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디지털 오디오 방송을 위한 터보 부호화된 OFDM

(Turbo Coded OFDM for Digital Audio Broadcasting System)

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요 약

본 논문에서는 펑쳐드 콘볼루션 부호기와 연판정 비터비 복호기를 사용하는 기존의 COFDM DAB 시스템에서 부호 율이 1/4인 모체부호로부터 부호화 된 4비트 중 처음 한 비트는 펑쳐링되지 않고 언제나 전송된다는 사실에 근거하여 기존 COFDM DAB 시스템에서 정의된 펑쳐링 절차를 수정함이 없이 터보 부호를 적용한 TCOFDM(Turbo Coded OFDM) DAB 시스템 모델을 제안한다. COFDM DAB 시스템에 터보 부호기를 적용하기 위해 유효 자유거리가 최대인 터보 부호기를 설계하며, 기존의 펑쳐링 과정을 수정함이 없이 터보 부호기로 대체하기 위해 새로운 펑쳐링 과정을 정의한다. 또한 제안된 터보 부호기에 대한 복호기 구조를 제안하고 DAB 시스템에서 정의된 네 가지 전송모드 중에서 단일 주파수망(SFN) 방송 시스템 구성에 유리한 전송모드 I과 위성방송에 적합한 전송모드 III에 대해서 기존의 COFDM DAB 시스템과 제안된 TCOFDM DAB 시스템의 성능을 주파수 선택적 라이시안 페이딩 채널 및 주파수 선택적 레일레이 페이딩 채널 환경에서 컴퓨터 시뮬레이션을 통해 비교, 분석하여본다.

Abstract

The Pan-European Digital Audio Broadcasting(DAB) system's performance is characterized and improved with the aid of turbo codec. From the fact that the first bit among the four coded bits at the RCPC coding defined in the Eureka 147 DAB system is not punctured and always transmitted, this paper proposes a new turbo coded DAB system model that replaces the existing RCPC codec by a turbo codec without modifying the puncturing procedure and puncturing vectors defined in the standard DAB system for compatibility. The performance of a new system is compared to that of the conventional system under the Rician fading channel and the Rayleigh fading channel in conjunction with DAB transmission mode I and III suitable for the terrestrial single frequency network and satellite broadcasting.

I. INTRODUCTION

Although AM and FM sound broadcast can provide acceptable services to properly installed fixed receivers, they are not capable of delivering compact disc(CD) sound quality especially to vehicular receivers. In contrast to AM and FM radio, the new Digital Audio Broadcasting(DAB) service has to provide much better quality services to fixed, portable and vehicular receivers[1][2][3]. The European DAB system developed by the Eureka 147 project

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has been recommended as a digital sound broadcasting system standard in the ITU-R (International Telecommunication Union Radio sector). In order to allow DAB to be used in different transmission network configurations and over wide frequency range up to 3GHz, four transmission modes are specified[1][4].

The DAB system adopts Orthogonal Frequency Division Multiplexing(OFDM) for modulation scheme, which allows transmission over highly frequency selective channels at a low receiver implementation cost[4][5][6]. One of the advantages of OFDM is that it can convert a wideband frequency selective fading channel into a series of narrowband and frequency non-selective fading subchannels by using parallel and multicarrier transmission[7]. Because individual subchannels are affected by the flat fading, OFDM systems will always have to make use of an error correcting code in combination with interleaving to combat such fading on each subchannel. As a channel coding technique, the DAB system employs rate-compatible punctured convolutional(RCPC) codes which are described by the mother code, puncturing procedure and puncturing vectors. The channel coding and modulation scheme used in the Eureka 147 DAB system is known as Coded OFDM (COFDM), which is a combined technique of multicarrier transmission(OFDM) and convolutional coding(punctured convolutional coding) with Viterbi error correction[6][7].

Recently, a novel class of binary parallel concatenated recursive systematic convolutional codes termed turbo codes, having amazing error correcting capabilities and being therefore very attractive for application to digital mobile radio, was introduced[8][9][10][11][12]. Because each subchannel for OFDM systems is affected by the flat fading, it is necessary to mitigate the effects of flat fading by using a powerful error-correction code. To improve the system's performance at the cost of increased complexity, we suggest turbo-coding based improvement to the DAB system. This paper proposes a new turbo coded OFDM(TCOFDM) system that replaces the existing RCPC codec by a turbo codec without modifying the puncturing procedure and puncturing vectors defined in the Eureka 147 DAB system for compatibility. The performance of TCOFDM system is compared with that of standard COFDM system under the Rician fading channel and the Rayleigh fading channel in conjunction with terrestrial Single Frequency Network(transmission mode Ⅰ) and satellite (transmission mode Ⅲ) broadcasting.

The remainder of the paper is organized as follows. A succinct overview of DAB system model and turbo code is presented in Section Ⅱ, while the existing RCPC coding procedure for DAB is described in Section Ⅲ. Section Ⅳ describes a new turbo coded OFDM system, while our channel model is described in Section Ⅴ. Following this the performance of new turbo coded system is given by computer simulations in Section Ⅵ and Section Ⅶ summaries the conclusions of the paper

Ⅱ. DAB SYSTEM MODEL USING THE COFDM SCHEME

The DAB broadcast signal is arranged into a transmission frame which consists of three channels as shown in Fig. 1.

