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Presented at the 70th Convention 1981 October 30-November 2 New York

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AN AUDIO ENGINEERING SOCIETY PREPRINT

Channel Codings for Digital Audio Recordings

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Presented at 70th AES Convention
1981, Oct. 30 - Nov. 2 at N.Y., USA

"ABSTRACT"

Channel coding is very important in order to obtain high density recording and high reliability at the same time. Several new channel codes are developed for the improvement of digital audio recordings. In this paper, the fundamental parameters for the evaluation of channel codes are described, and several conventional codes are explained with the application to the present digital audio systems. In comparison to those, new codes are discussed. One of the new codes is applied to the format of professional digital audio recorders. The other is applied to Compact Disc Digital Audio Systems.

1. INTRODUCTION

The packing density of computer peripherals has improved greatly within several years, by virtue of the progress of (1) the channel codings (modulation schemes) and (2) the error correction schemes.

It is also observed that the packing density of digital audio recordings has made a remarkable progress and already achieved much higher density than computer peripherals. A lot of new Coding schemes both for error correction and for modulation have been proposed, and have contributed to this achievement.

The purpose of this paper is to describe these new channel codings, but in order to make the difference clear, it is necessary to explain about history of channel codings, the parameters for evaluation, conventional codings and their application to digital audio systems. Therefore, the first half of this paper is rather tutorial.

In the field of computer peripherals as well as early models of digital audio recorders, the channel coding of MFM (Modified Frequency Modulation = Delay Modulation = Miller Code) has been considered as a standard. But it was found that MFM is not optimum, and a lot of improvements are reported with a slight increase of the complexity of modulator/demodulator circuits. They are classified into three directions;

(1) DC-free,

(2) wider window margin, and

(3) higher density ratio.

In this paper, a series of new channel codings is proposed in the direction of (3) with some combination to the direction (1), which is named as HDM (High Density Modulation). One of them is adopted to the proposed format of stationary-head recorders for professional use [1].

As to the Compact Disc Digital Audio, the requirement of the combination of (1) and (3) is essential, and another new coding named EFM (Eight to Fourteen Modulation) [2] is developed with an additional feature of avoiding the possibility of error propagation.

2. RECENT PROGRESS IN PACKING DENSITY

Fig 1 shows the progress in packing density of magnetic storage for large computers [3]. Initially, it started with NRZI (Non Return to Zero Inverse) [3], but after several years it was found to be difficult to exceed the packing density of 1 KBPI (Kilo Bit Per Inch). In decoding NRZI, a clock information is necessary, which is provided by a separate track, and is limited by a phase margin between the clock and data which is defined by machanical tolerance between tracks.

A method called "self clock" was invented to derive clock information from data signal, and new channel codes PE (Phase Encoding = Bi Phase = Manchester Code) [3] and FM (Frequency Modulation) [3] made the packing density double.

However, this "self clock" was achieved by the sacrifice of "the window margin" and also "the minimum wave length to be recorded". Both are reduced half of NRZI. The minimum wave length is referred to be "the density ratio" and in this case the density ratio of PE or FM is 0.5 of NRZI.

This was improved by MFM (Modified FM) in which the density ratio is the same as that of NRZI but still "self clock" was kept with the aid of new technology of PLL (Phase Lock Loop). Again the packing density was doubled by MFM (Fig 1).

MFM has been considered as a standard channel coding for several years, but further improvements are studied toward the following three directions.

(1) DC-free code (Miller^2 [4] or ZM [5])

(2) Wider window margin (4/5 MNRZI [6])

(3) Higher Density Ratio (3 PM [7])

It was found that MFM is not optimum and all the directions showed fairly good results in the improvement of the packing density or the reliability. But it was impossible to discuss which direction is the best for magnetic recording, because it is deeply dependent on the characteristics of recording media or the design strategy of decoding circuits.

These improvements were achieved by an increase of hardware in encoding/decoding circuits, which is, however, very small compared with the total scale of systems.

Fig 2 shows the progress of the packing density of stationary-head digital audio recorders, which exactly traced 30-year history of computer peripherals but the period is shortened into five years. And the final packing density is five times higher than that of computer systems. This is because of the following reasons.

(1) The desired level of durability is slightly different.

(2) The decoding delay of digital audio recorders is not a serious problem and extremely strong error correction schemes are applied [1].

(3) A further progress in channel coding is studied.

A series of code named HDM (High Density Modulation) is developed and reported in this paper. HDM-1 is adopted for the proposed format for professional recorders [1].

As to the digital audio disk systems, the initial work started from FSK (Frequency Shift Keying) as is shown in Fig 3. This is because the study was based on video disk systems and the digital audio signal is modulated as pseudo-video signal and the rotation speed was the same as video; 1800 rpm. Soon the direct coding of MFM was tried and the packing density increased as five times, and the rotation speed was reduced to the half. The next step was the adoption of the channel code of higher density ratio such as 3 PM [7] and HDM-2 and then the packing density was increased twice and three times respectively, while the rotation speed was tried as 450 rpm and DLV (Constant Linear Velocity).

The final stage was the study for marketing. The interaction between channel coding and servo systems as well as yields of disks and players were carefully studied, and the packing density was reduced by 20% from the laboratory level. A new channel code named EFM (Eight to Fourteen Modulation) was developed especially for this application.

3. THE FUNDAMENTAL PARAMETERS FOR CHANNEL CODING (MODULATION SCHEME)

3-1 WINDOW MARGIN, PHASE MARGIN, JITTER MARGIN (Tw)

The tolerance of the location of the transition is called "window margin, Tw", "phase margin", or "jitter margin". The value of Tw is the larger, the better.

3-2 THE MINIMUM DISTANCE BETWEEN TRANSITION (Tmin), DENSITY RATIO (D.R.)

Tmin relates to the minimum wave length (λmin) to be recorded. λmin is one of the major factors which determine the packing density.

The spacing loss, caused by some particles attached on tapes or poor contact between head and tape, is greatly dependent on the wave length, and Tmin also relates to the reliability.

The ratio of Tmin to the length of the original one data bit-cell is referred as "density ratio".

3-3 THE MAXIMUM DISTANCE BETWEEN TRANSITION (Tmax)

In order to take "self-clock" strategy, the phase should be corrected at the transition point, thus Tmax must be finite and is the smaller, the better.

3-4 CLOCK RECOVERY

At the beginning of playing or after a long burst error, it is essential to pull in the decoding PLL circuit to the clock read from recorded media as quick as possible.

This is called "clock recovery", which relates to the following parameters.

(1) The value of Tmax or Tmax/Tmin; the smaller, the better.

(2) The probability of the transition; the higher, the better.

It is also noted that in order to pull in the servo of optical disc to CLV (Constant Linear Velocity), the clock frequency is often used and (2) is important.

3-5 DC OR LOW FREQUENCY CONTENT (DC)

Initially DC-free code is studied for the system which can not pass DC-component such as rotary-head recorders, but it is found that a lot of further merits are expected as follows.

(1) Decoding circuits can be simplified.

(2) The sensitivity against dusts, scratches, and finger prints can be reduced by inserting high-pass filter.

(3) The lower frequency band can be utilized for different purposes.

In optical disk systems, the servo signals for tracking and focusing are detected by the same photo detector of the main digital data, and DC-free code is greatly helpful to improve bandwidth as well as S/N of servo systems.

3-6 CONSTRAINT LENGTH (Lc)

Lc is defined as the length of prior bits which affect to the determination of the present "channel bit" (modulated bit).

If Lc is long, there is some possibility that an error of one channel bit will cause an error of many data bits at decoding stage. This is called "error propagation".

4 CONVENTIONAL CHANNEL CODES

Table 1 shows some of the conventional codes and their characteristics. Fig 4 is some examples of their wave form.

(i) NRZ (Non Return to Zero) [3]

The high and low level of wave from correspond to "1" and "0" of origial data, respectively.

(ii) NRZI (Non Return to Zero Inverse) [3]

"1" of original data corresponds to the transition, and "0", to the no-transition.

None of NRZ nor NRZI have applied to digital audio systems except very early models for experiments.

(iii) FSK (Frequency Shift Keying)

FSK is the same as Frequency Modulation of analogue signals; namely "1" and "0" of original data correspond to the different frequencies.

Most of rotary-head digital recorders such as various models based on EIAJ standard [14], PCM-1610 (Sony) [13], and BP-90 (JVC) [15], are using FSK. FSK requires wider frequency bandwidth but is stable and simple enough.

(iv) FM (Frequency Modulation = Double Frequency = Bi Phase-S)

FM in digital modulation schemes is the minimum frequency version of FSK, namely two transition at "1" and one transition at "0". Bi Phase-M is obtained by reversing "0" and "1" in FM.

(v) PE (Phase Encoding = Manchester Code = Bi Phase-L) [3]

The direction of transition (up or down) corresponds to "1" or "0" in original data. If the same code continues, another transition takes place in between.

FM and PE are almost identical in performance. Both are DC-free, both have transition once at least in one bit cell, and clock recovery is excellent. The only difference is FM is polarity free but PE is not.

FM and PE have not applied to digital audio systems for main signal recording except one or two experimental models, but are frequently used for auxiliary data recordings such as control tracks or time code tracks.

(vi) MFM (Modified Frequency Modulation = Delay Modulation = Miller Code) [3]

MFM is coded by the following rules.

(a) "1" Corresponds to transition.

(b) "0" corresponds to no-transition.

(c) If, "0's" are continuous, another transition is taken place at the edge of the bit cells.

MFM shows fairly good characteristics shown in Table 1, with self clock ability and relatively high density ratio (=1).

MFM is used in various digital audio recorders such as 3M [10], Mitsubishi [8], and Matsushita [9].

(vii) MILLER^2 (Miller Square) [4]

Miller^2 is one of the improved version of MFM, and is sometimes called as Modified MFM, MMFM or M^{2}FM, but there are so many modified version of MFM and the name Miller^2 is better to avoid confusion. The term comes from two different persons with the same name, original inventer of MFM (Mr. A. Miller) and the improver (Mr. J. W. Miller).

