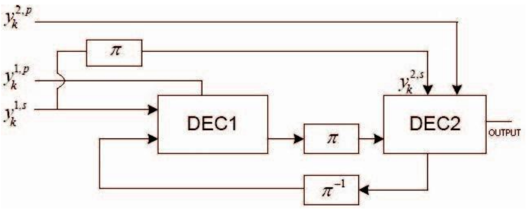

Figure 7. Turbo decoder – Block Diagram.

In Fig. 7, DEC1 and DEC2 are 2 APP decoders. ᴧ and $ᴧ^{-1}$ are random interleaver and deinterleaver respectively [14]. The symbol vector sent for each time are described by $y_k=(y_k^{1,s}, y_k^{1,p}, y_k^{2,p})$. The goal is to take these and make a guess about the transmitted vector and hence code bits which in turn decode u_k, the information bit.

Basic block diagram PAC

- upper panel: encoder
- lower panel: decoder

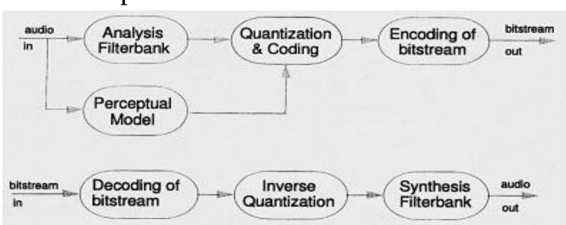

Figure 8. PAC – Block Diagram

Filter bank

- Used to decompose an input signal into subbands or spectral components (time-frequency domain)

Perceptual model (aka psychoacoustic model)

- usually analyzes the input signal instead of the filterbank outputs (time-domain input provides better time and frequency resolution)
- computes signal-dependent masked threshold based on psychoacoustics

Quantization and coding

- spectral components are quantized and encoded
- goal is to keep quantization noise below the masked threshold

Frame packing

- Bitstream formatter assembles the bitstream, which typically consists of the coded data and some side information

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.

TABLE II. SIMULATION PARAMETERS

Transmission Mode	Mode I
No. of sub-carriers	1536
Transmission frame duration (Ft)	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 1/3), Turbo coding (rate 1/3), PAC (rate 1/3)
Modulation	QPSK
Channel	AWGN

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TABLE III. CHANNEL CODING PARAMETERS

Channel	*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	--
	3	7, 6, 5	--
Turbo Coding	3	7, 5	7
	4	15, 13	15
	5	34, 27	34
	6	71, 57	71
	7	171, 133	171
PAC	3	7,5	7

B. Simulation Block Diagram

Fig. 9 and Fig. 10 show the simulation models for DAB using convolutional coding and turbo coding respectively. Simulations are carried out using Matlab/Simulink.

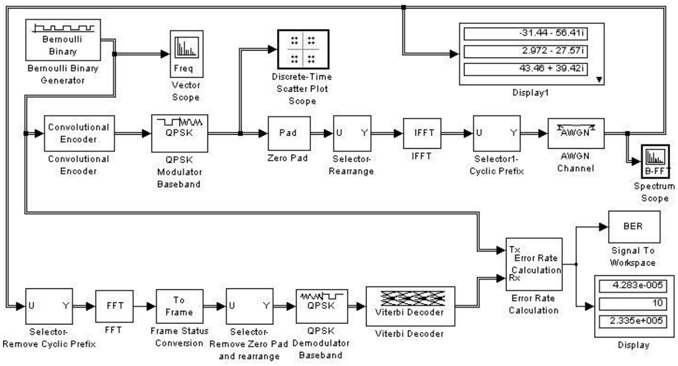

Figure 9. Simulation model for DAB transceiver using convolutional coding & viterbi decoding.

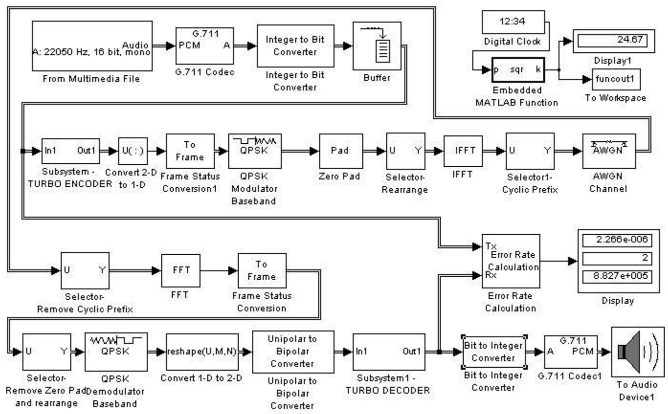

Figure 10. Simulation model for DAB transceiver using turbo coder and decoder. An audio signal is inputed and analysed.

