EFM—The Modulation Method for the Compact Disc Digital Audio System

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The modulation method employed in the Compact Disc (CD) digital audio system codeveloped by Philips N.V. (Eindhoven, The Netherlands) and Sony Corporation (Tokyo, Japan) is described. This method, called eight-to-fourteen (EFM), is an 8 (data bit) → 14 (channel bit) conversion block code with a space of 3 channel bits for every converted 14 channel bits which is used to connect the blocks. These 3 channel bits, called merging bits, are selectable, enabling the suppression of the low-frequency contents of the frequency spectrum.

First some of the major conditions are listed which are required of the modulation method used for recording/reproducing digital audio signals on an optical disk. The various parameters of EFM as a modulation method are explained in the second part, proving the suitability of EFM for optical disks. An actual example explains the method in detail; a frequency spectrum is also given to enhance understanding.

EFM is well matched with the error-correction method CIRC employed in the CD. The combination of these two methods plays an important role in stably reproducing a 2-channel 16-bit audio signal on a 12-cm-diameter optical disk for more than playing time, single sided.

0 INTRODUCTION

Sony Corporation (Tokyo, Japan) and Philips N.V. (Eindhoven, The Netherlands) have completed the development of an optical digital disk system, the Compact Disc digital audio (CDDA or CD) system, in 1980 June [1]–[3]. The CDDA disk contains a spiral-shaped track of successive shallow depressions, also called pits, in a reflective layer. The encoded digital audio information is stored in the length of the pits and the gaps between them. The disk rotates at a constant linear velocity (CLV) of 1.2–1.4 m/s.

A laser beam emitted by a solid-state GaAlAs laser (wavelength λ ~ 0.78 μm), focused on the disk, is reflected by the information layer. The reflected light, modulated by the information on the disk, is detected by a photodiode and will be processed further electronically. The readout is contactless. Electromechanical servo systems focus the laser spot on the disk and follow the track to within the specified accuracy.

This CD system permits recording and reproduction of a 16-bit 2-channel stereo signal for more than 60 min on a 12-cm-diameter disk single sided. This remarkable characteristic is obtained by the high-density recording eight-to-fourteen modulation (EFM) method and the highly efficient error-correction method CIRC (cross interleave Reed–Solomon code) [4], both of which have been developed in close cooperation between...
Sony and Philips.

Some of the requirements for the modulation method of the optical disk system are given in Section 1. Section 2 deals with the comparison of experimental results of modulation methods we have considered for the CD system. Section 3 gives details of the adopted modulation method, EFM. In Section 4, some of the frequency spectrum characteristics of EFM are presented, and a brief summary is given in Section 5.

1 REQUIREMENTS OF THE MODULATION METHOD FOR THE CD SYSTEM

The binary signal from the CIRC error-control encoder cannot be recorded directly. We need a modulation system as an intermediary, which converts the incoming data into another code satisfying the following requirements for the CD system.

1.1 Correct Readout at High Information Densities

The modulation transfer function or frequency characteristic of the CD readout system can be modeled as a low-pass filter with linear phase [4]. The cut-off frequency (ideal optics) is simply given by \(2\text{NA}/\lambda \cdot V\) where \(\lambda\) is the wavelength, NA the numerical aperture of the objective lens, and \(V\) the linear scanning velocity. Fig. 1 shows a typical frequency characteristic \((V = 1.25 \text{ m/s, } \text{NA} = 0.45, \lambda = 0.78 \text{ m}).\)

Nonideal optics, for example, due to defocusing or disk skewing with respect to the optical axis, will degrade the phase and amplitude characteristics. A modulation system should satisfy the requirement of low sensitivity to tolerances of the optical light path even at high information densities.

1.2 Clock Content

The bit clock is one of the most important signals in the CD system. It is used for the synchronization of the digital data and the motor control. The bit clock must be regenerated from the readout signal itself by detecting the pit edges. The signal must therefore have a sufficient number of transitions, and \(T_{\text{max}}\), the maximum distance between transitions pit–land or land–pit, must be as small as possible.