Fig 5 shows the difference between MFM and Miller^2. In MFM, the sequences of all "1" or all "0" are always DC-free because the high and the low level have the same length. It is also shown as DC-free when odd number of "1's" exist between two "0's". The problem is the even number of "1's" between two "0's", where the two "0's" become the same level and the balance of DC level is lost.

The modification in Miller^2 is just omit one transition at the final "1" of these even number of "1's", and as is shown in Fig 5, DC-content is much improved by the levels of two "0's" being different.

Miller^2 is better than MFM because it can be used for a system without DC transmission (as rotary-head recorders). It is also noted that error rate can be reduced by inserting high pass filter, and the low frequency band can be used for another purposes like servo control signals (optical disk systems) or system control signals.

Miller^2 is adopted to the proposed format from Ampex [11].

(viii) <u>ZM</u> (Zero Modulation) [5]

ZM is a kind of convolutional code transforming one data bit into two channel bits. The characteristics is almost identical with MFM except one point that DC-free is added. The basic principle requires infinite memory because it is using both the look-ahead parity counting bit number until the next "0" and the look-back parity counting all the number of "0's" from the beginning of the data sequence.

Practically it is solved easily because the size of the memory can be reduced to a certain length, if redundant data "0" is inserted every other interval of that length.

Normally frame synchronization pattern carries "0" and the size of memory is enough as long as one frame.

Compared with Miller[2], the encoder and the decoder of ZM is significantly complex and none of digital audio equipment is reported as using ZM.

(ix) <u>4/5 MNRZI</u> (Modified NRZI of 4/5 Rate) [6]

MNRZI is a method which converts m-bit of data bit into n-bit of channel bit and the channel bit is treated as NRZI. The most famous one is m=4 and n=5 (4/5-Rate), which is used as the high density tape storage of IBM.

The conversion table is shown in Table 2.

According to this conversion, the number of "0's" between "1" is equal or less than two, and Tmax = 2.4T, where T is the length of the original data bit cell. It is also found that the window margin Tw and the minimum length between transition Tmin are both equivalent to one channel bit cell which is equivalent to 0.8 of data bit cell.

$$Tw = Tmin = 0.8T$$

Tw is better than that of MFM, but Tmin is worse, therefore 4/5 MNRZI is suitable for recording systems which is not critical in wave length limitation but is marginal in jitter, signal to noise ratio or peak shift.

Sometimes the product of Tmin and Tw is referred as a figure of merit and in this case the value is 0.64 which is better than that of MFM (0.5) as is shown in Table 1.

(x) 3PM (Three Position Modulation)

3 PM is a kind of MNRZI, where m=3 and n=6, but the conversion is not linear at merging point. As is shown in the conversion table (Table 3), all "1's" are separated by two or more number of "0's".

$$T_{min} = 1.5T$$

While the window margin is the same as MFM.

$$T_w = 0.5T$$

Therefore the figure of merit Tmin x Tw is 0.75 which is better than 4/5 MNRZI.

At the merging point of the words shown in Table 3, sometimes the pattern of "101" appears which is a violation against Tmin = 1.5T. In this case "101" is replaced by "010" as is shown in Fig 4. In order to make this possible at any time, the last channel bit of Table 3 is always "0".

3PM is used for optical digital audio disk system [12] as well as proto type version of professional multi-channel recorder[16]. In both cases, it proved to bring on fairly good improvements in packing density as well as reliability, however, as to the final version of the formats of both disk and recorder systems, a further improvement was achieved which is discribed in the followings.

Fig 6 shows frequency spectrum of PE, MFM, and 3PM, where the bit rate is 3.2193 M bit/sec, thus the maximum frequency to be recorded of each channel code is as follows.

Code	Maximum Frequency to be Recorded
PE	3.2193 MHz
MFM	1.60965 MHz
3PM	1.0731 MHz

5. NEWLY DEVELOPED CHANNEL CODES

Table 4 shows newly developed channel coding and their characteristics. Fig 7 shows some examples of their wave form.

5-1 HDM (HIGH DENSITY MODULATION) SERIES

(xi) HDM-1

Fig 8 shows the coding rule of HDM-1.

For a pattern of "01" of input data bits, a transition only exists in the middle of the bit-cell "1". Consecutive "1's" are devided by groups of every 2 bits, and if the number of "1's" is odd the final group consists of three "1's". Then transition is set at the boundary of every group of two "1's" or three "1's".

When the above-mentioned input code includes a pattern of continuous "0" bits, the transition shall occur at the edge of the bit cell no less distant than 3.5T from the preceding transition and no less distant than 1.5T from the center of the immediately following "1".

The characteristics of HDM-1 is improved from 3PM at the following three points namely, Tmax becomes from 6T into 4.5T, constraint length is shorter from 9T into 5.5T, and hardware for encoder/decoder is much simpler. It is also noted that 3PM is based on 3-bit block which is not suitable for normal digital audio equipments basically designed as 16-bit word systems. HDM-1 is not a block code and does not have any restriction in blocking.

HDM-1 is adopted in the format of stationary recorders for professional use [1].

(xii) HDM-2

HDM-2 is an improved version of HDM-1 in Tmax, while the constraint length becomes a little bit worse.

The main difference lies on the sequence of "1" shown in Fig 8. Supposing there is a very long sequence of "1", the distance between transition in HDM-1 is 2T and the final distance can be either 2T or 3T. But in HDM-2, the final one is always 2T, because the final 2T-3T sequence is replaced by 3T-2T (see Fig 9 (1)).

Fig. 9 (2) shows a waveform of HDM-2 at "0" sequence, in which the minimum distance between transition is 3T and it is still distinct as 0's because the final distance of "1" sequence is always 2T.

Fig 10 shows a modulator and a demodulator of HDM-2, which need some memory to make "3T-2T" replacement.

Fig 11 shows the frequency spectrum of HDM-2.

(xiii) <u>HDM-3</u>

The basic principle of HDM-3 is different from HDM-1 or HDM-2, but it is a kind of MNRZI in which m=4 and n=12. The most significant point of HDM-3 is that Tmin = 2T, in other words the minimum wave length to be recorded is twice as long as that of MFM or four times of PE/FM. On the other hand, the window margin Tw is T/3 which is shorter than MFM, HDM-1 or HDM-2.

Therefore, HDM-3 is suitable for the recording system in which S/N or jitter margin are not severe, but wave length should be as long as possible by some reason, for instance, to improve reliability.

The basic conversion is shown in Table5. While the merging rule is non linear as is shown in Table 6. The word number 10 is a peculiar case and special rule (2) is required.

The maximum length between transition occurs at the following connection, and the value is 26 channel bits which is 8 2/3 data bits.

$$\left.\begin{array}{l}\#2\\ \#6\\ \#14\end{array}\right\} - \#12 - \#3 \left\{\begin{array}{l}\#12\\ \#13\\ \#14\\ \#15\end{array}\right.$$

Where the number corresponds to Table5.

The encoder of HDM-3 is based on Rom-Table and small additional circuit or software for merging rules. The rules for decoding is shown in Fig 12 which is rather simple, because the order of conversion table is carefully selected.

Fig 13 shows frequency spectrum of HDM-3.

(xiv) HDM-0, THE STRATEGY TO REDUCE DC CONTENT

It is well known that D.C. or low frequency content of channel coding can be reduced by adding some redundant bits and adopt appropriate strategy.

Fig 14 shows examples of adding two redundant bits to every 32 or 46 data bits, where the channel code is HDM-2. The spectrum of original HDM-2 is rather flat between DC and 20 KHz excluding some fluctuation caused by FFT. On the other hand, a little redundancy of 4% (46+2) or 6% (32+2) makes great improvement of spectrum content below 5 KHz.

DC content of channel coding is often evaluated by DSV (Digital Sum Value), which is an amount of residual DC level from the beginning of the bit stream where high and low level in each channel bit are counted as "+1" and "-1", respectively.

In above mentioned cases of Fig 14, the redundant bits are two and there are four possibilities, 00, 01, 10, and 11, in their pattern. The selection of the pattern is decided by either of, or the combination of the following strategies.

(a) To select the redundant bit pattern so as to minimize the DSV at the end of the next block.

(b) To select the pattern which will make the maximum number of zero crossing of DSV, during the next block period.

(c) To select the pattern so as to minimize the maximum peak value of DSV, during the next period.

(d) Same as (c), but watch two or more blocks instead of one block.

Fig 15 shows the DSV of the original HDM-2, which is of course not stable at all. Fig 16 shows that of HDM-0 with adding 2 redundant bits to HDM-2.

5-2 CHANNEL CODING FOR COMPACT DISC DIGITAL AUDIO

The preconditions for the development of Compact Disc Digital Audio System were as follows.

(1) The size of the disc should be as small as possible. The maximum diameter is determined by the size of the dash board of automobiles which is approximately 12cm.

(2) The playing time should be more than one hour, hopely 75 minutes.

(3) The wave length of laser source is not shorter than 0.78μm.

(4) Yields of disc should be as good as possible, hopely final inspection can be omitted.

(5) The bandwidth for both tracking and focus servo systems should be as wide as possible, in order to make automobile application be possible.

(6) The hardware for demodulator should be as simple as possible.

These conditions are very severe and the realization had been impossible until a new channel coding was developed [2], [7].

In order to make the yield of discs better, the numerical aperure of the objective lenses should be as small as possible, which is conflicting to the conditions (1) and (2). Thus the density ratio should be greater than 1 and lower frequency component should be greatly suppressed for the condition (5).

The other conditions are:

(7) It should be easy to pull in the CLV (Constant Linear Velocity) servo control.

(8) Since the error correction scheme is based upon 8-bit block, it is better that the error propagation does not exceed the boundary of the block.

(xv) EFM (Eight to Fourteen Modulation)

The basic conversion of EFM is a kind of MNRZI with m=8 and n=14, but three more channel bits are added for merging and for the suppresion of low frequency component.

The conversion table was developed by a special computer algorithm to select $2^8 = 256$ patterns out of $2^{14} = 16,384$ patterns under the following conditions.

'''Every "1" in channel bits (corresponding to the transition) should be separated by a number of "0's" between two and ten.'''

It was found that there are 277 patterns covering above condition and 21 patterns are dropped by relatively pour characteristics in merging points or transition probability.

