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High Speed Low Bit Error Rate Digital Audio Broadcasting System Using Parallel Concatenated Convolution Turbo Codes

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Abstract: *The major parameters to analyze the performance Digital Audio Broadcasting (DAB) system is Speed and Bit Error Rate. Orthogonal Frequency Division Multiplexing (OFDM) and Multiple Input and Multiple Output (MIMO) techniques combination gives 4th Generation Long Term Evolution (LTE). Using multiple carriers in OFDM provides higher level of spectral efficiency as compared to Frequency Division Multiplexing (FDM). In OFDM because of loss of orthogonality between the subcarriers there is inter carrier interference (ICI) and inter symbol interference (ISI) and to overcome this problem use of cyclic prefixing (CP) is required, which uses 20% of available bandwidth. 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 and Coded Orthogonal Frequency Division Multiplexing (COFDM) technique is studied in which convolution codes and turbo codes are employed and computed the resulting bit error rates (BER) then summarize the parallel concatenated convolution turbo codes is improves the performance of DAB.*

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I. Introduction

A set of requirements are specified for 4th generation of cellular systems by International Telecommunication Union Radio communication Sector (ITU-R). The requirement of data rate was specified in International Mobile Telecommunications Advanced project (IMT-Advanced). 3rd Generation Partnership Project (3GPP) was established in 1998 [1]. 3GPP started working on the LTE project to define the Radio Access Network (RAN) and core network [1]. 3GPP's candidate for 4G was LTE-Advanced. 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 Intersymbol Interference (ISI). 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.

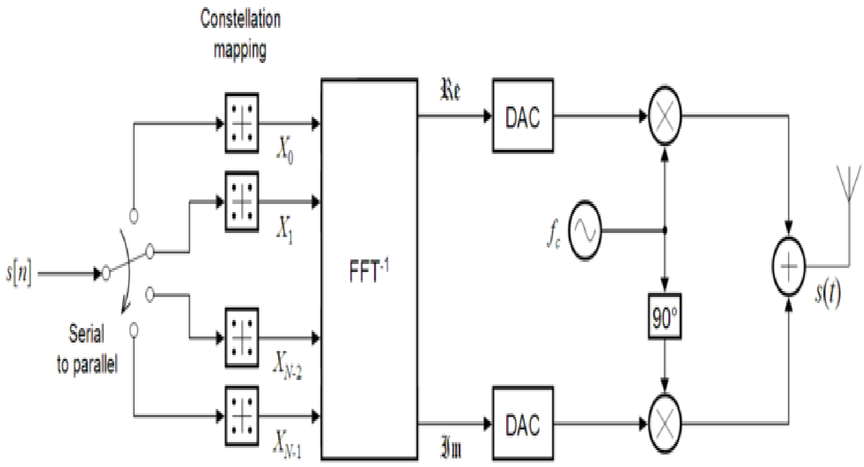
Wolfgang Hoeg, Thomas Lauterbach,**etal, (2003),**The new digital broadcast system family is very different from existing conventional broadcast systems. It is standardized in a large number of documents (from ITU-R, ISO/IEC, ETSI, EBU, and others) which are often difficult to read.

BernadSklar, et al, (2001),The clear, easy-to-understand introduction to digital communications Completely updated coverage of today's most critical technologies Step-by-step implementation coverage Trellis-coded modulation, fading channels, Reed-Solomon codes, encryption, and more Exclusive coverage of maximizing performance with advanced "turbo codes" "This is a remarkably comprehensive treatment of the field, covering in considerable detail modulation, coding (both source and channel), encryption, multiple access and spread spectrum. It can serve both as an excellent introduction for the graduate student with some background in probability theory or as a valuable reference for the practicing ommunication system engineer. For both communities, the treatment is clear and well presented." – Andrew Viterbi, The Viterbi Group Master every key digital communications technology, concept, and technique. Digital Communications, Second Edition is a thoroughly revised and updated edition of the field's classic, best-selling introduction. With remarkable clarity, Dr. Bernard Sklar introduces every digital communication technology at the heart of today's wireless and Internet revolutions, providing a unified structure and context for understanding them -- all without sacrificing mathematical precision. Sklar begins by introducing the fundamentals of signals, spectra, formatting, and baseband transmission. Next, he presents practical coverage of virtually every contemporary modulation, coding, and signal processing technique, with numeric examples and step-by-step implementation guidance.