1.3 Low-Frequency Contents

As the signal of the CD is covered by a 1.2-mm-thick transparent plastic layer, it is protected against dirt and scratches. The dirt and scratches on the surface of the disk change the envelope of the readout signal, giving rise to low-frequency noise. These low-frequency disturbances can be filtered out provided that the signal contains no low-frequency components of its own. Another reason for imposing the low-frequency constraint is the use of servo systems for radial tracking and focusing. Low-frequency components cause interference in the servo systems, making them unstable.

1.4 Error Propagation

Error propagation of the modulation system must be as small as possible and should match the CIRC error-correction system that treats 8 data bits as a unit [5].

2 COMPARISON BETWEEN MODULATION METHODS

As the requirements given in Section 1 are conflicting, we have made some experiments with several well-known modulation methods [6], [7] with the purpose of developing the most effective modulation method for the CD system. Table 1 gives the main parameters of the modulation systems we considered. \(T_{\text{min}}\) and \(T_{\text{max}}\) are the minimum and maximum times between transitions per data bit time, and the (timing) window is the sampling distance per data bit time.

As an extreme example, methods A and B were compared first. Fig. 2 shows that the number of errors are counted by taking first a pseudo-random generator signal of a 12-bit shift register as an input, carrying out the modulation, recording/reproducing it on/from the disk under identical conditions, and then comparing the signal with the original one. The experiment, of which the results are given below, is carried out mainly under normal conditions. Measurements made with an objective lens of NA = 0.5 and wavelength \(\lambda = 0.78 \text{ m}\) showed the following.

Table 1. Modulation system parameters.

<table>
<thead>
<tr>
<th>Method</th>
<th>(T_{\text{min}})</th>
<th>(T_{\text{max}})</th>
<th>Window</th>
<th>Low-Frequency Components</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>1</td>
<td>2.5 (3)</td>
<td>0.5</td>
<td>Good</td>
</tr>
<tr>
<td>B</td>
<td>2</td>
<td>8.3</td>
<td>0.33</td>
<td>Bad</td>
</tr>
<tr>
<td>C</td>
<td>1.5</td>
<td>4.0 (4.5)</td>
<td>0.5</td>
<td>Bad</td>
</tr>
</tbody>
</table>

Fig. 1. Frequency response of the CD optical readout system with ideal optics. \(V = 1.25 \text{ m/s, } \text{NA} = 0.45, \lambda = 0.78 \text{ m};\) cutoff frequency 1.44 MHz. The phase characteristic of the readout system is linear for ideal optics. Defocusing and tilting of the disk, however, have a disastrous effect on both amplitude and phase response. For example, defocusing of 2 \(\text{m}\) almost halves the bandwidth of the readout system.
1) The average bit error rate at a window margin of 0.5 is about $1.5 \times 10^{-5}$, while that at a window margin of 0.3 is approximately three times this value. It follows that a window margin of 0.5 may be sufficient for the CD system, provided that an error correction is applied, the effective combination with the error-correction of 0.3 is approximately three times this value. It follows C by giving up a small percentage of the overall in-

Next the properties of methods A and C were tested for the following points.
1) The minimum NA value that permits a stable readout at the same information density (0.6 µm/bit, 0.5 µm/bit, etc.)
2) Tolerance against defocusing
3) Tolerance against mistracking
4) Tolerance against disk skewing
5) Tolerance against noise
6) Tolerance against defects, such as scratches and fingerprints.

In the optical system the lower the NA of the lens, the more stable the system, and judging from this point, it has become clear that the optical tolerance of method C is larger than that of method A for the same density. Experiments showed that method C enables a 25–40% higher density recording/playback than method A.

Both methods A and C show similar tolerance against mistracking and noise. However, method C naturally has a larger tolerance against defocusing and disk skewing. As for defects, the tolerances of the two methods show little difference. Unfortunately code C has a much larger power at low frequencies than code A, giving rise to possible servo problems.

We improved the low-frequency properties of code C by giving up a small percentage of the overall information density. In a later stage we took into account the effective combination with the error-correction system. This is an 8-bit base block code satisfying the various requirements, called EFM.

3 DESCRIPTION OF EFM

EFM is a modulation method based on a 8-to-14-bit conversion. With EFM an 8-data-bit unit from CIRC, called symbol, is mapped onto 14 channel bits. Accordingly the error propagation of EFM is also limited to 8 data bits, thus satisfying the requirement of Section 1.4.