The redundant three bits are determined by the similar strategy as HDM-0, but it should be noted that above mentioned condition should be kept also at the merging points.

Fig 17 shows one of the examples of this strategy. At the merging point between the blocks #1 and #2, the pattern "000" can not be inserted because the number of consecutive "0's" will be twelve, which is the violation of the above mentioned condition. Therefore, there are three possibilities in redundant bit patterns, namely "100", "010", and "001", which are named Case A, B and C, respectively.

Similarly, the merging bit patterns between blocks #2 and # 3 have two possibility which is indicated in Fig 17 as case D and case E.

DSV of all the possibilities are shown in Fig 18. If one block strategy is taken, then A→E will be selected.

More precise strategy is, of course, possible by the sacrifice of hardware and/or software for decision making, however, the cost of the machine will not be a great problem, because they will be set besides cutting machine of Compact Disc which will be fairly large and expensive.

The decoding is, on the other hand, very simple, because the redundant bits can merely skipped and the rest is decoded by a look-up table.

Fig 19 shows frequency spectrum of EFM.

An additional feature of EFM is, because it is a linear block code, any error propagation out of one block (8 bit) is not expected. Therefore this code is quite suitable for Reed-Solomon structure over Galois Field (2^8) [2].

While any other codes described above has the possibility of error propagation beyond the boundary of the blocks, which will make one-block error into double-block error and the correctability will get worse.

6. CONCLUSION

Channel codes (modulation schemes) for digital audio recordings is described here. Initially, the history of channel codes is discussed with the relation to the packing density of computer peripherals, stationary head digital audio recorders and digital audio disk systems. Then the fundamental parameters for the evaluation of channel coding are described, and conventional codes with their application to the present digital audio systems are explained.

In the beginning, the progress of digital audio systems just followed that of computer peripherals in terms of channel codes. However, due to severe requirements of digital audio systems in packing density and reliability, as well as interaction between digital data and servo signals, a great progress was achieved which made the systems more practical.

A series of codes named HDM is developed and applied to the stationary head recorder for professional use, which realized 48 channels on one-inch tape, 24 channels on half inch, and 8 channels on quater inch at the speed of 30 i.p.s..

The condition for the channel coding for Compact Disc Digital Audio is rather special, because it accepts huge, expensive encoder as far as decoding is simple enough. On account of this condition, an unique high performance channel code, named EFM (Eight to Fourteen Modulation), is developed and fulfilled severest requirements of optical systems.

ACKNOWLEDGEMENT

The author wishes to thank Mr. M. Tanaka and Mr. S. Furukawa of Sony Corporation for their contribution in development of HDM, and also Mr. H. Ogawa of Sony Corporation and Mr. K. A. Immink of N.V. Philips for their contribution in development of EFM.

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Items	Symbols	(i) (ii) NRZ NRZI	(iv) (v) PE FM	(vi) MFM	(vii) Miller^2	(viii) ZM	(ix) 4/5MNRZI	(x) 3PM
1 Window Margin	Tw	T	0.5T	0.5T	0.5T	≃0.5T	0.8T	0.5T
2. Minimum Transition	Tmin	T	0.5T	T	T	≃T	0.8T	1.5T
3 Maximum Transition	Tmax		T	2T	3T	≃2T	2.4T	6T
4 D.C. Content	DC	Bad	Good	Bad	Good	Good	Bad	Bad
5 Constraint Length	Lc	T	T	T	3T	depend on redundancy	4T	9T
6 Clock Rate	CLK	No	2/T	2/T	2/T	2/T	1.25/T	2/T
7 Density Ratio	D.R.	1	0.5	1	1	≃1	0.8	1.5
8 Figure of Merit	Tw×Tmin (T^2)	1	0.25	0.5	0.5	0.5	0.64	0.75
9 Max-Min Ratio	Tmax/Tmin		2	2	3	2	3	4
References		(3)	(3)	(3) (8) (9) (10)	(4) (11)	(5)	(6)	(7) (12)

Table.1 Parameters of Conventional Channel Codes

T = The length of one bit of original

Table 2 The Conversion Table of 4/5MNRZI

No	Data Bit	Channel Bit
0	0000	11001
1	0001	11011
2	0010	10010
3	0011	10011
4	0100	11101
5	0101	10101
6	0110	10110
7	0111	10111
8	1000	11010
9	1001	01001
10	1010	01010
11	1011	01011
12	1100	11110
13	1101	01101
14	1110	01110
15	1111	01111

Table 3 The Conversion Table of 3PM

No	Data Bit	Channel Bit
0	000	000010
1	001	000100
2	010	010000
3	011	010010
4	100	001000
5	101	100000
6	110	100010
7	111	100100

Items	Symbols	(xi) HDM-1	(xii) HDM-2	(xiii) HDM-3	(xiv) HDM-0	(xv) EFM
1 Window Margin	Tw	0.5T	0.5T	0.33T	0.33T~0.5T	0.471T
2 Minimum Transition	Tmin	1.5T	1.5T	2T	1.5T~2T	1.41T
3 Maximum Transition	Tmax	4.5T	4T	8.33T	4T~8.33T	5.18T
4 D.C. Content	D.C.	Bad	Bad	Bad	Good	Good
5 Constraint Length	Lc	5.5T	7.5T	12T	3T~12T	8T
6 Clock Rate	CLK	2/T	2/T	3/T	2/T~3/T	17/8T
7 Density Ratio	D.R.	1.5	1.5	2	1.5~2	1.41
8 Figure of Merit	Tw*Tmin(T^2)	0.75	0.75	0.67	0.67~0.75	0.664
9 Max-Min Ratio	Tmax/Tmin	3	2.67	4.17	2.67~4.17	3
Remarks		reference [1]		m=4 n=12 Non Linear Transformation	DC Content is improved by adding some redundancy	reference [2] [17]

Table.4 Parameters of Newly Developed Channel Codes

Table 5 Basic Conversion Table for HDM-3

No.	Data Bits		Channel Bits
0	0 0 0 0	↔	0 0 0 0 0 1 0 0 0 0 0 0
1	0 0 0 1		0 0 0 0 0 0 1 0 0 0 0 0
2	0 0 1 0		0 0 0 0 0 0 0 1 0 0 0 0
3	0 0 1 1		0 0 0 0 0 0 0 0 1 0 0 0
4	0 1 0 0		0 1 0 0 0 0 0 0 0 0 0 0
5	0 1 0 1		0 0 0 0 1 0 0 0 0 0 0 0
6	0 1 1 0		0 1 0 0 0 0 0 1 0 0 0 0
7	0 1 1 1		0 1 0 0 0 0 0 0 1 0 0 0
8	1 0 0 0		0 0 0 1 0 0 0 0 0 0 0 0
9	1 0 0 1		0 0 1 0 0 0 0 0 0 0 0 0
10	1 0 1 0		0 0 0 1 0 0 0 0 0 1 0 0
11	1 0 1 1		0 0 1 0 0 0 0 0 1 0 0 0
12	1 1 0 0		1 0 0 0 0 0 0 0 0 0 0 0
13	1 1 0 1		1 0 0 0 0 0 1 0 0 0 0 0
14	1 1 1 0		1 0 0 0 0 0 0 1 0 0 0 0
15	1 1 1 1		1 0 0 0 0 0 0 0 1 0 0 0

Table 6 Merging Rule of HDM-3

(1) Conversion table for merging point

0 1 0 0 0	1 0	←→	0 0 0 0 1	0 0
0 1 0 0 0	0 1		0 0 0 1 0	0 0
1 0 0 0 0	1 0		0 0 1 0 0	0 0
	connecting point			connecting point

(2) Conversion table for (10-α) merging

(α)		(10)	(α)
10 - 4	←→	0 0 0 1 0 0 0 0 0 0 0 1	0 0 0 0 0 0 0 0 0 0 0 0
10 - 6		0 0 0 1 0 0 0 0 0 0 0 1	0 0 0 0 0 1 0 0 0 0 0 0
10 - 7		0 0 0 0 1 0 0 0 0 0 0 1	0 0 0 0 0 1 0 0 0 0 0 0
10 - 9		0 0 0 0 1 0 0 0 0 0 1 0	0 0 0 0 0 0 0 0 0 0 0 0
10 - 11		0 0 0 0 1 0 0 0 0 0 1 0	0 0 0 0 0 1 0 0 0 0 0 0
10 - 12		0 0 0 1 0 0 0 0 0 0 1 0	0 0 0 0 0 0 0 0 0 0 0 0
10 - 13		0 0 0 1 0 0 0 0 0 0 1 0	0 0 0 0 1 0 0 0 0 0 0 0
10 - 14		0 0 0 1 0 0 0 0 0 0 1 0	0 0 0 0 0 1 0 0 0 0 0 0
10 - 15		0 0 0 0 1 0 0 0 0 0 1 0	0 0 0 0 1 0 0 0 0 0 0 0

