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Compact Disc: system aspects and modulation

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In this article we shall deal in more detail with the various factors that had to be weighed one against the other in the design of the Compact Disc system. In particular we shall discuss the EFM modulation system ('Eight-to-Fourteen Modulation'), which helps to produce the desired high information density on the disc.

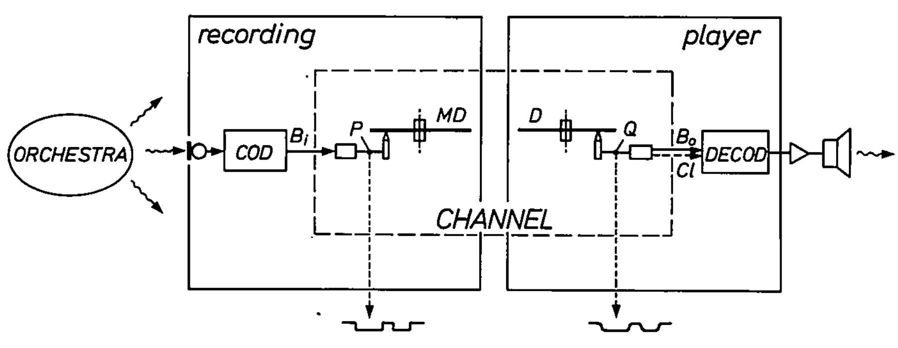

Fig. 1. The Compact Disc system, considered as a transmission system that brings sound from the studio into the living room. The transmission channel between the encoding system (*COD*) at the recording end and the decoding system (*DECOD*) in the player, 'transmits' the bit stream B_i to *DECOD* via the write laser, the master disc (*MD*), the disc manufacture, the disc (*D*) in the player and the optical pick-up; in the ideal case B_o is the same as B_i. The bits of B_o, as well as the clock signal (*Cl*) for further digital operations, have to be detected from the output signal of the pick-up unit at *Q*.

Fig. 1 represents the complete Compact Disc system as a 'transmission system' that brings the sound of an orchestra into the living room. The orchestral sound is converted at the recording end into a *bit stream* B_i, which is recorded on the master disc. The master disc is used as the 'pattern' for making the discs for the user. The player in the living room derives the bit stream B_o — which in the ideal case should be identical to B_i — from the disc and reconverts it to the orchestral sound. The system between *COD* and *DECOD* is the actual *transmission channel*; B_i and B_o consist of 'channel bits'.

Fig. 2 shows the encoding system in more detail. The audio signal is first converted into a stream B_1 of 'audio bits' by means of pulse-code modulation. A number of bits for 'control and display' (C&D) and the parity bits for error correction are then added to the bit stream [1][2]. This results in the 'data bit stream' B_2. The modulator converts this into channel bits (B_3). The bit stream B_i is obtained by adding a synchronization signal.

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[1] M. G. Carasso, J. B. H. Peek and J. P. Sinjou, The Compact Disc Digital Audio system, this issue, p. 151.
[2] H. Hoeve, J. Timmermans and L. B. Vries, Error correction and concealment in the Compact Disc system, this issue, p. 166.

The number of data bits n that can be stored on the disc is given by:

$$n = \eta A/d^2,$$

where A is the useful area of the disc surface, d is the diameter of the laser light spot on the disc and η is the 'number of data bits per spot' (the number of data bits that can be resolved per length d of track). A/d^2 is the number of spots that can be accomodated side by side on the disc. The information density n/A is thus given by:

$$n/A = \eta/d^2. \quad (1)$$

The spot diameter d is one of the most important parameters of the channel. The modulation can give a higher value of η. We shall now briefly discuss some of the aspects of the channel that determine the specification for the modulation system.