The DAB is digital radio technology for broadcasting radio stations, used in several countries across Europe and Asia Pacific. The DAB standard was initiated as a European research project in the 1980s.[1] The Norwegian Broadcasting Corporation(NRK) launched the very first DAB channel in the world on 1 June 1995 (NRK Klassisk),[2] and the BBC and SR launched their first DAB digital radio broadcasts in September 1995. DAB receivers have been available in many countries since the end of the 1990s. DAB may offer more radio programmes over a specific spectrum than analogue FM radio. An upgraded version of the system was released in February 2007, which is called **DAB+**. DAB is not forward compatible with DAB+, which means that DAB-only receivers are not able to receive DAB+ broadcasts.[6] However, broadcasters can mix DAB and DAB+ programs inside the same transmission and so make a progressive transition to DAB+. DAB+ is approximately twice as efficient as DAB due to the adoption of the AAC+ audio codec, and DAB+ can provide high quality audio with bit rates as low as 64 kbit/s.[7] Reception quality is also more robust on DAB+ than on DAB due to the addition of Reed-Solomon error correction coding. In spectrum management, the bands that are allocated for public DAB services, are abbreviated with **T-DAB**, where the "T" stands for terrestrial. More than 20 countries provide DAB transmissions, and several countries, such as Norway, Australia, Italy, Malta, Switzerland, The Netherlands and Germany,[8] are transmitting DAB+ stations. See Countries using DAB/DMB. In many countries it is expected that DAB will gradually replace FM radio. Norway was the first country to announce national FM radio switch-off starting from 2017.

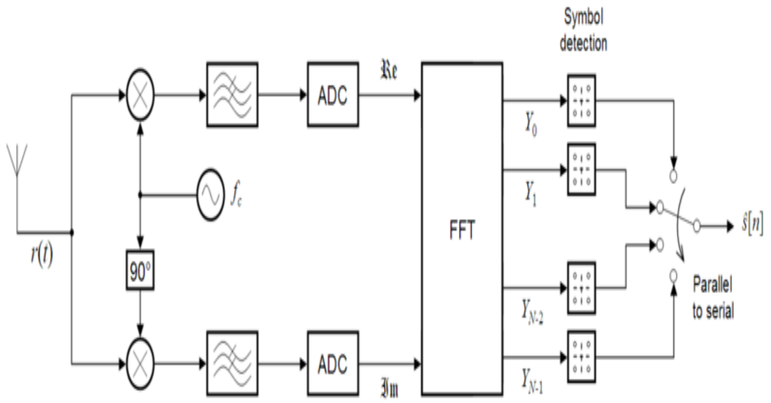
II. Related work

Transmitter



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, f_c, respectively. These signals are then summed to give the transmission signal s(t).