FIG 1   PACKING DENSITY OF MAGNETIC STORAGE FOR COMPUTER

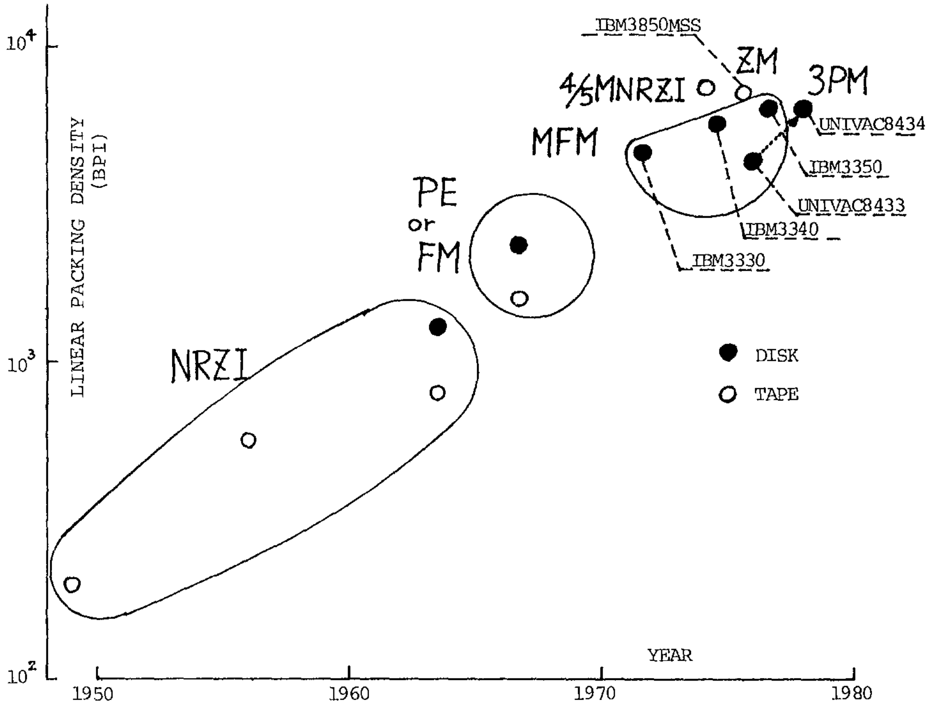


FIG 2 PACKING DENSITY OF STATIONARY-HEAD DIGITAL AUDIO RECORDER

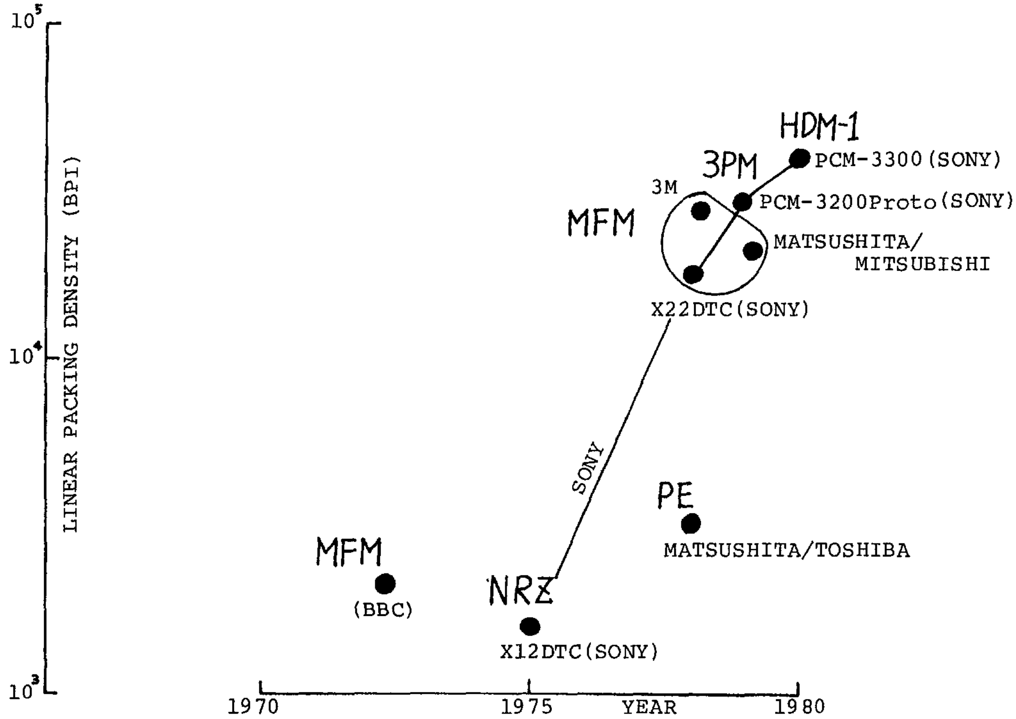
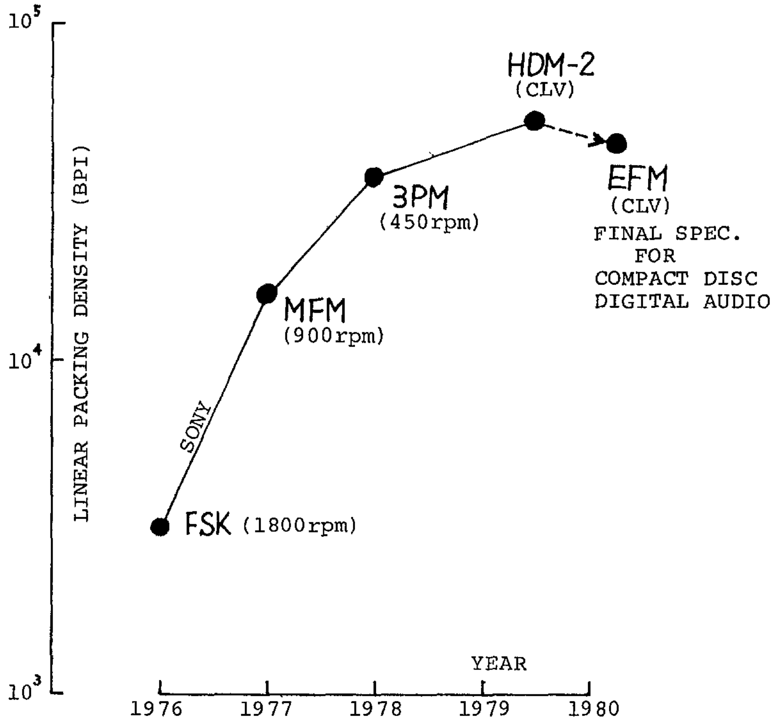


FIG 3 PACKING DENSITY OF OPTICAL DIGITAL AUDIO DISK



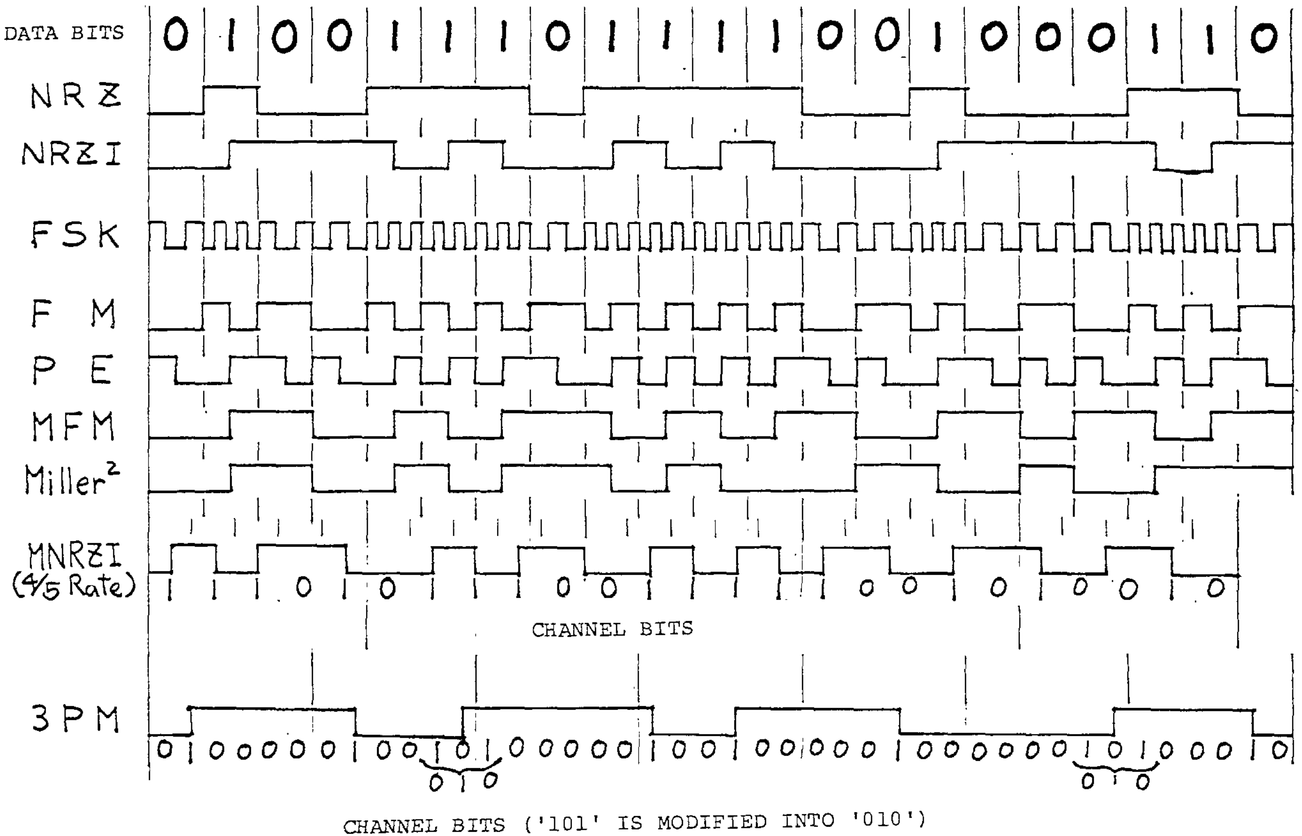
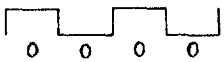


Fig 4 WAVE FORM OF CONVENTIONAL CHANNEL CODING

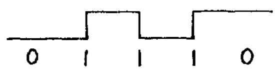
Fig 5 Miller^2

"MFM Code"



All "0" is DC free.

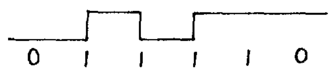
All "1" is DC free.



Odd number of "1's" between "0s" is DC free.

Even number of "1's" between "0's" is <u>not</u> DC free.

"Miller^2 Code"



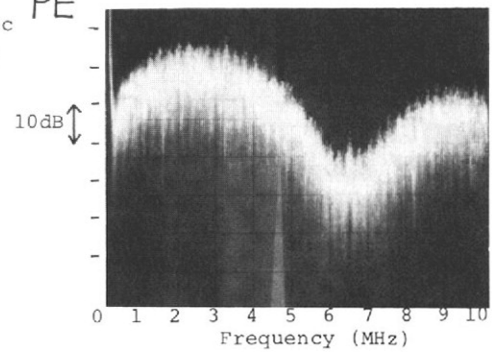
When even number of "1's" exist between "0s", the transition of the final "1" is omitted.