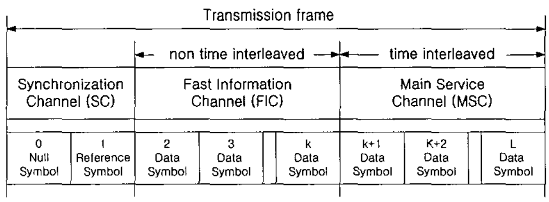


그림 1. DAB 전송 프레임
Fig. 1. DAB transmission frame.

The first is the synchronization channel(SC) that further comprises two symbols. One is the null symbol, which is a duration of no signal transmitted. It is used to perform coarse time(frame) synchronization by envelope detection of the signal[5].

The other is the phase reference symbol(PRS), which has fixed magnitudes and known phases on each subcarrier. It gives the precise reference for symbol timing and frequency offset synchronization since the contents of this symbol are standardized and fixed[4][5]. The second is the fast information channel(FIC), which is used to transmit the control data that is necessary for demultiplexing and decoding of the MSC part in each transmission frame. The third is the main service channel(MSC), which is used to transmit each service(audio and data). Each transmission frame consists of consecutive OFDM symbols, each symbol consisting of a number of subcarriers. The number of OFDM symbols, the number of subcarriers and other various parameters are dependent on the transmission mode. These parameters for transmission mode Ⅰ and Ⅲ are shown in Table Ⅰ[1].

표 1. 전송모드 Ⅰ, Ⅲ에 대한 파라미터 .
Table 1. PARAMETERS FOR TRANSMISSION MODE Ⅰ and Ⅲ

Parameters		DAB transmission mode	
		Mode I	Mode III
No. of symbols/frame	L	76	153
No. of subcarriers/symbol	K	1536	192
Symbol duration	T_s	1.246ms	156 μs
Effective symbol duration	T_u	1ms	125 μs
Guard interval duration	Δ	246 μs	31 μs
Frame duration	T_F	96ms	24ms

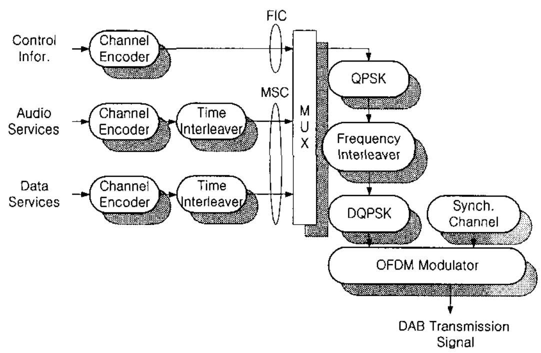


그림 2. 표준 DAB 시스템의 COFDM 블록도
Fig. 2. Block diagram of COFDM for standard DAB system.

A block diagram of the COFDM modulator for DAB system is shown in Fig. 2[1][4]. Control Information in the FIC and audio and data services in the MSC are encoded by means of RCPC codes with code rates available from 8/32, 8/31 · · · 8/9. RCPC encoding allows the application of equal and unequal error protection profiles(EEP, UEP), as wll. Then, time interleaving over several symbols (interleaving depth : 384ms) is adopted in the MSC to overcome the time selectivity of the mobile channel. No time interleaving is adopted for the FIC, because the control data is necessary for demultiplexing and decoding of the MSC part in each transmission frame. The coded and time interleaved bits are paired into dibits and mapped on the QPSK symbols. The QPSK symbols are re-ordered over the wideband multicarrier set to overcome the frequency selectivity of the mobile channels by means of frequency interleaving(interleaving depth : 1.536MHz). Then, $\pi/4$ shift DQPSK modulation is applied to the QPSK symbols on each subcarrier. Virtual carriers are then padded with zeros to make the number of subcarriers per symbol become a power of 2 and applied to an IFFT which performs the OFDM modulation. The guard interval is inserted at the transition between successive symbols to absorb the intersymbol interference(ISI) created by multipath in the channel. The OFDM transmitted signal can be expressed as.

$$s(t) = Re\Big\{ e^{j2\pi f_c t} \cdot \sum_{m=-\infty}^{\infty} \sum_{l=0}^{L} \sum_{k=-K/2}^{K/2} z_{mlk} \cdot g_{kl}(t - mT_F - T_{NULL} - (l-1)\,T_s) \Big\} \quad (1)$$

$$g_{kl}(t) = \begin{cases} 0 & if \quad l=0 \\ e^{j2\pi k(t-\Delta)/T_u} & if \quad l=1,2,\cdots L \end{cases} \quad (2)$$

where z_{mlk} is the complex DQPSK symbol associated to subcarrier k of OFDM symbol l during transmission frame m. After being sent through the mobile channel, the COFDM received signal is first synchronized and demodulated with a FFT. The data on each subcarrier are then differentially decoded and deinterleaved in frequency and in time. The output of the deinterleaver is quantized before being fed to the Viterbi decoder. Soft decision Viterbi decoding is performed to correct the random errors.

In addition to implementing the existing DAB system as a benchmark, we have improved the system's performance with the aid of a turbo codec. The block diagram of the turbo encoder is shown in Fig. 3. The turbo encoder employs two recursive systematic convolutional(RSC) codes connected in parallel, with a turbo interleaver preceding the second RSC encoder[8][13]. The two RSC codes are called the constituent codes of the turbo code. The information bits are encoded by both encoders. The first encoder operates on the input bits in their original order, while the second encoder operates on the input bits as permuted by the turbo interleaver. The information bits which are systematic bits from the first encoder are always transmitted across the channel. Depending on the code rate desired, the parity bits from the two constituent encoders are punctured before transmission.