Receiver

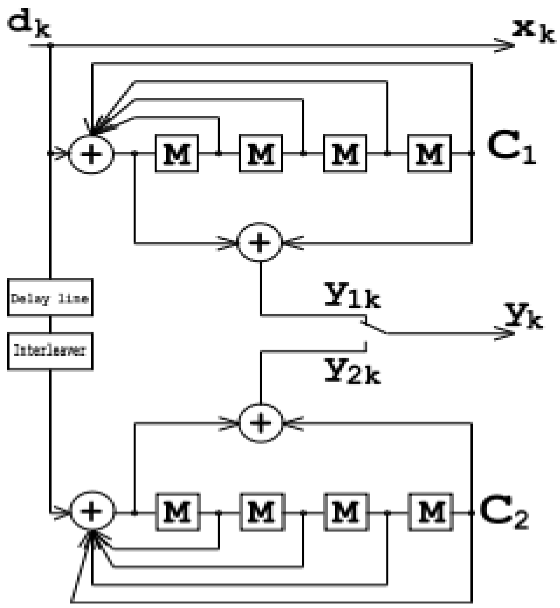


The receiver picks up the signal r(t) , which is then quadrature-mixed down to baseband using cosine and sine waves at the carrier frequency. This also creates signals centered on $2f_c$, so low-pass filters are used to reject these. The baseband signals are then sampled and digitised using analog-to-digital 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 $\hat{s}[n]$, which is an estimate of the original binary stream at the transmitter. If N sub-carriers are used, and each sub-carrier is modulated using M alternative symbols, the OFDM symbol alphabet consists of M^N combined symbols. The low-pass equivalent OFDM signal is expressed as:

$$f(t) = \sum_{n=1}^{N-1} X_n e^{j2\pi nt/T}$$

Where $\{x_k\}$ are the data symbols, N is the number of sub-carriers, and T is the OFDM symbol

This encoder implementation sends three sub-blocks of bits. The first sub-block is the *m*-bit block of payload data. The second sub-block is *n/2* parity bits for the payload data, computed using a recursive systematic convolutional code (RSC code). The third sub-block is *n/2* parity bits for a known permutation of the payload data, again computed using an RSC code. Thus, two redundant but different sub-blocks of parity bits are sent with the payload. The complete block has $m + n$ bits of data with a code rate of $m/(m + n)$. The permutation of the payload data is carried out by a device called an interleaver. Hardware-wise, this turbo-code encoder consists of two identical RSC coders, C_1 and C_2, as depicted in the figure, which are connected to each other using a concatenation scheme, called *parallel concatenation*:



In the figure, *M* is a memory register. The delay line and interleaver force input bits d_k to appear in different sequences. At first iteration, the input sequence d_k appears at both outputs of the encoder, x_k and y_{1k} or y_{2k} due to

the encoder's systematic nature. If the encoders C_1 and C_2 are used respectively in n_1 and n_2 iterations, their rates are respectively equal to
The decoder is built in a similar way to the above encode. Two elementary decoders are interconnected to each other, but in serial way . the decoder operates on lower speed thus it is intended for the encoder and is for correspondingly. Yields a soft decision which causes delay. The same delay is caused by the delay line in the encode.
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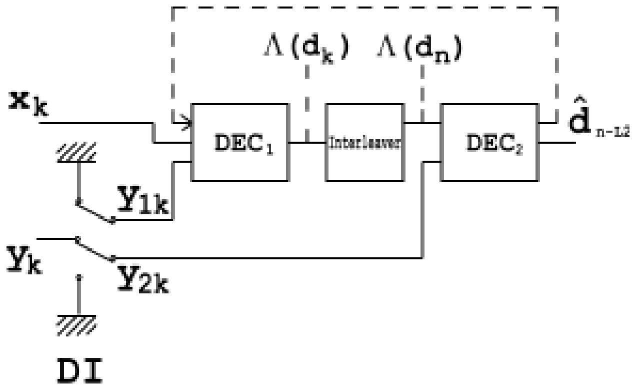


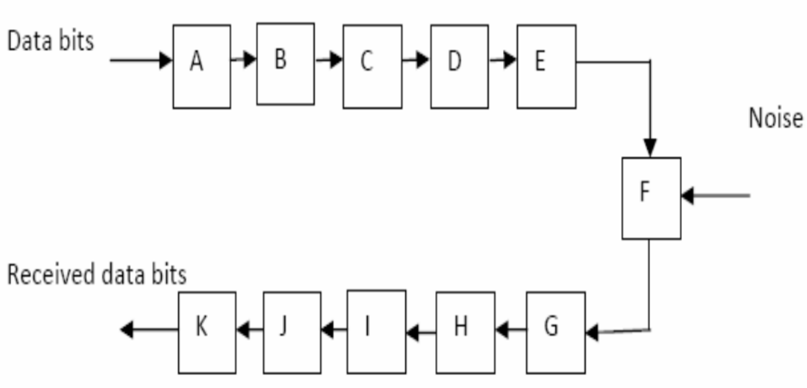
An interleaver installed between the two decoders is used here to scatter error bursts coming from output. DI blocks is a demultiplexing and insertion module, it works as a switch redirecting inputs bits to at one moment and to at another. In OFF state, it feeds both and inputs with padding bits.