Fig 6   Frequency Spectrum of Channel Codes

PE

bit rate 3.2193Mbit/sec

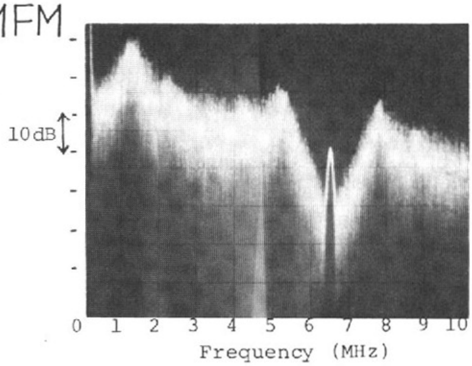
Maximum Freq. 3.2193MHz



MFM

bit rate 3.2193Mbit/sec

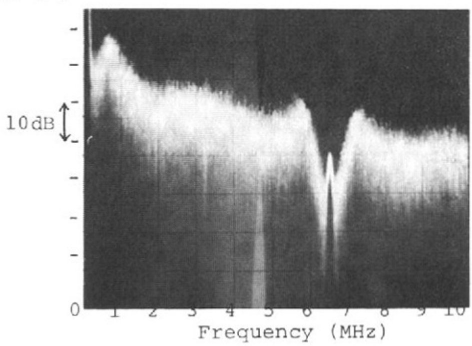
Maximum Freq. 1.60965MHz



3PM

bit rate 3.2193Mit/sec

Maximum Freq. 1.0731MHz



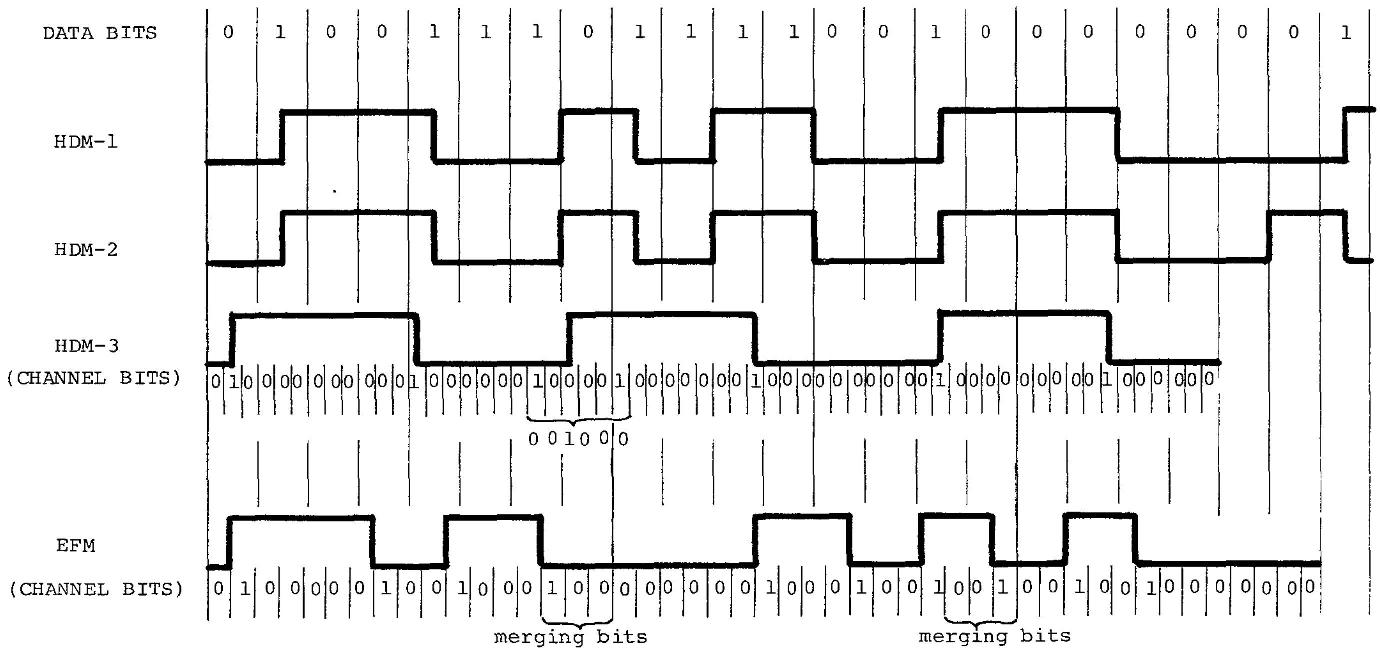
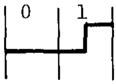


Fig 7 Wave Form of Newly Developed Channel Codes

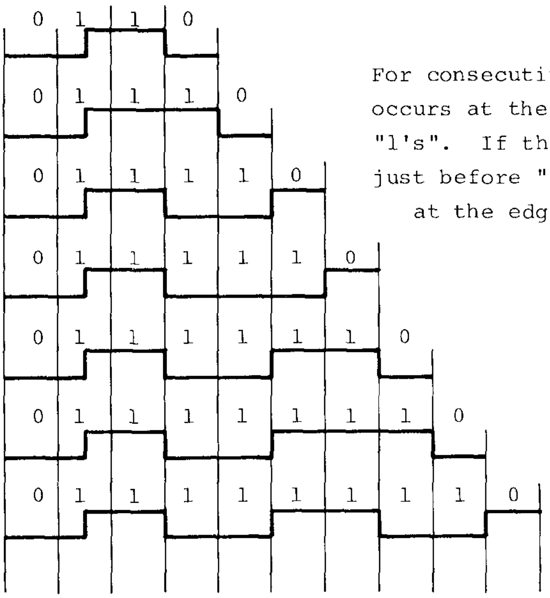
Fig 8 Coding Rules of HDM-1

(1) 0 → 1



Transition at the center of "1" bit cell.

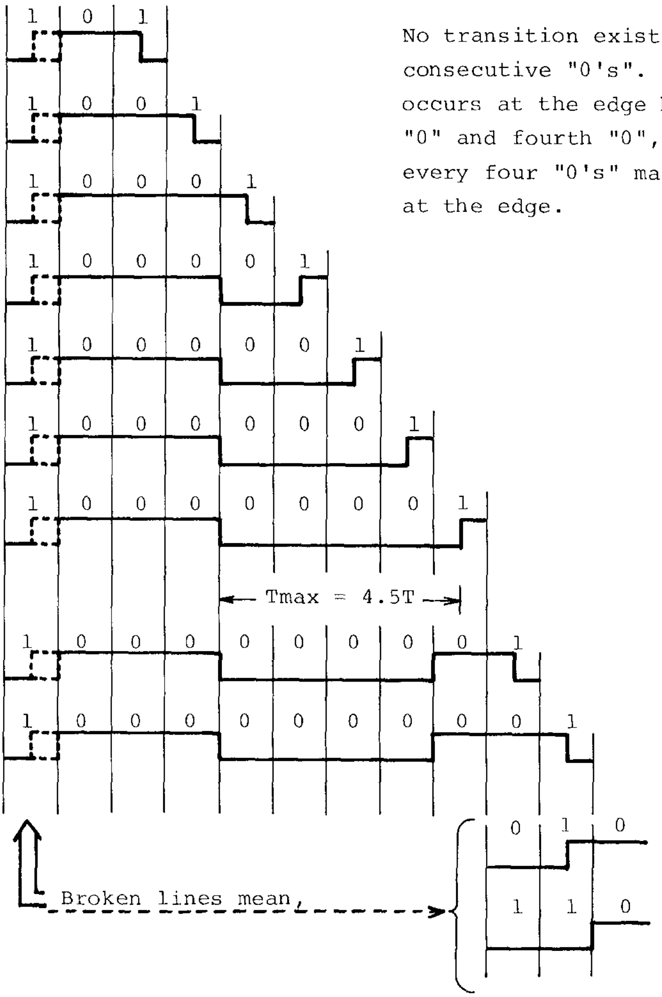
(2) Consecutive "1"



For consecutive "1", transition occurs at the edge of every two "1's". If three "1's" are left just before "0", transition occurs at the edge between "1" and "0".

Fig 8 (Continued)

(3) Consecutive "0"



No transition exists up to three consecutive "0's". Transition occurs at the edge between third "0" and fourth "0", after that every four "0's" makes transition at the edge.

Fig 9 Coding Rules of HDM-2

(1) Consecutive "1's"