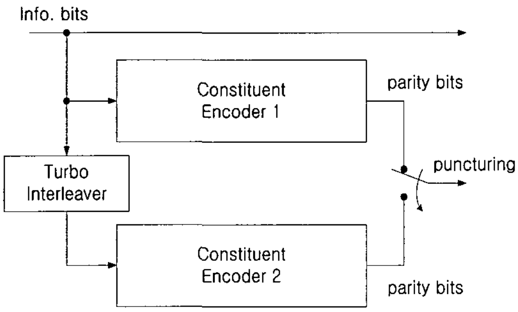


그림 3. 터보부호기 블록도
Fig. 3. Block diagram of turbo encoder.

Ⅲ. RCPC CODES FOR DAB

The channel coding process for DAB system is based on RCPC coding, which allows both equal and unequal error protection(EEP, UEP), matched to bit error sensitivity characteristics[1][4]. RCPC coding generates from a vector $(a_i)_{i=0}^{I-1}$ of I bits the resulting codeword $(b_i)_{i=0}^{M-1}$ of M bits. As a mother encoder, a rate 1/4 convolutional code with constraint length 7 and octal polynomial(133, 171, 145, 133) is used as shown in Fig. 4[1].

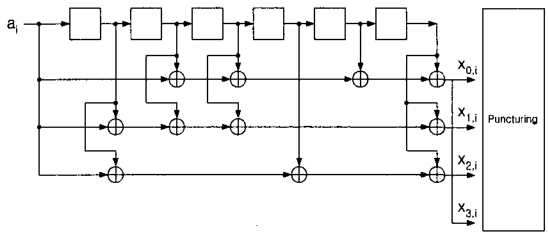


그림 4. 모체 콘볼루션 부호기
Fig. 4. Mother convolutional encoder.

The mother convolutional code generates from I information and six tail bits a codeword $(x_{0,i}, x_{1,i}, x_{2,i}, x_{3,i})_{i=0}^{I+5} = (u_i)_{i=0}^{4I+23}$. The codebits generated by mother code are not transmitted by the puncturing procedure. The first 4I bits $(u_i)_{i=0}^{4I-1}$ generated from I information bits are split into consecutive blocks of 128 bits. Each block is divided into four consecutive sub-blocks of 32 bits. All sub-blocks belonging to the same block are punctured by the puncturing vector V_{PI}, given by the value of the puncturing index(PI). Each index PI corresponds to a puncturing vector V_{PI} , denoted by

$$V_{T}=(\nu_{PI,0},\ \nu_{PI,1},\cdots\cdots,\ \nu_{PI,i},\cdots\cdots,\ \nu_{PI,31}) \quad (3)$$

where $V_{PI,i}=1$ connotes that the corresponding bit is transmitted and $V_{PI,i}=0$ indicates a deleted position. The values of the puncturing vectors are given in Table Ⅱ, where the value of the code rate 8/(8+PI) is also given. The puncturing procedure allows the effective code rate to vary between 8/9 and 1/4. The last 24 bits $(u_i)_{i=4I}^{4I+23}$ coded by six tail bits are also punctured using the puncturing vector given by

$$V_T = (1100\ 1100\ 1100\ 1100\ 1100\ 1100) \quad (4)$$

The resulting 12 bits are called punctured tail bits. Protection profile contains the puncturing indices and the length of the blocks that the puncturing indices are applied. Table III shows a protection profile applied in the FIC for transmission mode I and III[1]. For transmission I, the serial mother codeword generated from each I-bit vector is split into L consecutive blocks of 128 bits. The first L_0 blocks are punctured according to the puncturing index V_{PI0}. The remaining L_1 blocks are punctured according to the puncturing index V_{PI1}. This corresponds to a code rate of approximately 1/3. Finally, the last 24 bits of the serial mother codeword generated from the six tail bits are punctured. Therefore, the resulting punctured convolutional codeword of M bits is obtained. The same encoding procedure is applied to the four groups of I-bit vector. So, total M*4=9216 bits are obtained. After these bits are divided into three consecutive blocks, each block of 3072 bits is transmitted in an OFDM symbol. For transmission III, the same encoding procedure is applied except that the punctured convolutional codeword of M bits are divided into 8 consecutive blocks of 384 bits, which are transmitted in 8 OFDM symbols.

표 2. 펑처링 벡터

Table 2. PUNCTURING VECTORS.