SIMULATION RESULTS AND DISCUSSION

A. Bit-Error-Rate (BER) Analysis

Bit-Error-Rate (BER) is measured and plotted for uncoded, convolutional coded and turbo coded simulations of DAB system. Simulation and analysis is done on the basis of different code generator polynomials, having different constraint lengths.

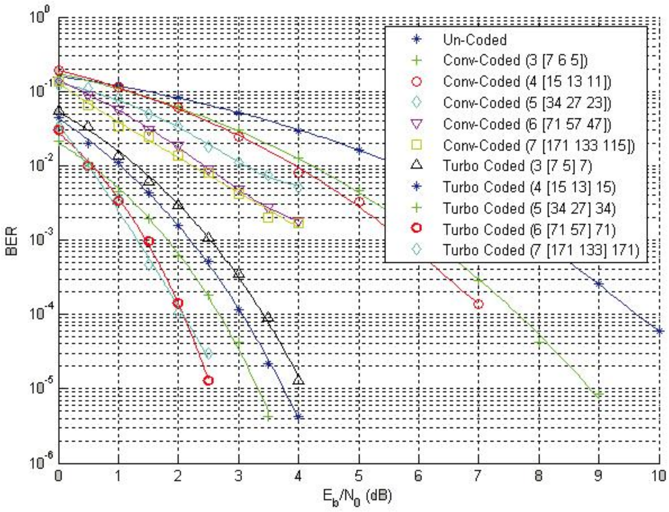

Figure 11. BER simulation results for convolutional coded and turbo coded DAB, with code rate=w̄, in AWGN channel.

From Fig. 11, we can see that, a coding gain of nearly 6 dB is achieved using turbo coding when compared to convolutional coding. A good BER for audio is considered to be 10^{-4}. Using turbo coding, it is nearly achieved with an E_b/N_o of 3 dB.

B. Frame Processing Time Comparison

As a part of study on Quality of Service (QoS), the transmission frame processing time is also measured. In DAB, data bits are grouped together to form frames, then processed and transmitted. Here 10^6 or 10^7 data bits are transmitted for each simulation. Frame processing time can be calculated in MATLAB by dividing the simulation time with the number of frames transmitted. The frame processing time taken by the DAB system using both coding schemes with different constraint lengths is measured and given in Table-IV. Simulation is carried out on computers with different processor speeds and memory capacity.

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A comparison chart is prepared and graphically represented in Fig. 8. The notation, Conv_CnstrLn3 means channel coding used is the convolutional code with constraint length K=3.

TABLE IV. FRAME PROCESSING TIME COMPARISON

Coding Scheme	Frame Processing computer (milli seconds) time on low speed	Frame Processing time on high speed computer (milli seconds)
Conv_CnstrLn3	3.93	2.59
Conv_CnstrLn4	4.18	2.81
Conv_CnstrLn5	4.6	3.13
Conv_CnstrLn6	5.51	3.86
Conv_CnstrLn7	7.07	5.18
Turbo_CnstrLn3	28.36	16.79
Turbo_CnstrLn4	44.43	27.66
Turbo_CnstrLn5	77.89	50.2
Turbo_CnstrLn6	142.5	92.57
Turbo_CnstrLn7	270.9	180.3

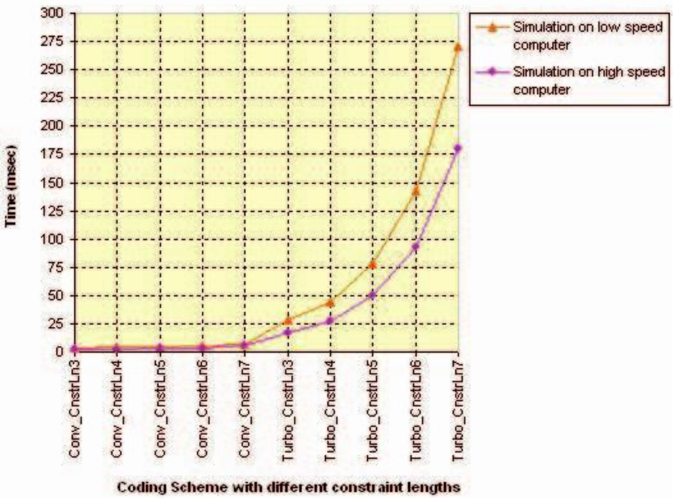

Figure 12. Comparison chart of frame processing time for DAB.