We shall consider one example here to illustrate the way in which such tolerances affect the design: the choice of the 'spot diameter' d. We define d as the half-value diameter for the light intensity; we have

$$d = 0.6\,\lambda/NA,$$

where λ is the wavelength of the laser light and NA is the numerical aperture of the objective. To achieve a high information density (1) d must be as small as possible. The laser chosen for this system is the small CQL10 [3], which is inexpensive and only requires a low voltage; the wavelength is thus fixed; $\lambda \approx 800$ nm. This means that we must make the numerical aperture as large as possible. With increasing NA, however, the manufacturing tolerances of the player and the disc rapidly become smaller. For example, the tolerance in the local 'skew' of the disc (the 'disc tilt') relative to the objective-lens axis is proportional to NA^{-3}. The

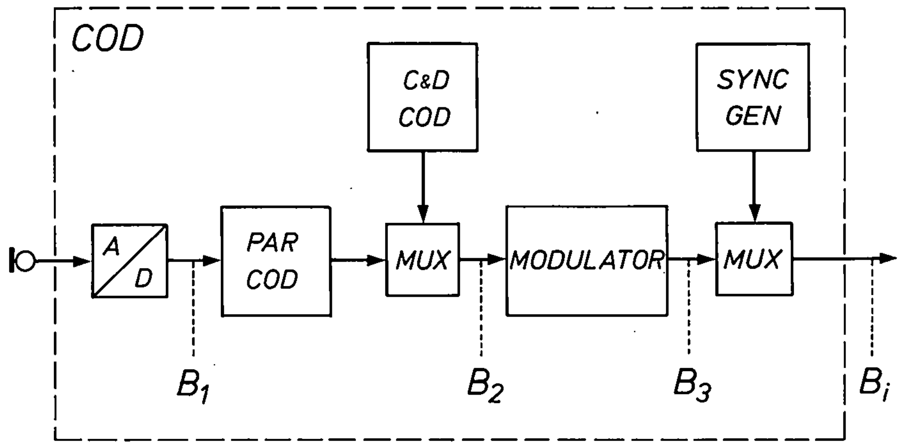

Fig. 2. The encoding system (*COD* in fig. 1). The system is highly simplified here; in practice for example there are two audio channels for stereo recording at the input, which together supply the bit stream B_1 by means of PCM, and the various digital operations are controlled by a 'clock', which is not shown. The bit stream B_1 is supplemented by parity and C&D (control and display) bits (B_2), modulated (B_3), and provided with synchronization signals (B_i). *MUX*: multiplexers. Fig. 9 gives the various bit streams in more detail.

The channel

The bit stream B_i in fig. 1 is converted into a signal at P that switches the light beam from the write laser on and off. The channel should be of high enough quality to allow the bit stream B_i to be reconstituted from the read signal at Q.

To achieve this quality all the stages in the transmission path must meet exacting requirements, from the recording on the master disc, through the disc manufacture, to the actual playing of the disc. The quality of the channel is determined by the player and the disc: these are mass-produced and the tolerances cannot be made unacceptably small.

tolerance for the disc thickness is proportional to NA^{-4}, and the depth of focus, which determines the focusing tolerance, is proportional to NA^{-2}. After considering all these factors in relation to one another, we arrived at a value of 0.45 for NA. We thus find a value of 1 μm for the spot diameter d.

The quality of the channel is evaluated by means of an 'eye pattern', which is obtained by connecting the point Q in fig. 1 to an oscilloscope synchronized with the clock for the bit stream B_o; see *fig. 3a*. The signals originating from different pits and lands are superimposed on the screen; they are strongly rounded, mainly because the spot diameter is not zero and the pit walls are not vertical. If the transmission quality is adequate, however, it is always possible to determine whether the signal is positive or negative at the 'clock times' (the dashes in fig. 3*a*), and hence to reconstitute the bit stream. The lozenge pattern around a dash in this case is called the 'eye'. Owing to channel imperfections the eye can become obscured; owing