From the requirement of Section 1.1 the code is generated in such a way that the minimum distance \( t_{\text{min}} \) between two transitions is 3 channel bits (= 1.5 data bits), while the sampling window is 1 channel bit (= 0.5 data bit).

From the requirement of Section 1.2, 267 different patterns of 14-bit length satisfy the constraints: \( t_{\text{min}} = 3 \) channel bits and \( T_{\text{max}} = 11 \) channel bits [8]. Since only 256 patterns are required, 11 patterns are skipped. A part of the EFM conversion table is given in Fig. 3. Here a 1 implies a transition. The information to EFM refers to the presence or absence of a transition. Whether it is an upward or a downward transition is irrelevant.

It is also necessary to add at least 2 channel bits for connecting these patterns without violating the \( t_{\text{min}} \) constraint. In order to increase its flexibility, however, EFM has 3 channel bits for merging, the bits being selected within the constraints of \( t_{\text{min}} \) and \( T_{\text{max}} \) [9].

Fig. 3. Part of the EFM conversion table. The left column shows the decimal representation of the 8-bit data word in the middle column. The data are translated to the 14-bit EFM output words shown in the right column. The 1s in the output words indicate a transition pit–land or land–pit on the CD "groove." Note that the table is so chosen that at least two 0s are found between 1s. In this way the minimum distance between transitions (pit–land or land–pit) is 3 channel bits, equivalent to $3 \cdot (2^{3} + 14) = 1.41$ data bits. The conversion table was compiled with the aid of computer optimization in such a way that the translation in the player can be carried out with the simplest possible circuitry, that is, a circuit that contains the minimum of logic gates.

**Fig. 2.** Block diagram of the circuitry used to measure the bit error rate of experimental CDs. A known pseudo-random sequence is recorded on the disk. During readout the data from the disk are compared with the known sequence. If the data from the disk do not equal the known data, an error flag is generated. The bit error signal can be used to acquire knowledge of the error statistics of the disk. In the experimental stage of the CD this method was used to compare modulation systems.
In the CD system a frame consists of one frame-synchronization pattern (24 channel bits), one subcoding data symbol (14 channel bits), and 32 music data and parity symbols (14 channel bits each). The frame format of the CD system is shown in Fig. 4.

As shown in Fig. 5, these 98 frames form a subcoding frame. Here \( S_0 \) and \( S_1 \) are used for synchronizing the subcoding frames. They are selected from the above-mentioned 11 patterns skipped from the 267 combinations. These synchronization patterns are

\[
\text{main frame sync } \quad 10000000000100000000000010 \\
S_0 \quad 001000000000000001 \\
S_1 \quad 000000000100010010 
\]

In Fig. 6 a circle indicates a possible merging bit (3 channel bits for connection); two or more circles indicate an option of selection.

In order to satisfy the requirement in Section 1.3, the low-frequency content is reduced by a proper choice of the merging bit. The low-frequency content is suppressed by balancing the digital sum value (DSV). The digital sum value is defined as the sum of the 1s and 0s (where a 0 is counted as -1).

A simple example is given to show one of the possible algorithms of the EFM. If a symbol of 01110111 follows a converted symbol that ends with ...010, then this symbol is converted to 14 bits corresponding to symbol 119 by means of the conversion table. As a result, a 3-channel-bit space is provided between the symbol and its preceding symbol, as shown in Fig. 7.

The digital sum value up to time \( t_0 \) is assumed to be -3 with a polarity of 1. As for the merging bits, there are three possibilities:

\[
\begin{align*}
000 \\
010 \\
001
\end{align*}
\]

Fig. 5. Subcoding format. With subcoding (or control and display) additional information can be stored on the CD. It has a capacity of approximately 70 kbit/s. The additional information can be used for SMPTE time code, catalog numbers, and general information.

Fig. 4. Frame format. A frame consists of 588 channel bits. One may observe a unique sync pattern of 24 channel bits, followed by the control and display symbol. The rest of the frame contains audio information and the parity checks of the error-correction system.
In the above case, 100 must be skipped in virtue of the $T_{\text{min}}$ constraint. ($T_{\text{min}}$ must be at least 3 channel bits long.)

Next the digital sum value at time $t_1$ is evaluated for each possible case, as illustrated in Fig. 8.