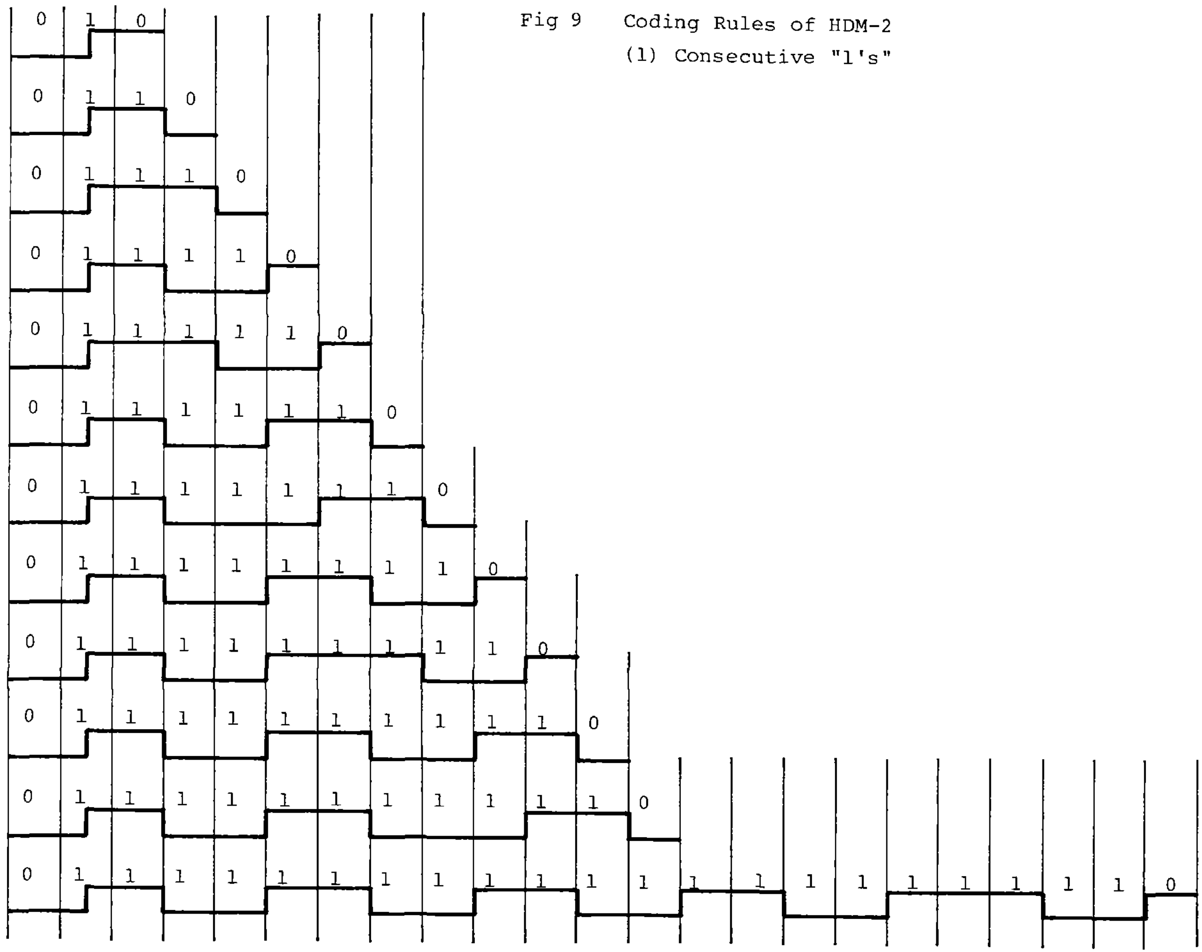


Fig 9 (Continued) (2) Consecutive "0's"

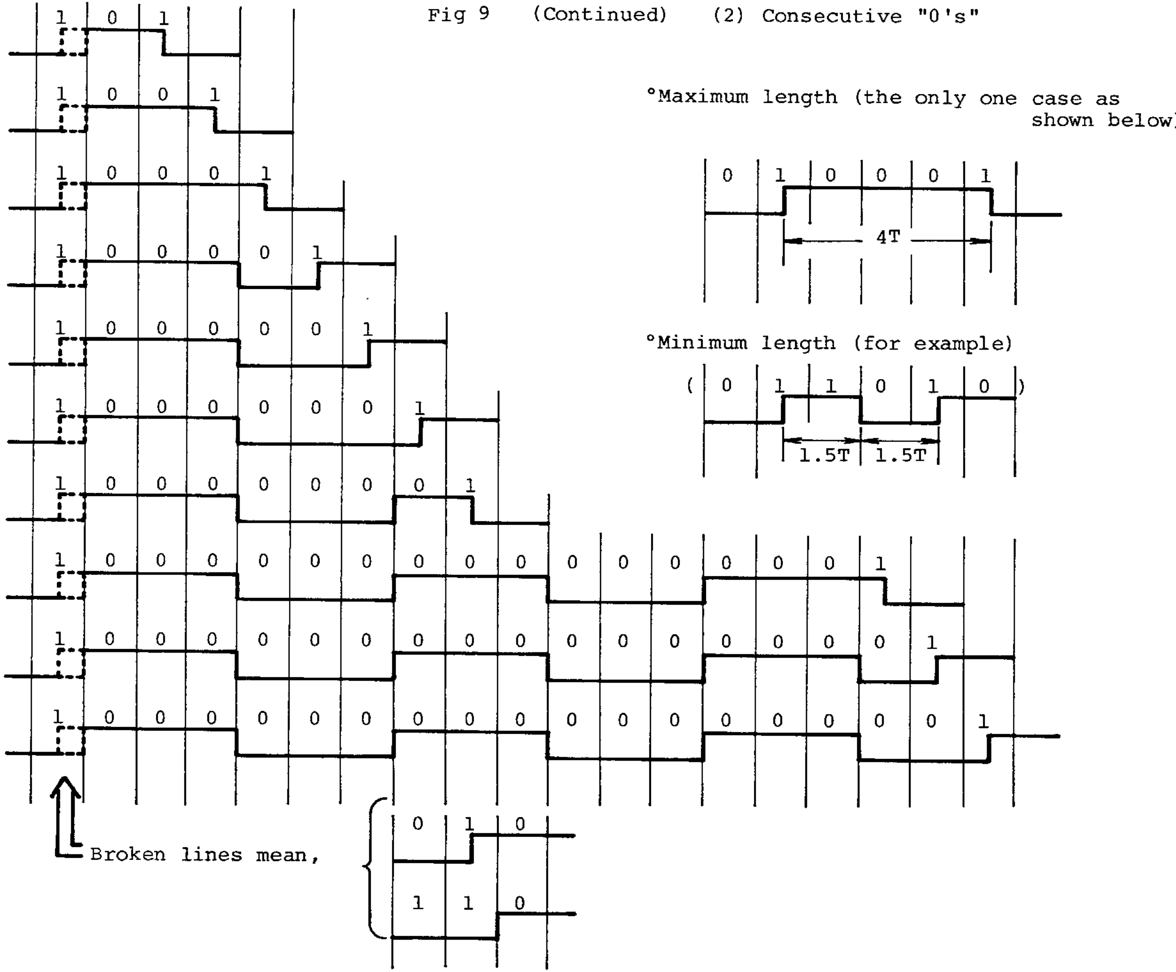
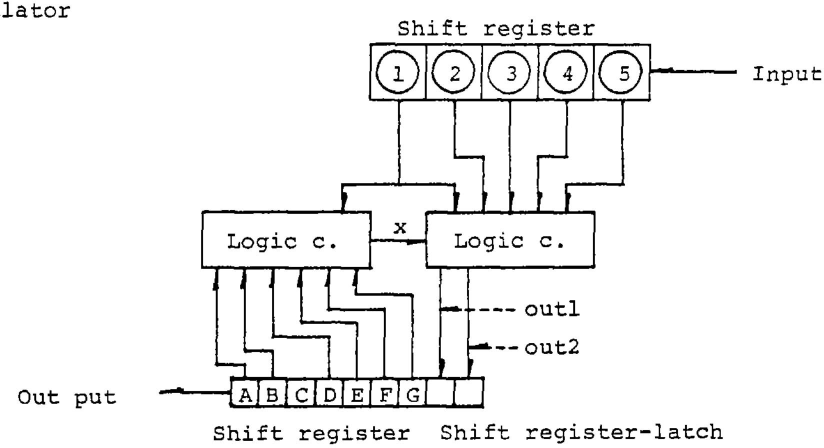


Fig. 10 Modulator & demodulator of HDM-2

∘ Modulator



$$x = (A+B)\cdot\overline{(D+E)}\cdot\overline{(F+G)}\cdot\overline{1} + (F+G)\cdot 1$$

$$out1 = x\cdot\overline{1}\cdot\overline{2} + \overline{x}\cdot 1\cdot(\overline{2}+3\cdot\overline{4\cdot\overline{5}})$$

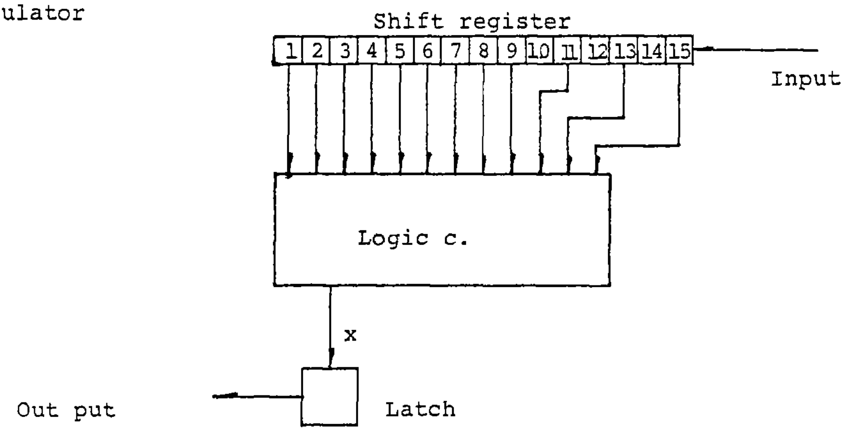
$$out2 = \overline{1}\cdot 2$$

or

$$out1 = [(A+B)\cdot\overline{(D+E)}\cdot\overline{2}+1\cdot(\overline{2}+3\cdot\overline{4\cdot\overline{5}})]\cdot\overline{(F+G)}$$

$$out2 = \overline{1}\cdot 2$$

∘ Demodulator



$$x = 6+5\cdot\overline{8}\cdot 11\cdot 15+9\cdot(3\cdot 13+4+5)$$
$$+7\cdot(1\cdot 11+2+3+4)$$

Fig 11 Frequency Spectrum of HDM-2

bit rate = 2.0338 Mbit/sec

Maximum Freq. = 677.93 KHz

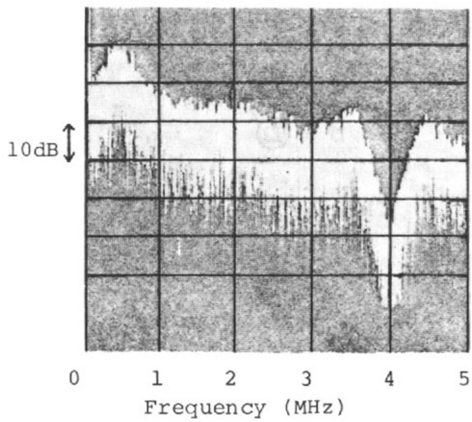


Fig 12 Basic Decoder for HDM-3

A	B	C	D	1	2	3	4	5	6	7	8	9	10	11	12

where A,B,C and D are defined by the prior word's bits $1' \sim 12'$ as follows;

$$A = (4' + 5') \cdot (11' + 12') + 4' \cdot 10'$$

$$B = 4' \cdot (11' + 12') + 10'$$

$$C = 11'$$

$$D = 12'$$

The 4 bits to be decoded

E	F	G	H

are defined as follows;