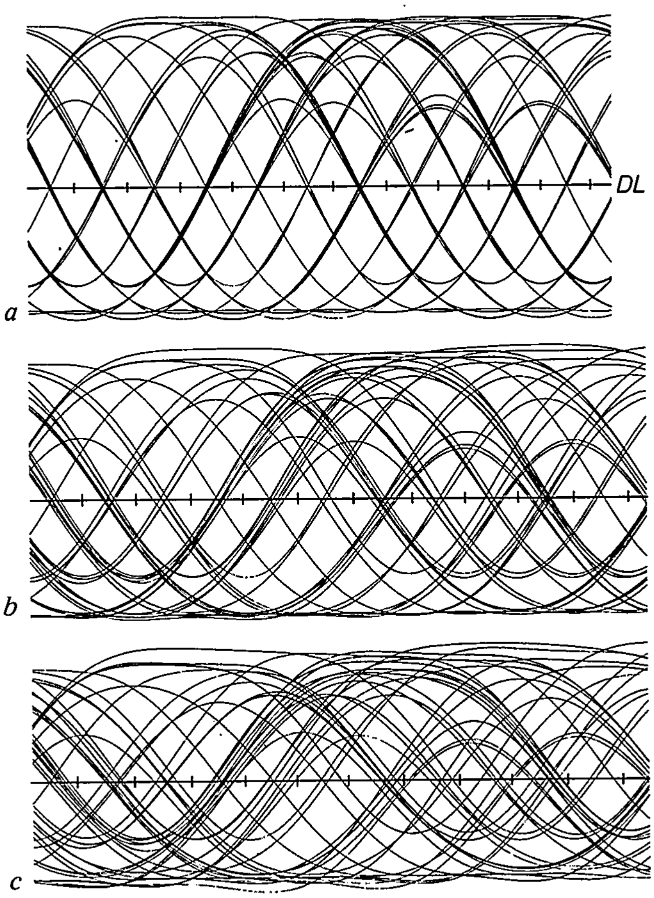

Fig. 3. Eye pattern. The figures give the read signal (at Q in fig. 1) on an oscilloscope synchronized with the bit clock. At the decision times (marked by dashes) it must be possible to determine whether the signal is above or below the decision level (*DL*). The curves have been calculated for *a*) an ideal optical system, *b*) a defocusing of 2 μm, *c*) a defocusing of 2 μm and a disc tilt of 1.2°. The curves give a good picture of experimental results.

to 'phase jitter' of the signal relative to the clock an eye becomes narrower, and noise reduces its height. The signals in fig. 3*a* were calculated for a perfect optical system. Fig. 3*b* shows the effect of defocusing by 2 μm and fig. 3*c* shows the effect of a radial tilt of 1.2° in addition to the defocusing. In fig. 3*b* a correct decision is still possible, but not in fig. 3*c*.

This example also gives some idea of the exacting requirements that the equipment has to meet. A more general picture can be obtained from *Table I*, which gives the manufacturing tolerances of a number of important parameters, both for the player and for the disc. The list is far from complete, of course.

With properly manufactured players and discs the channel quality can still be impaired by dirt and scratches forming on the discs during use. By its nature the system is fairly insensitive to these [1], and any errors they may introduce can nearly always be corrected or masked [2]. In the following we shall see that the modulation system also helps to reduce the sensitivity to imperfections.

Table I. Manufacturing tolerances.

Player	Objective-lens tilt ± 0.2°
	Tracking ± 0.1 μm
	Focusing ± 0.5 μm
	R.M.S. wavefront noise of read laser beam 0.05 λ (40 nm)
Disc	Thickness 1.2 ± 0.1 mm
	Flatness ± 0.6° (at the rim corresponding to a sag of 0.5 mm)
	Pit-edge positioning ± 50 nm
	Pit depth 120 ± 10 nm

Bit-stream modulation

The playing time of a disc is equal to the track length divided by the track velocity v. For a given disc size the playing time therefore increases if we decrease the track velocity in the system (the track velocity of the master disc and of the user disc). However, if we do this the channel becomes 'worse': the eye height decreases and the system becomes more sensitive to perturbations. There is therefore a lower limit to the track velocity if a minimum value has been established for the eye height because of the expected level of noise and perturbation. We shall now show that we can decrease this lower limit by an appropriate bit-stream modulation.

We first consider the situation without modulation. The incoming data bit stream is an arbitrary sequence of ones and zeros. We consider a group of 8 data bits in which the change of bit value is fastest (*fig. 4a*). Uncoded recording (1: pit; 0: land, or vice versa) then gives the pattern of fig. 4*b*. This results in the rounded-off signal of fig. 4*c* at Q in fig. 1; fig. 4*d* gives the eye pattern. The signal in fig. 4*c* represents the highest frequency (f_{m1}) for this mode of transmission, and we have $f_{m1} = \frac{1}{2} f_d$, where f_d is the data bit rate. The half eye height a_1 is equal to the amplitude A_1 of the highest-frequency signal.