PI	Code Rate	$(v_{PI,0}, v_{PI,1}, \cdots\cdots, v_{PI,i}, \cdots\cdots, v_{PI,31})$
1	8/9	1100 1000 1000 1000 1000 1000 1000 1000
2	8/10	1100 1000 1000 1000 1100 1000 1000 1000
3	8/11	1100 1000 1100 1000 1100 1000 1000 1000
4	8/12	1100 1000 1100 1000 1100 1000 1100 1000
5	8/13	1100 1100 1100 1000 1100 1000 1100 1000
6	8/14	1100 1100 1100 1000 1100 1100 1100 1000
7	8/15	1100 1100 1100 1100 1100 1100 1100 1000
8	8/16	1100 1100 1100 1100 1100 1100 1100 1100
9	8/17	1110 1100 1100 1100 1100 1100 1100 1100
10	8/18	1110 1100 1100 1100 1110 1100 1100 1100
11	8/19	1110 1100 1110 1100 1110 1100 1100 1100
12	8/20	1110 1100 1110 1100 1110 1100 1110 1100
13	8/21	1110 1110 1110 1100 1110 1100 1110 1100
14	8/22	1110 1110 1110 1100 1110 1110 1110 1100
15	8/23	1110 1110 1110 1110 1110 1110 1110 1100
16	8/24	1110 1110 1110 1110 1110 1110 1110 1110
17	8/25	1111 1110 1110 1110 1110 1110 1110 1110
18	8/26	1111 1110 1110 1110 1111 1110 1110 1110
19	8/27	1111 1110 1111 1110 1111 1110 1110 1110
20	8/28	1111 1110 1111 1110 1111 1110 1111 1110
21	8/29	1111 1111 1111 1110 1111 1110 1111 1110
22	8/30	1111 1111 1111 1110 1111 1111 1111 1110
23	8/31	1111 1111 1111 1111 1111 1111 1111 1110
24	8/32	1111 1111 1111 1111 1111 1111 1111 1111

표 3. FIC에 대한 보호 프로화일

Table 3. PROTECTION PROFILE APPLIED IN THE FIC.

	I	M	L	L_0	L_1	V_{PI0}	V_{PI1}
Mode I	768	2304	24	21	3	16	15
Mode III	1024	3072	32	29	3	16	15

The encoding procedure in the MSC depends on the type of service carried, the net bit rate and the desired level of protection. The input vector of the mother convolutional encoder consists of I-bit vector, where I is a function of audio bit rate. Table IV shows a protection profile for the audio bit rate 192 kbits/s and protection level 3[1]. The serial mother codewords are split into 144 blocks of 128 bits. Using the puncturing vectors shown in Table II, we obtain the punctured convolutional codeword 8960 bits, which are transmitted in 3 and 24 OFDM symbols for transmission mode I and III, respectively.

표 4. 전송모드 I, III에서 오디오 비트율 192kbit/s, 보호레벨 3에 대한 보호 프로화일

Table 4. PROTECTION PROFILE FOR THE AUDIO BIT RATE 192kbits/s AND PROTECTION LEVEL 3 FOR MODE I and III.

I	M	L	L_0	L_1	L_2	L_3	V_{PI0}	V_{PI1}	V_{PI2}	V_{PI3}
4608	8960	144	11	24	106	3	16	10	6	11

IV. TURBO CODED OFDM

From the puncturing vectors defined in Table II, we know that the first bit among the four coded bits is not punctured and always transmitted. Therefore, we can substitute the existing convolutional code with the turbo code because the information bits can be always transmitted as the systematic bits. For replacing a existing convolutional encoder by a turbo encoder, it is necessary not to modify the existing puncturing procedure and puncturing vectors for compatibility. First of all, to design a turbo encoder we must select a constraint length. The existing convolutional encoder shown in Fig. 4 needs six tail bits to flush the registers to zero state. Because the turbo encoder considered in this paper consists of the parallel concatenation of two constituent encoders, each constituent encoder with constraint length 4 is selected to separately flush the registers to zero state. Hence, a total of 6 tail bits which are equivalent to the standard convolutional encoder are needed to flush both of the constituent encoders. Next, we must choose a code rate for each constituent code. To obtain an overall code rate 1/4, each constituent code with code rate less than 1/2 is required. Therefore we choose a rate 1/3 code for each constituent code. It is known that maximizing the weight of output codewords corresponding to weight-2 data sequences, which weight dominates the performance characteristics, gives the best BER performance for a moderate SNR. A design for the best constituent codes for turbo codes by maximizing the effective free distance of the turbo code, in other words, the minimum output weight of codewords due to weight-2 input sequences, was reported in[14]. So, as a rate 1/3 constituent codes we use best rate 1/3 constituent codes with a maximum effective free distance[14]. The best rate 1/3 constituent code is given by the code generator as follows

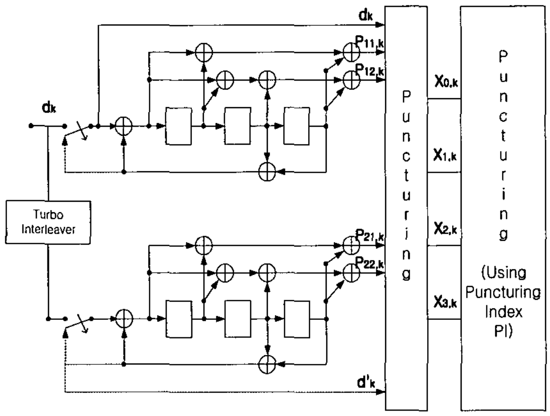


그림 5. 터보부호기의 구성

Fig. 5. Configuration of the turbo encoder.

$$G=\left[1,\ \frac{1+D+D^3}{1+D^2+D^3},\ \frac{1+D+D^2+D^3}{1+D^2+D^3}\right] \quad (5)$$

Fig. 5 shows the configuration of the designed turbo code encoder with a rate 1/3 constituent codes with constraint length of 4.

As a trellis termination, the switch allows to take input bits from register feedback[11][13]. The designed turbo code results in an overall code rate 1/5. Therefore, the appropriate puncturing of the parity bits is required to obtain an overall code rate 1/4. This additional puncturing must be considered according to the puncturing vectors. Table V shows additional puncturing tables for each PI value. The puncturing tables for this additional puncturing are designed to transmit all the systematic bits from the first encoder and the same amount of parity bits from both encoders when the puncturing vectors are considered together.