1° When $A \cdot (C + D) = 0$,

$$E = \overline{A} \cdot (B + D) + 1 + 3 + 4 + I$$

$$F = \overline{A} \cdot B + C + D + 1 + 2 + 5 \cdot \overline{I}$$

$$G = 8 + 9 + 10 + 11 + 12$$

$$H = 3 + 7 + 9 + (5 + 11 + 12) \cdot \overline{I}$$

where $I = (4 + 5) \cdot (11 + 12)$

2° When $A \cdot (C + D) = 1$

$$E = C$$

$$F = B + D + 5$$

$$G = \overline{B} \cdot 5 + 6$$

$$H = \overline{B} + 5$$

Fig 13 Frequency Spectrum of HDM-3

bit rate = 3.5721 Mbit/sec

Maximum Freq. = 893.025 KHz

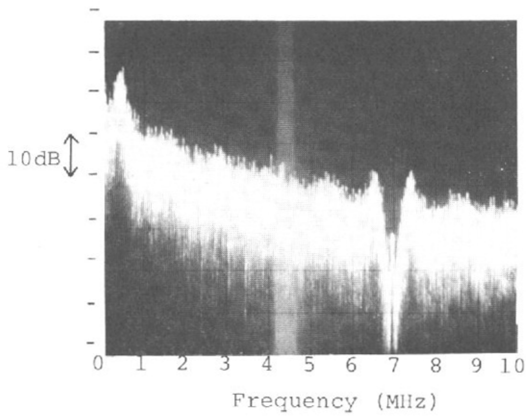


FIG. 14 * HDM-2 SIMULATION *

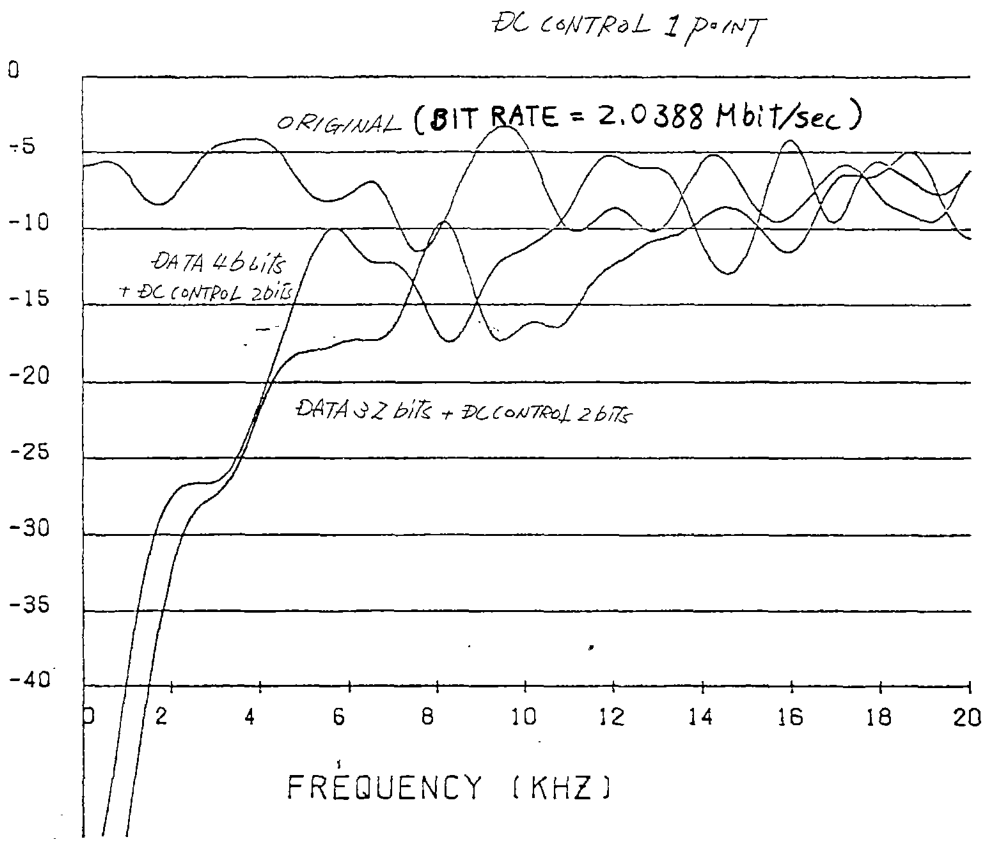
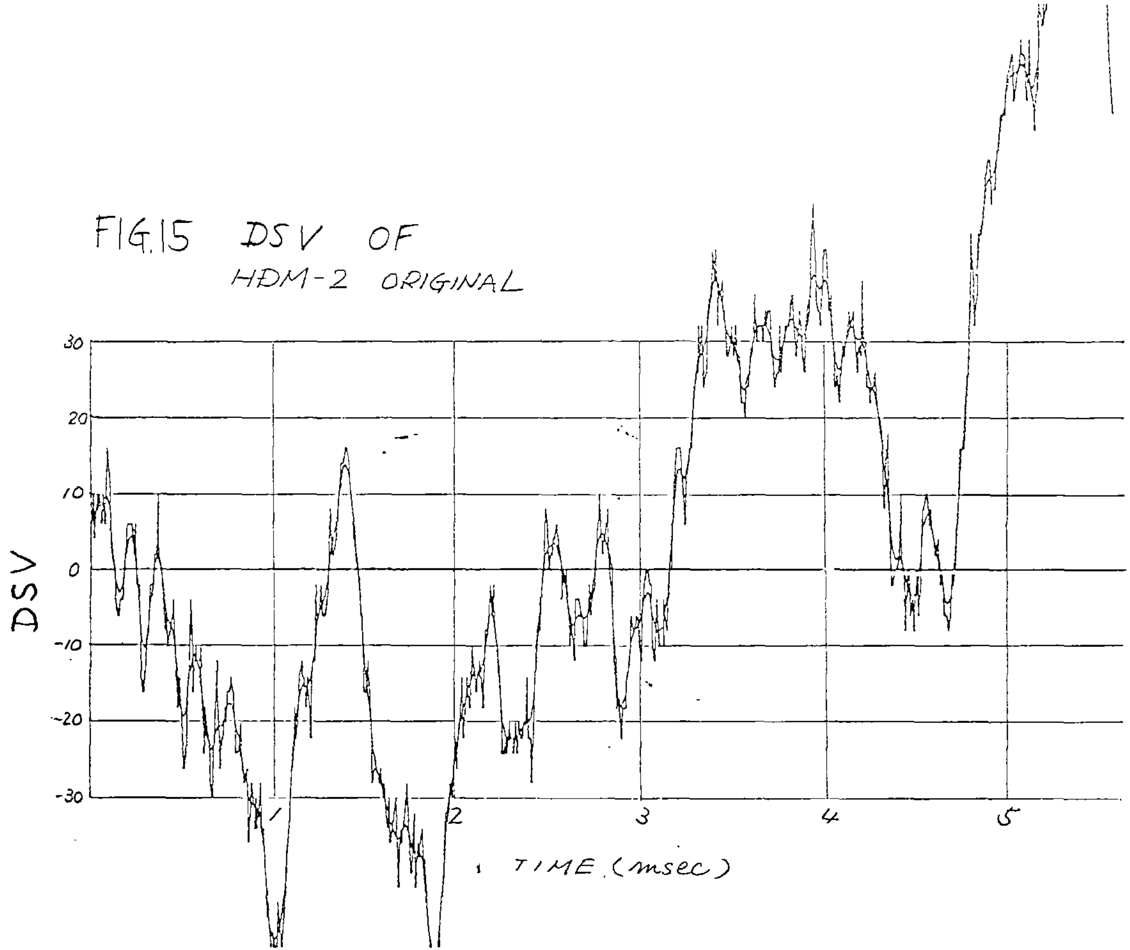


FIG. 15 DSV OF HDM-2 ORIGINAL



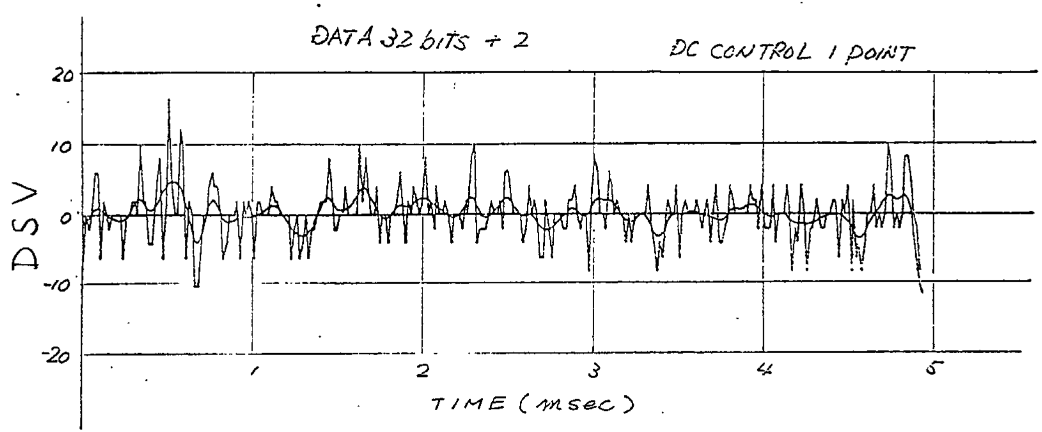
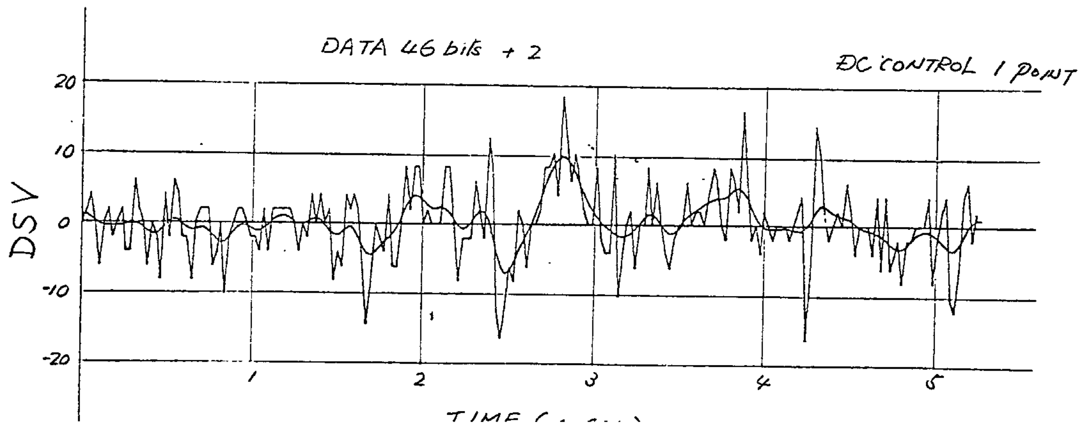