[3] J. C. J. Finck, H. J. M. van der Laak and J. T. Schrama, A semiconductor laser for information read-out, Philips tech. Rev. 39, 37-47, 1980.

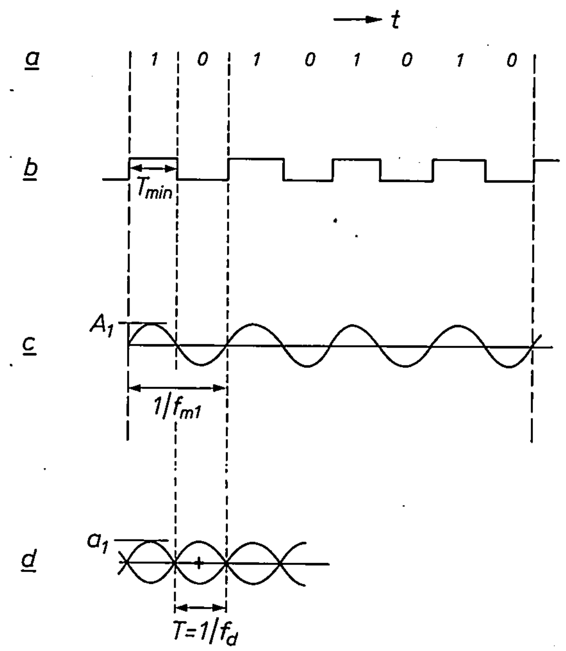

Fig. 4. Direct recording of the data bit stream on the disc. *a*) Data bit stream of the highest frequency that can occur. *b*) Direct translation of the bit stream into a pattern of pits. *c*) The corresponding output signal (at Q in fig. 1); its amplitude A_1 is found with the aid of fig. 5. *d*) The eye pattern that follows from (*c*). $T_{\min}$ minimum pit or land length; f_{m1} highest frequency; T data bit length; f_d data bit rate. We have $T_{\min} = T$; $f_{m1} = \frac{1}{2} f_d$.

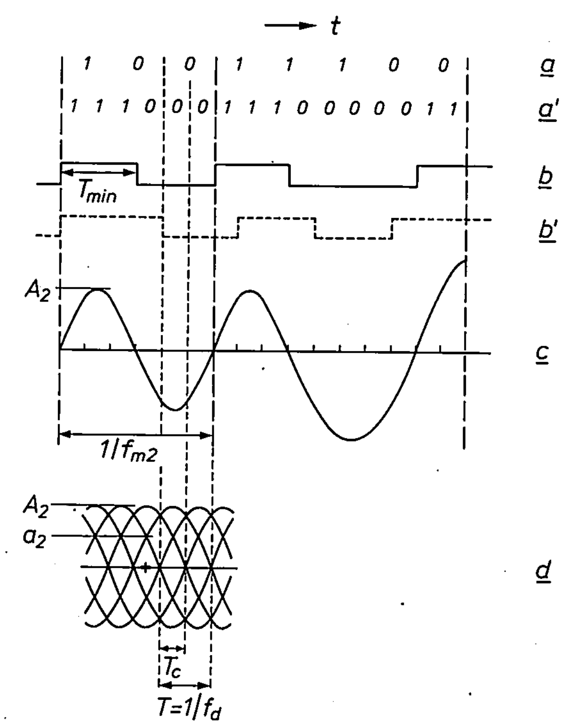

Fig. 6. Eight-to-sixteen modulation. Each group of 8 data bits (*a*) is translated with the aid of a dictionary into 16 channel bits (*a'*), in such a way that the run length is equal to at least three channel bits. *b*) Pattern of pits produced from the bit stream (*a'*). *b'*) pattern of pits obtained with a different input signal. *c*) The read signal corresponding to (*b*); its amplitude is again determined from fig. 5. *d*) The resultant eye pattern. The half eye height (a_2) here is only half the amplitude (A_2) of the approximately sinusoidal signal of maximum frequency (f_{m2}).