III. Model

Introduction An OFDM system was modeled using Matlab to allow various parameters of the system to be varied and tested. The aim of doing the simulations was to measure the performance of OFDM under AWGN channel and RAYLEIGH channel conditions, for different modulation schemes like BPSK, QPSK used in IEEE 802.11a wireless LAN standard. Following this introduction, section 5.2 discusses model used in simulation, steps in OFDM simulation, modulation schemes and their constellation diagrams. Section 5.3 presents the parameters used in simulation. Section 5.4 provides the simulation results of OFDM system for different modulation schemes. It also shows the results to compare the performance of OFDM using coded and uncoded OFDM 5.2 simulation model Since the main goal of this thesis was to simulate the COFDM system by utilizing turbo code.

The block diagram of the entire system is shown in above Figure Simulation model of turbo coded OFDM Here A = turbo encoder, B = BPSK/QPSK modulation, C = serial to parallel converter, D = IFFT, E = parallel to serial convertor, F = channel with noise, G = serial to parallel Converterr, H = FFT, I = parallel to serial converterr, J = BPSK/QPSK demodulation and K = turbo decoder.

simulation parameters



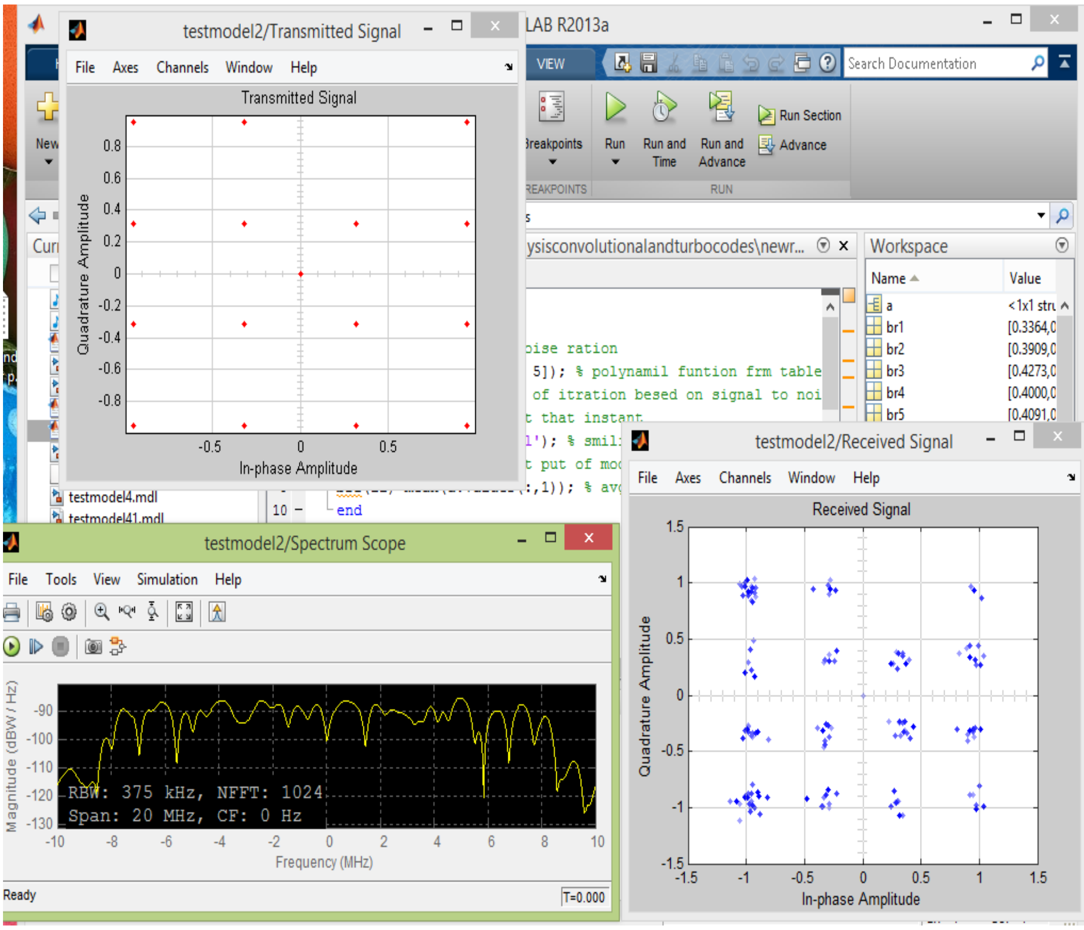
Digital Modulation : BPSK QPSK, QAM Turbo code rates : 1/3 SISO Decoder :Log-MAP Code Generator : {111, 101} Interleaver : pseudo random interleaver We measured the performance of the turbo coded OFDM through