FIG. 16 $\overline{DSV}$ OF HDM-0 (MODIFIED HDM-2)



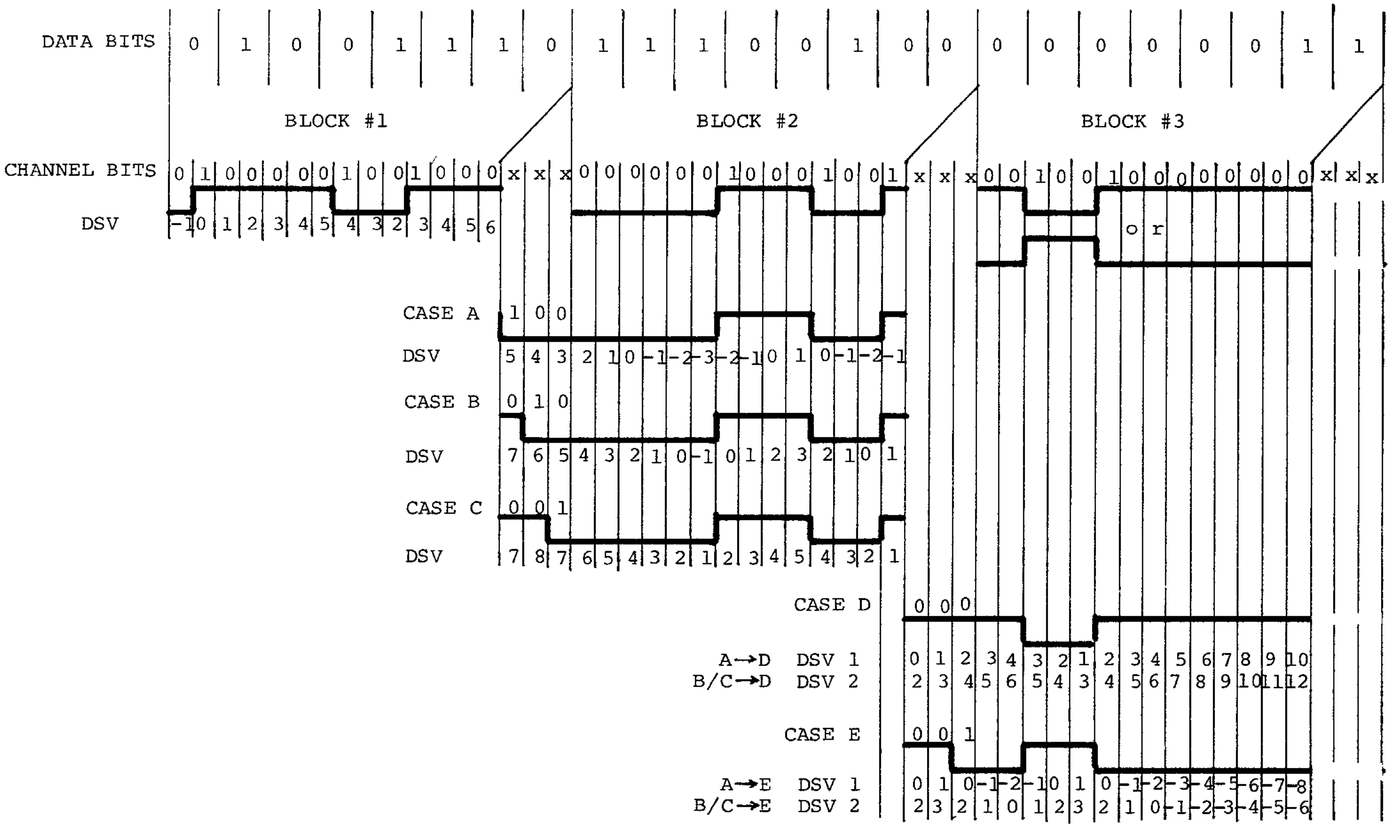


Fig 17 The Strategy of EFM

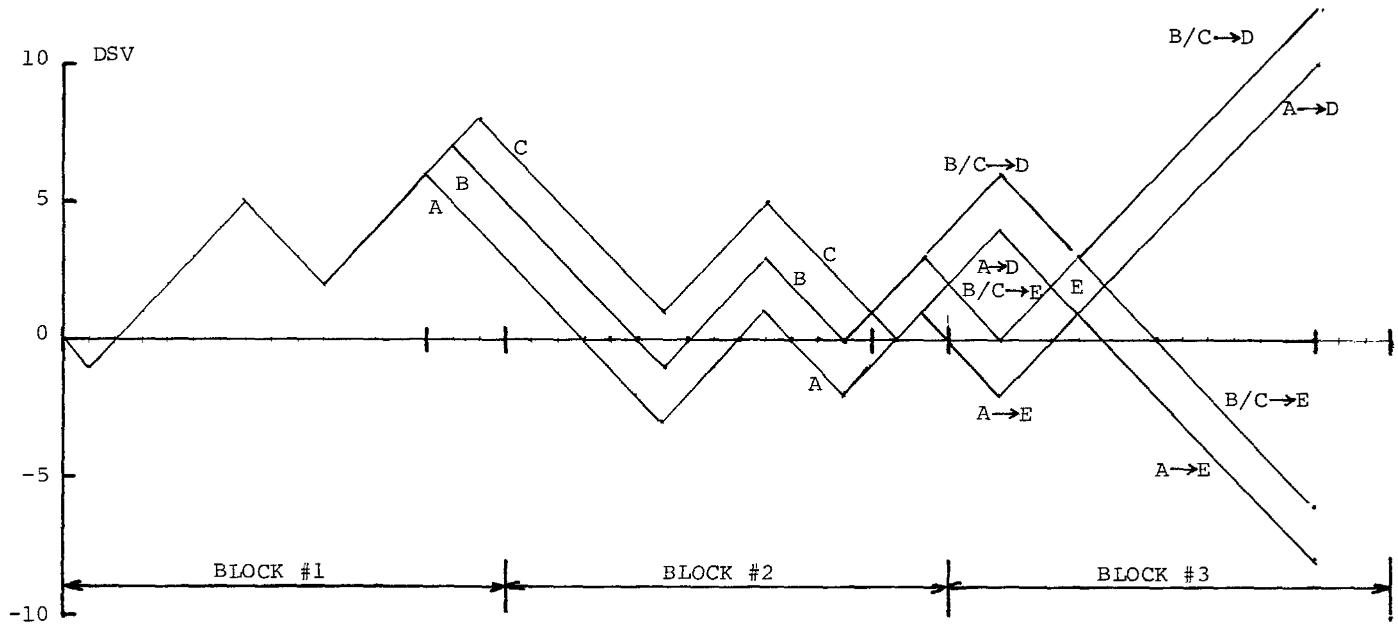


Fig 18 DSV of Each Selection in EFM

Fig 19 Frequency Spectrum of EFM

bit rate = 2.0338 Mbit/sec

Maximum Freq. = 721.2 KHz

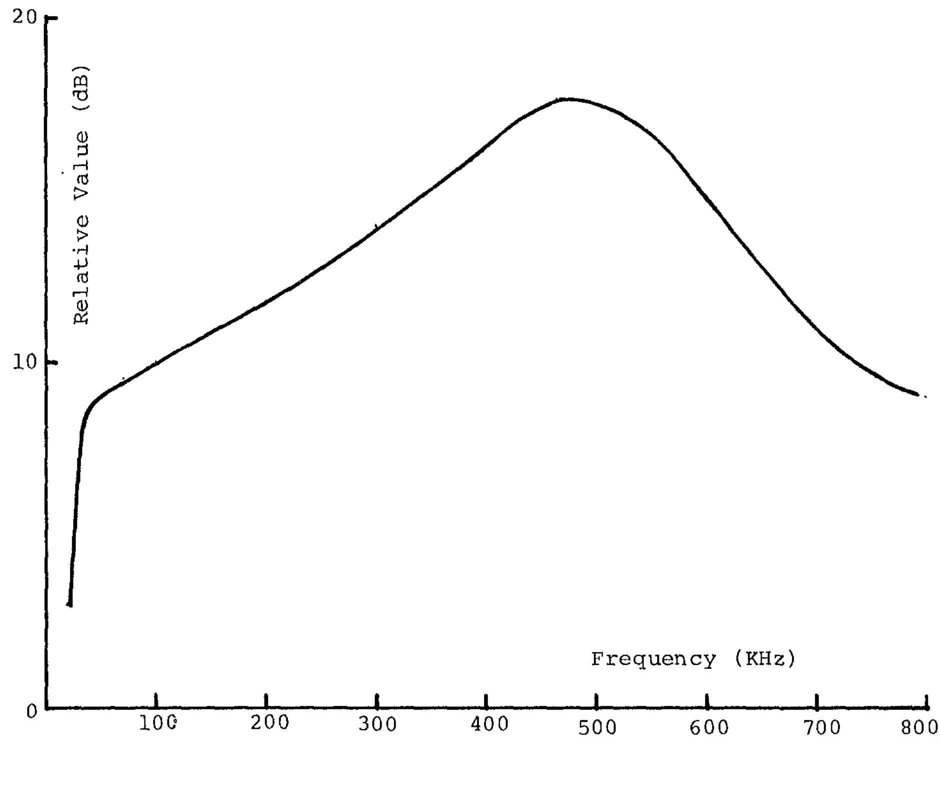


Fig. 19 (continued)