The relation between the eye height and the track velocity now follows indirectly from the 'amplitude-frequency characteristic' of the channel; see *fig. 5*. In this diagram A is the amplitude of the sinusoidal signal at Q in fig. 1 when a sinusoidal unit signal of frequency f is presented at P. With the aid of Fourier analysis and synthesis the output signal can be calculated from $A(f)$ for any input signal. The line in the diagram represents a channel with a perfect optical system. In the first part of this section we shall take this for granted. The true situation will always be less favourable. The 'cut-off frequency' is determined by the spot diameter and the track velocity v; in the ideal case $f_c = (2NA/\lambda)v$.

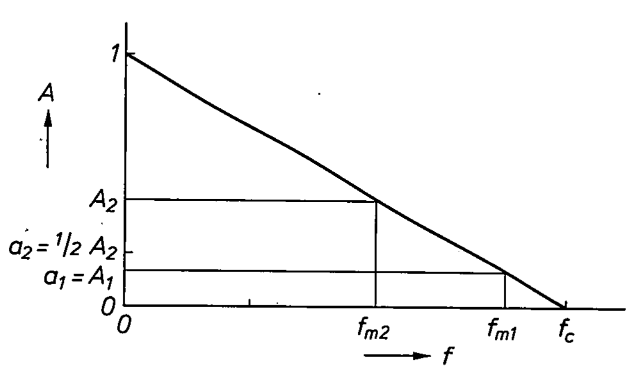

Fig. 5. Amplitude-frequency characteristic of the channel. The diagram gives the amplitude A of the sinusoidal signal at Q (fig. 1) when a sinusoidal unit signal is presented at P as a function of the frequency f. The transfer is 'cut off' at the frequency f_c, which is given by $f_c = (2NA/\lambda)v$. The line shown applies to an ideal optical system; in reality A is always somewhat lower; the cut-off frequency is then effectively lower. The 'maximum frequencies' f_{m1}, f_{m2}, the amplitudes A_1, A_2 and the 'half eye heights' a_1, a_2 relate to the 'direct' and 'modulated' writing of the data bits on the disc; see figs 4 and 6.

For a given track velocity we now obtain the half eye height a_1 in fig. 4 directly from fig. 5: it is equal to the amplitude A_1 at the frequency f_{m1}. If v, and hence f_c, is varied, the line in fig. 5 rotates about the point 1 on the A-axis. For a given minimum value of a_1, the figure indicates how far f_c can be decreased; this establishes the lower limit for v. In particular, if the minimum value for a_1 is very small, f_c can be decreased to a value slightly above f_{m1} $(= \frac{1}{2} f_d)$.

Fig. 6 gives the situation *with* modulation: an imaginary 8→16 modulation, which is very close to EFM, however. Each group of 8 incoming data bits (fig. 6*a*) is converted into 16 channel bits (fig. 6*a'*). This is done by using a 'dictionary' that assigns unambiguously but otherwise arbitrarily to each word of 8 bits a word of 16 bits, but in such a way that the resultant channel bit stream only produces pits and lands that are at least three channel bits long (fig. 6*b*). On the time scale the minimum pit and land lengths ('the minimum run length' T_{min}) have become $1\frac{1}{2}$ times as long as in fig. 4, but a simple calculation shows that about as much information can nevertheless be transmitted as in fig. 4 (256 combinations for 8 data bits), because there is a greater choice of pit-edge positions per unit length (see fig. 6*b* and *b'*); the 'channel bit length' T_c has decreased by a half.

With the modulation we have managed to reduce the highest frequency (f_{m2}) in the signal (see fig. 6*c*, *left*; $f_{m2} = \frac{1}{3} f_d = \frac{2}{3} f_{m1}$). Therefore f_c and v can be reduced by a factor of $1\frac{1}{2}$ for the case in which a very small eye height is tolerable (see fig. 5); this represents an increase of 50% in playing time.