표 5. 펑처링 표

Table 5. PUNCTURING TABLES.

PI	$(d_k, P_{11,k}, P_{21,k}, P_{12,k}, P_{22,k})$
1	11011 10111 10111 10111 10111 10111 10111 10111
2	11011 10111 10111 10111 10111 10111 10111 10111
3	11011 10111 10111 10111 11011 10111 10111 10111
4	11011 10111 10111 10111 11011 10111 10111 10111
5	11011 10111 11011 10111 10111 10111 11011 10111
6	11011 10111 11011 10111 10111 11011 10111 10111
7	11011 10111 11011 10111 11011 10111 11011 10111
8	11011 10111 11011 10111 11011 10111 11011 10111
9	11110 11011 10111 11011 10111 11011 10111 11011
10	11110 11011 10111 11011 11110 10111 11011 10111
11	11110 11011 11110 10111 11110 11011 10111 11011
12	11110 11011 11110 10111 11110 11011 11110 10111
13	11110 11101 11110 11011 11110 10111 11110 11011
14	11110 11101 11110 11011 11110 11101 11110 10111
15	11110 11101 11110 11101 11110 11101 11110 11011
16	11110 11101 11110 11101 11110 11101 11110 11101
17	11110 11110 11101 11110 11101 11110 11101 11110
18	11110 11110 11101 11110 11101 11101 11110 11101
19	11110 11110 11101 11101 11110 11110 11101 11110
20	11110 11110 11101 11101 11110 11110 11101 11101
21	11110 11101 11110 11110 11101 11101 11110 11110
22	11110 11101 11110 11110 11101 11110 11101 11101
23	11110 11101 11110 11101 11110 11101 11110 11110
24	11110 11101 11110 11101 11110 11101 11110 11101

Fig. 6 gives a block diagram for a turbo code decoder. Punctured bits marked with "0" in the puncturing tables and puncturing vectors are depunctured with zeros. Soft-decision(likelihood) information for the systematic and parity bits from the first constituent code is sent to the first decoder. The decoder generates updated soft-decision likelihood values for the information bits that are passed to the second decoder as a priori information after reodering in accordance with the turbo interleaver. In addition,

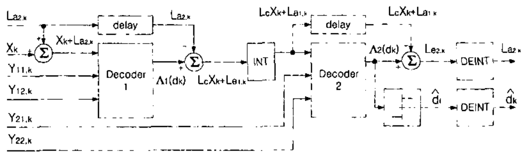


그림 6. 터보복호기
Fig. 6. Turbo code decoder.

the second decoder accepts the updated likelihood information for the systematic bits and the soft-decision information from the channel for the parity bits from the second constituent encoder. The soft-decision output of the second decoder regarding updated likelihood information for the systematic bits is then fed back to the first decoder to repeat the process. The process can be iterated as many times as desired. As a decoding algorithm, the MAP algorithm is applied[8][15][16]. The logarithm of likelihood ration(LLR) $\Lambda(d_k)$ associated with each decoded bit d_k by the decoder is given by

$$\Lambda(d_k) = \log \frac{\sum_m \sum_m \gamma_1(R_k, m', m)\alpha_{k-1}(m')\beta_k(m)}{\sum_m \sum_m \gamma_0(R_k, m', m)\alpha_{k-1}(m')\beta_k(m)} \quad (6)$$

where R_k is the input to a decoder at time k.

$$R_k = (X_k,\ Y_{11,k},\ Y_{12,k},\ L_{a2,k}) \quad (7)$$

The quantity $\alpha_k(m)$ is determined in the forward recursion and the backward recursion yields $\beta_k(m)$[8]. The branch transition probabilities are

$$\gamma_i(R_k, m', m) = \exp\left\{ \frac{2}{\sigma^2}[(X_k + L_{a2,k})d_k + Y_{11,k}P_{11,k} + Y_{12,k}P_{12,k}] \right\} \quad (8)$$

where $P_{11,k}$ and $P_{12,k}$ correspond to the transition between the trellis state $S_{n-1} = m'$ and $S_n = m$. σ^2 is the variance of the white Gaussian noise. Using the systematic data X_k present at the input to the turbo-code decoder, the variance is estimated as

$$\hat{\sigma} = \frac{\sum_{k=0}^{N-1} X_k^2}{N} - \left[\frac{\sum_{k=0}^{N-1} |X_k|}{N} \right]^2 \quad (9)$$

V. CHANNEL MODEL

Multipath fading is one of the major causes of communication impairments in radio links[18][19]. A radio link is said to be affected by multipath fading when the received signal can be modeled as a combination of several rays, with each ray arriving at the receiver via a different path and therefore having a different attenuation and delay. In general, a multipath fading channel may be represented in the base-band as a direct(line-of-sight : LOS) component and the sum of multipath components which vary in number, amplitude, phase and delay. A generalized equation to represent the received signal is given by

$$y(t) = \frac{\rho_0 \chi(t) + \sum_{i=1}^{N} \rho_i e^{-j2\pi\theta_i} \chi(t-\tau_i)}{\sqrt{\sum_{i=0}^{N} \rho_i^2}} \quad (10)$$