Because pattern 000 produces the minimum absolute digital sum value at $t_1$ of the three patterns, pattern 000 is selected. When the absolute values of the digital-sum-value merging alternatives are equal, the one including a transition is selected preferentially.

However, some exceptions must be given to this algorithm in order to avoid generating a synchronization circuitry for the selection of the optimum merging bits. Demodulation of the 14 → 8 conversion is achieved in ROM or array logic, disregarding the merging bits.

The merging criterion is the dc content expressed as the digital sum value at time $t_1$ of the three patterns, pattern 000 bit density = 0.58 in a ROM for the 8 → 14 conversion and by logic circuitry. The logics for the selection of the optimum merging bits are rather complex. However, with the above-mentioned symbols, a small area of the signal-processing large-scale integrated circuit.

Demodulation using a ROM requires large-scale terms, whereas that using array logic requires only about 50 product terms, and the circuit will use only a small area of the signal-processing large-scale integrated circuit. The logics for the selection of the optimum merging bit are rather complex. However, with the CD system (a play-only device) this presents no problems, because the merging bits are not used in the demodulation of the EFM.

Although the algorithm itself introduced in this section is simple, it is very effective, as can be seen from the spectrum later on, and it is considered to be sufficient for a CD system. Noise in the servo loop other than containing 24 channel bits for processing.

The final EFM performance is

$$T_{\text{min}} = 1.41T$$

$$T_{\text{max}} = 5.18T$$

Window margin = 0.471T

bit density = 0.58 \mu m/bit (at linear velocity of 1.25 m/s).

The modulation may be realized with a look-up table in a ROM for the 8 → 14 conversion and by logic circuitry for the selection of the optimum merging bits. Demodulation of the 14 → 8 conversion is achieved in ROM or array logic, disregarding the merging bits.

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the low-frequency component of the signal is disk noise (from dirt and scratches on the disk surface). The balance of the two noise sources must also be considered. It is possible to extend the algorithm in the same format, which will be presented in the near future. (See [10] for one example.)

4 SOME FREQUENCY SPECTRUM CHARACTERISTICS OF EFM

Some simulation based on the final CD format has been carried out. The simulated music data and subcoding data are generated by a random sequence generator. Figs. 10 and 11 are the results of the frequency spectrum with and without low-frequency suppression, respectively. (However, even in Fig. 10 the \( T_{\text{min}} \) and \( T_{\text{max}} \) constraints are satisfied.) Here a substantial improvement in the low-frequency range is obtained.

Fig. 12 shows the frequency spectrum of the CD signal in a wide frequency range. The frequency spectrum for an actual music signal is similar to that shown in Figs. 11 and 12.

When the digital audio data become a fixed pattern and the peak of the spectrum lies within a frequency of which the lowest common multiplier is either the frame frequency (7.35 kHz) or half the frame frequency (3.675 kHz), the conversion table was selected in such a way that the peak does not rise above the envelope shown in Fig. 11.

5 SUMMARY

A brief explanation of the modulation method EFM employed in the CD system has been given. EFM is a block code, basically an 8-to-14 conversion code, where the low-frequency content is suppressed using selectable merging bits. EFM has the following characteristics.

1) The minimum time between transitions is 3 channel bits. Thus high-density recording and playback in an optical system can be achieved without being affected by intersymbol interference.

2) The run length lies between 3 and 11 channel bits, resulting in a stable clock recovery of the CD system.

3) The low-frequency content of EFM is small, enabling a stable servo system.

4) Error propagation of the EFM is limited to 8 bits, which is suitable for a CIRC error-correction system.

6 ACKNOWLEDGMENT

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Fig. 11. Power density function of EFM in the low-frequency region with dc suppression. Notice the improvement in the frequency range used by the servo systems (<20 kHz).

Fig. 12. Overall power density function of EFM signal. The channel bit frequency is 4.3218 Mbit/s.
Further they want to mention Mr. M. Mizushima and Mr. J. P. Sinjou for being cooperative intermediaries between Japan and The Netherlands.

The authors would also like to note with regret that, due to space limitation, only part of the acquired data have been presented in this paper.

7 REFERENCES


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In 1979 he joined the optical recording group working on channel codes for optical disk systems, in particular the digital audio Compact Disc system. As a result of his work he holds patents in the fields of servo, acoustical, and modulation systems.