1) Analysis on low speed computer (Intel Pentium-4 CPU, 1.6 GHz single processor, 256 MB RAM)

From Table-IV, we can see that frame processing time taken by highest complex convolutional code (with K=7) = 7.07 m sec and frame processing time taken by least complex turbo code (with K=3) = 28.36 msec.
Performance factor = 28.36 / 7.07 = 4.01

Hence, as shown in Fig. 7, coding gain is achieved using turbo code by compromising to nearly 4 times the computational time needed for convolutional code.

2) Analysis on high speed computer (Intel Pentium-4 CPU, 2.66 GHz single processor, 1 GB RAM)

From Table-IV, we can see that frame processing time taken by highest complex convolutional code (with K=7) = 5.18 msec and frame processing time taken by least complex turbo code (ie; with K=3) = 16.79 msec.Performance factor = 16.79 / 5.18 = 3.24

Hence, as shown in Fig. 7, coding gain is achieved using turbo code by compromising to nearly 3 times the computational time needed for convolutional code.

The large value of frame processing time is due to the fact that, here a computer simulation study is done where the same computer is doing all the jobs like, coding, modulation, transmission, reception, demodulation, decoding, etc. In real time scenario, separate transmission and reception systems are implemented in hardware using high speed processors. Hence the Quality of Service can be achieved with in the limit specified by industrial standard.

C.PAC Graph.

New Joint Multiple Program Encoding Technique in the context of the perceptual audio coding (PAC) type of algorithms.

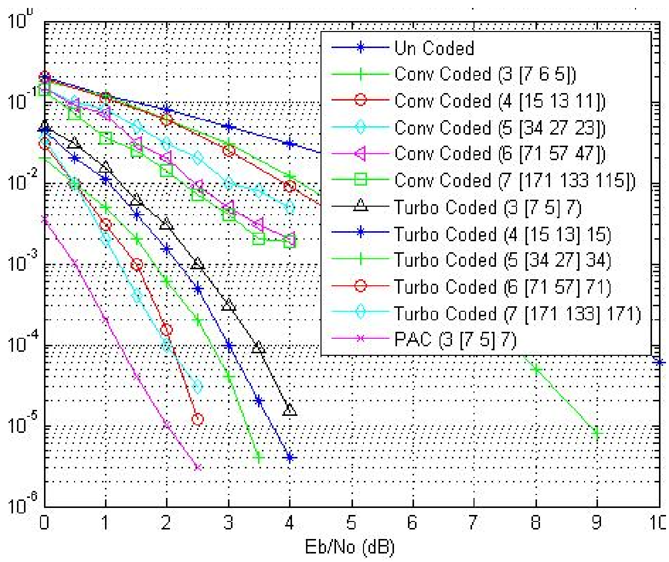

Figure 13. BER simulation results for convolutional and turbo coded DAB, with code rate=1/3, in AWGN channel.

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CONCLUSION AND FUTURE WORK

Digital Audio Broadcasting system using Coded OFDM is implemented and studied over an AWGN channel. Bit-Error-Rate (BER) is measured and compared by employing error correcting codes like, Convolutional Code and Parallel Concatenated Convolutional Turbo Code. A good BER for audio is considered to be 10^{-4}. Using turbo coding, it is nearly achieved with an E_b/N_o of 3 dB. A coding gain of nearly 6 dB is achieved using turbo coding, when compared to convolutional coding, at a cost of high computational complexity.

Also, simulation is done on low speed and high speed computers and frame processing time is measured as a part of study on Quality of Service (QoS). Thus coding gain is achieved using turbo code by compromising on computational time required.

The disadvantages of the traditional codes like convolutional codes is that, in an effort to approach the theoretical limit for Shannon's channel capacity, we need to increase the constraint length of a convolutional code, which, in turn, causes the computational complexity of a maximum likelihood decoder to increases exponentially. In this paper, it is aimed to justify these conclusions with simulations.

The channel selected only introduces Gaussian noise. But problems faced by fading and multipath can be analyzed by choosing other channel models. Instead of QPSK modulation scheme, the same system can be analyzed using other modulation techniques like DQPSK and QAM. Instead of parallel concatenated convolutional turbo codes, serial concatenated convolutional turbo code also can be implemented and analyzed.

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