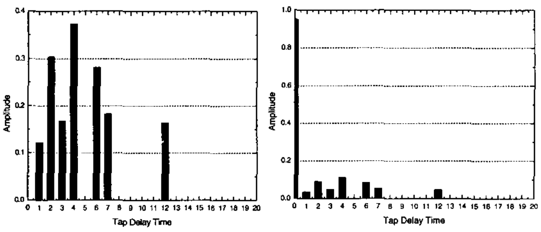
where ρ_0 is the direct component, N is the number of multipath components, ρ_i is the amplitude of the ith multipath component, θ_i is the phase shift of the multipath component and τ_i represents the propagation delay of the ith multipath component. In the absence of a direct component the channel is modeled as Rayleigh fading channel and the received signal is given by

$$y(t) = \frac{\sum_{i=1}^{N} \rho_i e^{-j2\pi\theta_i} \chi(t-\tau_i)}{\sqrt{\sum_{i=0}^{N} \rho_i^2}} \quad (11)$$

When a fading channel has LOS component together with multipath component, the channel is modeled as Rician fading channel, which is categorized depending on the Rician factor, K. This is defined as the ratio between the direct path and the multipath powers such as

$$K = \frac{\rho_0^2}{\sum_{i=1}^{N} \rho_i^2} \tag{12}$$

As a channel model employed in this study we use the Rician fading channel with K value of 10dB and the Rayleigh fading channel in conjunction with terrestrial(DAB transmission mode Ⅰ) and satellite(DAB transmission mode Ⅲ) broadcasting. The numerical values of the statistical parameter of the amplitudes, phases and delays are based on the parameters specified in European Telecommunication Standard Institute(ETSI)[20]. The delay profiles for each fading channel are shown in Fig. 7, which represent 12 paths frequency selective fading channel model.



(a) Rayleigh channel (b) Rician channel

그림 7. 다중경로 지연 프로파일
Fig. 7. Multipath delay profiles.

Ⅵ. SIMULATION RESULTS AND DISCUSSION

In this section, the performance of a new turbo coded OFDM(TCOFDM) system is evaluated by means of computer simulation under the Rician fading channel with K-factor value of 10 dB and Rayleigh fading channel in DAB transmission mode Ⅰ and Ⅲ, which are suitable to the terrestrial and satellite broadcasting. The channel model of Fig. 7 is employed. The average bit error rate(BER) versus E_b/N_0 performance between a new TCOFDM system and a standard COFDM system is compared in the FIC and MSC for DAB transmission mode Ⅰ and Ⅲ. Table Ⅵ lists the main system environments used for simulation. The time interleaving depth is 384msec and the frequency interleaving depth is 1.536MHz. For the existed COFDM, 16 levels soft decision is used. For the TCOFDM, random interleaver is used.

표 6. 시뮬레이션 환경
Table 6. SIMULATION ENVIRONMENTS.

	Standard COFDM system	New TCOFDM system
Encoder	Fig. 4	Fig. 5
Decoding algorithm	16 levels soft decision Viterbi	MAP
Turbo interleaver		Random
Time interleaver	384msec	
Frequency interleaver	1.536MHz	
Simulation parameters	TABLE I	
Protection profile of FIC and MSC	TABLE III and IV	

1. BER Performance in DAB Transmission Mode I

Fig. 8 and Fig. 9 show the average BER curves after Viterbi or MAP decoding in the FIC, which is a non-time interleaved with fixed equal error protection. The results, which are plotted in Fig. 8, show that at a BER= 10^{-4} the turbo coded system achieves a gain of about 1dB after 2 iterations on the Rician fading channel. The results in Fig. 9 correspond to the Rayleigh fading channel. At a BER= 10^{-4}, the TCOFDM scheme offers an improvement of about 1.2dB after 2 iterations over the COFDM scheme. With results in Fig. 8 and 9, a considerable performance improvement can be obtained only after 2 iterations. But the amount of performance improvement is decreased with the increased iterations. This is because the more iterations are increased, the more correlation of encoded bits is increased.

Fig. 10 and Fig. 11 show the BER performance in the MSC, which is a time interleaved with unequal error protection. At a BER= 10^{-4}, the simulation results offer an improvement of 1.3 dB and 2.5dB after 2 iterations over the COFDM on the Rician fading channel and the Rayleigh fading channel, respectively. In the experiment, you can see the convergence at four or five iterations. Also, because

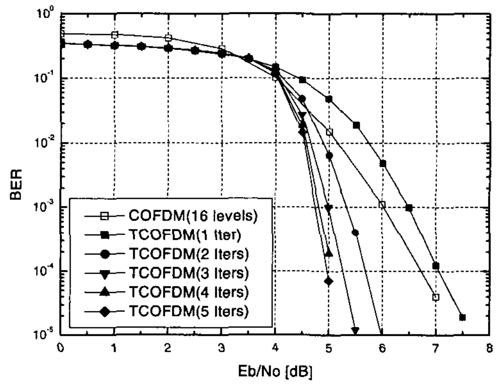


그림 8. 전송모드 I FIC(라이시안 채널)
Fig. 8. FIC(Rician channel).

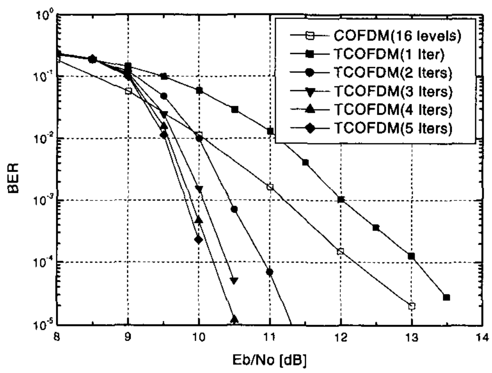


그림 9. 전송모드 I FIC(레일레이 채널)
Fig. 9. FIC(Rayleigh channel).