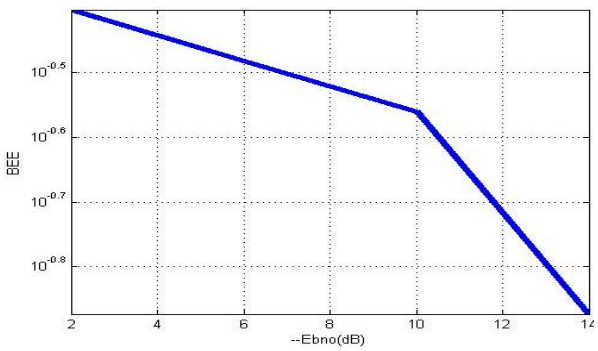
Algorithm

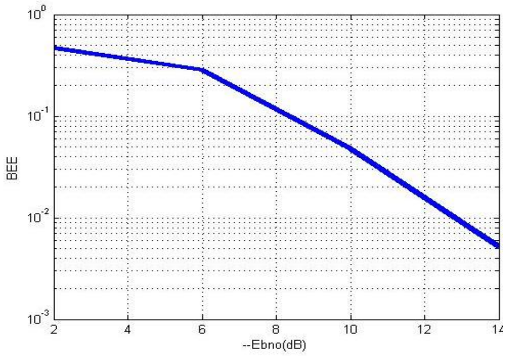
1. Generate the information bits randomly. 2. Encode the information bits using a turbo encoder with the specified generator matrix. 3. Use QPSK or different QAM modulation to convert the binary bits, 0 and 1, into complex signals (before these modulation use zero padding) 4. Performed serial to parallel conversion. 5. Use IFFT to generate OFDM signals, zero padding is being done before IFFT. 6. Use parallel to serial convertor to transmit signal serially. 7. Introduce noise to simulate channel errors. We assume that the signals are transmitted over an AWGN (Additive White Gaussian Noise) and Rayleigh channel. 8. At the receiver side, perform reverse operations to decode the received sequence. 9. Count the number of erroneous bits by comparing the decoded bit sequence with the original one. 10. Calculate the BER and plot it.