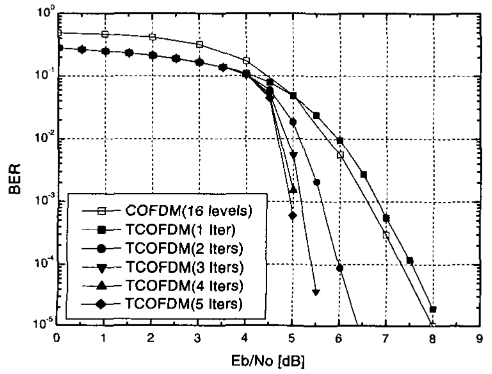


그림 10. 전송모드 I MSC(라이시안 채널)
Fig. 10. MSC(Rician channel).

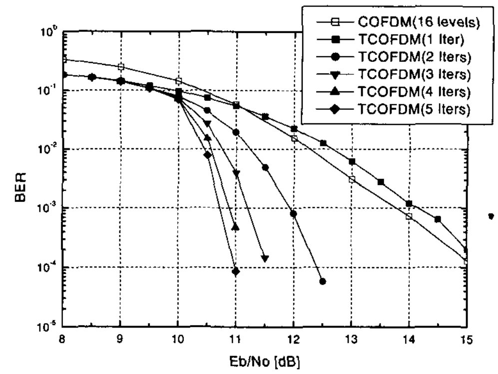


그림 11. 전송모드 I MSC(레일레이 채널)
Fig. 11. MSC(Rayleigh channel).

the length of frame which inputs to interleaver in the MSC channel is longer than that of FIC channel, you can see the better performance improvement in the MSC channel than FIC channel.

2. BER Performance in DAB Transmission Mode III

Fig. 12 and Fig. 13 show the BER performance in the FIC. As can be seen from the plot in Fig. 12, at the BER threshold of 10^{-4} the turbo coded system is superior to the standard system by about 1 dB after 2 iterations on the Rician fading channel. The results in Fig. 13 correspond to the Rayleigh fading channel. At a BER= 10^{-4}, the TCOFDM scheme achieves a gain of about 1.5 dB after 2 iterations over the COFDM scheme.

Fig. 14 and Fig. 15 show the BER performance in the MSC. At a BER= 10^{-4}, the BER performance after MAP decoding offers an improvement of 1.6 dB

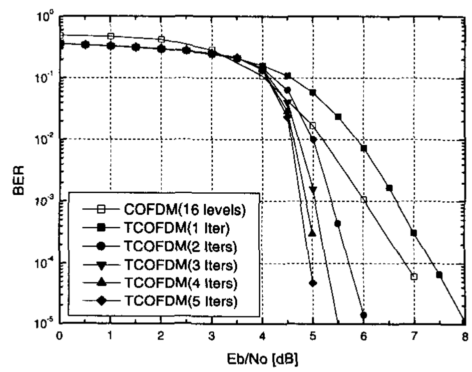


그림 12. 전송모드 III FIC(라이시안 채널)
Fig. 12. FIC(Rician channel).

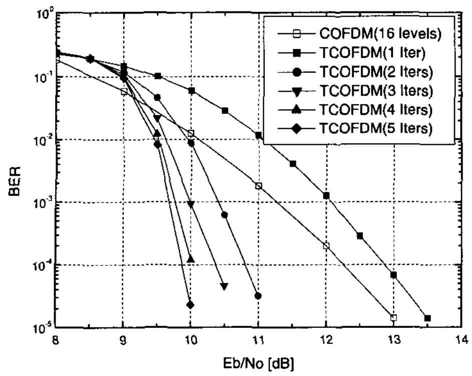


그림 13. 전송모드 Ⅲ FIC(레일레이 채널)
Fig. 13. FIC(Rayleigh channel).

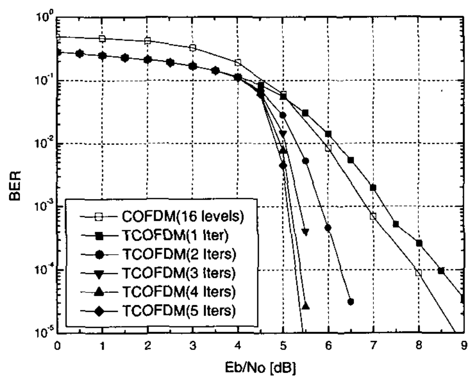


그림 14. 전송모드 Ⅲ MSC(라이시안 채널)
Fig. 14. MSC(Rician channel).

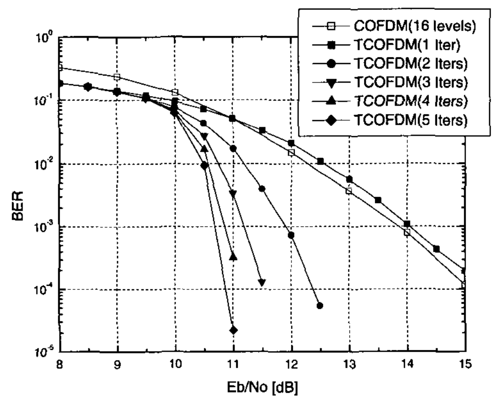


그림 15. 전송모드 Ⅲ MSC(레일레이 채널)
Fig. 15. MSC(Rayleigh channel).

and 2.6 dB after 2 iterations over the BER performance after 16 levels soft decision Viterbi decoding on the Rician fading channel and the Rayleigh fading channel, respectively. But the trade-off of these performance improvements is the increased complexity and time delay in decoding.

Ⅶ. CONCLUSIONS

In this paper, we have investigated the performance of a turbo coded DAB system in transmission mode Ⅰ and Ⅲ. The convolutional codec specified in the standard system was substituted with the turbo codec without modifying the puncturing procedure and puncturing vectors defined in the Eureka 147 DAB system. A range of system performance results were presented based on the standard DAB system as well as on a turbo coded system. Simulation results indicate that the turbo coded system results in a substantial coding gain, in other words, the transmitted power requirements of the standard system employing convolutional codec can be reduced upon invoking more complexity but more powerful turbo codec.

To get a desired error performance, the proposed system can decrease the power of transmission or improve the error performance at the same power condition as the existing system. On the other hand, by using turbo code, we can predict processing delay, an increased complexity and shortening of the power length of life. But, when we consider that the turbo code is recommended for the method of the standard channel code of IMT-2000, we anticipate that it will not make any problems.

REFERENCES

[1] ETSI, Digital audio broadcasting(DAB) to mobile, portable and fixed receivers, ETS 300 401, May 1997.

[2] L. Thibault and M. T. Le, "Performance Evaluation of COFDM for Digital Audio Broadcasting : Part I," IEEE trans. On Broadcasting, vol. 43, pp. 64~75, March1997.

[3] K. Taura and H. Kato, "A Digital Audio Broadcasting(DAB) Receiver," IEEE trans. On

Consumer Elec., vol. 42, pp.322~326, Aug. 1996.

[4] U. Liebenow and G. Zimmermann, "Investigations of Single Frequency Networks for Digital Mobile Radio Systems Based on COFDM," in VTC'98, pp. 2227~2231, 1998.

[5] J. A. Huisken, F. A. M. Laar, M. J. G. Bekooij, G. C. M. Gielis, P. W. F. Gruijters and F. P. J. Welten, "A Power-Efficient Single-Chip OFDM Demodulator and Channel Decoder for Multimedia Broadcasting," IEEE J. of Solid-State Circuit, vol. 33, pp.1793~1798, Nov. 1998.

[6] M. Alard and R. Lassalle, "Principles of Modulation and Channel Coding for Digital Broadcasting for Mobile Receivers," EBU Tech. Rev. No. 224, pp. 168~190, August 1987.

[7] W. Y. Zou and Y. Wu, "COFDM : An Overview," IEEE Trans. Broadcasting vol. 41, No. 1, pp. 1~7, March 1995.

[8] C. Berrou and A. Glavieux, "Near Optimum Error Correcting Coding and Decoding : Turbo-Codes," IEEE trans. On Comm., vol. 44, pp. 1261~1271, Oct. 1996.

[9] C. S. Lee, S. Vlahoyiannatos and L. Hanzo, "Satellite Based Turbo-Coded, Blind-Equalized 4-QAM and 16-QAM Digital Video Broadcasting," IEEE trans. On Broadcasting, vol. 46, pp. 23~33, March 2000.

[10] V. Kuhn, "Evaluating the Performance of Turbo Codes and Turbo-Coded Modulation in a DS-CDMA Environment," IEEE JSAC, vol. 17, pp. 2138~2147, Dec. 1999.

[11] D. Divsalar and F. Pollara, "Turbo Codes for PCS Applications," in ICC'95, Seattle, Washington, June, 1995.

[12] H. J. Kim, "Turbo Coded Orthogonal Frequency Division Multiplexing for Digital Audio Broadcasting," in ICC'00, New Orleans, pp. 420~424, June 2000.

[13] M. C. Reed and S. S. Pietrobon, "Turbo-Code Termination Schemes and a novel alternative for Short Frames," in PIMRC'96, vol. 2, pp. 354~358, Oct. 1996.

[14] D. Divsalar and F. Pollara, "On the Design of Turbo Codes," TDA Progress Report, Nov. 1995.

[15] L. R. Bahl, J. Cocke, F. Jelinek and J. Raviv, "Optimal Decoding of Linear Codes for Minimizing Symbol Error Rate," IEEE Trans. Inform. Theory, vol. IT-20, pp. 248~287, Mar. 1974.

[16] P. Jung, "Comparison of Turbo-Code Decoders Applied to Short Frame Transmission Systems," IEEE JSAC, vol. 14, pp. 530~537, April 1996.

[17] M. Reed and J. Asenstorfer, "A Novel Variance Estimator for Turbo-Code Decoding," in ICT '97, pp. 173~178, April 1997.

[18] H. Goldman, "Mathematical Analysis of the Three-Ray Dispersive Fading Channel Model," IEE Proceedings, vol. 138, pp.87~94, April 1991.

[19] C. Loo and N. Secord, "Computer Models for Fading Channels with Application to Digital transmission," IEEE Trans. on Vehi. Tech., vol. 40, pp.700~707, Nov. 1991.

[20] ETSI, Digital broadcasting systems for television, sound and data services , DRAFT pr ETS 300 744, March 1996.

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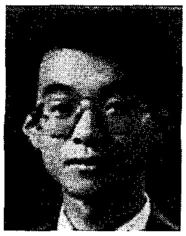

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