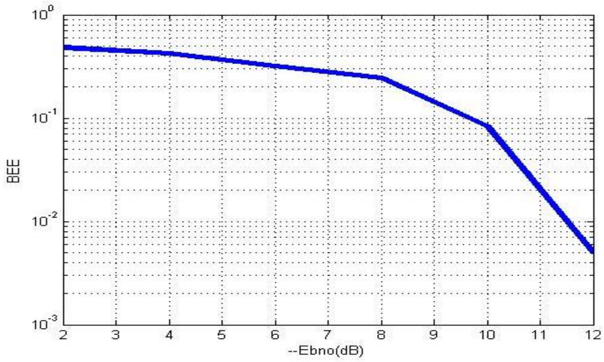
IV. Result

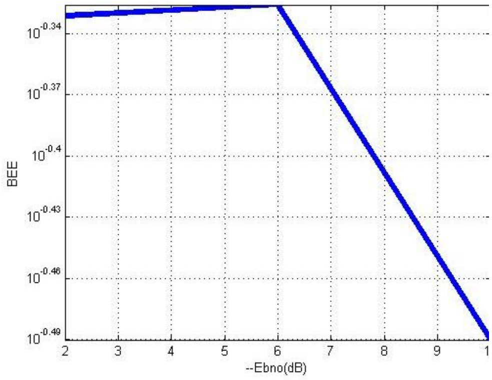
All the simulations are done to achieve BER at .For simulation results two channel are AWGN and RAYLEIGH are used. The BER performance of TCOFDM system is compared with uncoded OFDM system. As mentioned before, bursty errors deteriorate the performance of the any communications system. The burst errors can happen either by impulsive noise or by deep frequency fades.To improve the performance of this system FEC code can be used. Convolution code is a good example of FEC code. Convolution coding in OFDM can give performance improvement of some 5 db on AWGN channel over the uncoded OFDM system at required BER. Further improvement in the performance can be obtained by applying turbo coding instead of convolution code. The turbo code gives better performance at low SNR. The BER performance of TCOFDM system is compared with the respective uncoded system under the fading AWGN channel and RAYLEIGH fading channel.

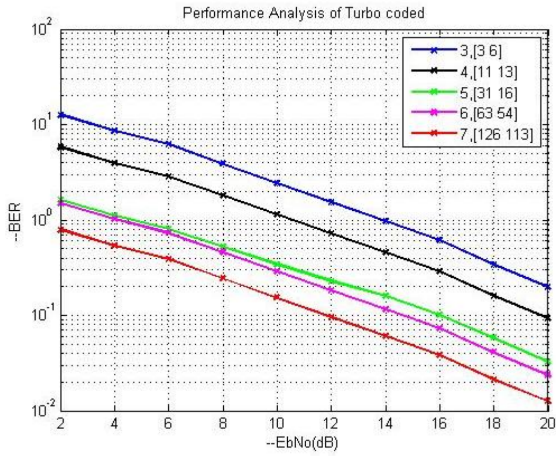












V. Conclusion

To conclude, this major project gives the detail knowledge of a current key issue in the field of communications named Orthogonal Frequency Division Multiplexing (OFDM). We focused our attention on turbo codes and their implementation. We described the encoder architecture. In our case, the code is the result of the parallel concatenation of two identical RSCs. The code can be punctured in order to fulfill bit rate requirements. The decoder succeeded in its duty thanks to the decoding algorithms that it is built around. We focused mainly on the study of the MAP. We discovered that the power of the scheme came from the two individual decoders performing the MAP on interleaved versions of the input. Each decoder used information produced by the other as a priori information and outputted a posteriori information. We elaborated on the performance theory of the codes Then we tied concepts of OFDM and turbo coding with a target-based, modulation scheme. First I developed an OFDM system model then try to improve the performance by applying forward error correcting codes to our uncoded system. From the study of the system, it can be concluded that we are able to improve the performance of uncoded OFDM by convolutional coding scheme. Further improvement on the performance has been achieved by applying turbo coding to uncoded OFDM system. Turbo codes with low order decoding iterations have been evaluated. The SNR performance for BER 10^{-2} and 10^{-4} , that are suitable for speed and data applications, are analyzed. As a result, the TCOFDM system with least number of decoding iterations, 3 to 5 iterations are shown to be sufficient to provide good BER performance.

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