The modulation also has its disadvantages. In the first place the half eye height (a_2) in this case is only *half* of the amplitude (A_2) of the signal at the highest frequency (see fig. 6*d*). This has consequences if the minimum eye height is not very small. For example, the modulation becomes completely unusable if the half eye height in fig. 5 has to remain larger than $\frac{1}{2}$ ($a_2 > \frac{1}{2}$ implies $A_2 > 1$); uncoded recording is then still possible ($A_1 = a_1$). In the second place, the tolerance for time errors and for the positioning of pit edges, together with the eye width (T_c), has decreased by a half. In designing a system, the various factors have to be carefully weighed against one another.

To show qualitatively how a choice can be made, we have plotted the half eye height in *fig. 7* as a function of the 'linear information density' σ (the number of incoming data bits per unit length of the track; $\sigma = f_d/v$) for three systems: '8→8 modulation' (i.e. uncoded recording), 8→16 modulation, and a system that also has about the same information capacity (256 combinations for 8 data bits) in which, however, the minimum run length has been increased still further, again at the expense of eye width of course ('8→24 modulation', $T_{min} = 2T$, $T_c = \frac{1}{3} T$). The figure is a direct consequence of the reasoning above, with the assumption that the cut-off frequency is 20% lower than the ideal value $(2NA/\lambda)v$, as a first rough adjustment to what we find in practice for the function $A(f)$.

In qualitative terms, the 8→16 system has been chosen because the nature of the noise and perturbations is such that the eye can be smaller than at A in fig. 7, but becomes too small at C. An improvement is therefore possible with 8→16 modulation, but not with 8→24 modulation.

For our Compact Disc system we have $\sigma = 1.55$ data bits/μm ($f_d = 1.94$ Mb/s, $v = 1.25$ m/s [1]); the operating point would therefore be at P in fig. 7. The model used is however rather crude and in better models A, B and C lie more to the left, so that P approaches C. But 8→16 modulation is still preferable to 8→24 modulation, even close to C, since the eye width is $1\frac{1}{2}$ times as large as for 8→24 modulation.

EFM is a refinement of 8→16 modulation. It has been chosen on the basis of more detailed models and many experiments. At the eye height used, it gives a gain of 25% in information density, compared with uncoded recording.

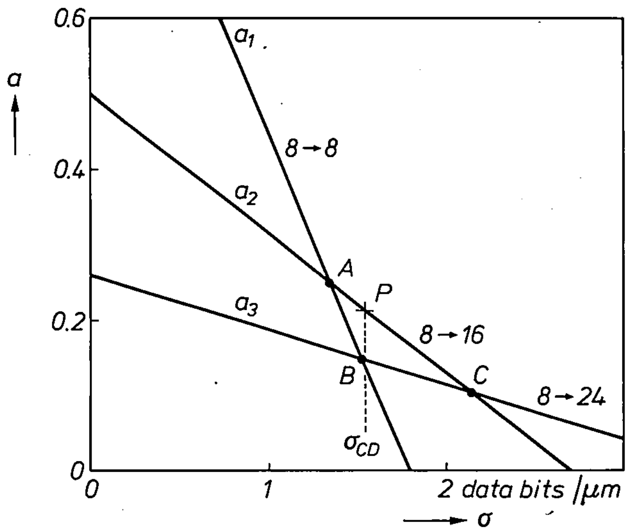

Fig. 7. Half eye height a as a function of the linear information density σ, for 8→8, 8→16 and 8→24 modulation. These systems are characterized by the following values for the channel bit length T_c and the minimum run length T_{min}:
8→8: $T_c = T$, $T_{min} = T$ (fig. 4),
8→16: $T_c = \frac{1}{2} T$, $T_{min} = \frac{3}{2} T$ (fig. 6),
8→24: $T_c = \frac{1}{3} T$, $T_{min} = 2T$,
where T is the data bit length. The straight lines give the relations that follow from fig. 5:
$a_1 = c_1(1 - f_{m1}/f_c) \rightarrow a_1 = 1 - \sigma/1.8$,
$a_2 = c_2(1 - f_{m2}/f_c) \rightarrow a_2 = 0.5(1 - \sigma/2.7)$,
$a_3 = c_3(1 - f_{m3}/f_c) \rightarrow a_3 = 0.26(1 - \sigma/3.6)$,
where σ is the numerical value of the linear information density, expressed in data bits per μm. The c's are the ratios of the half eye height to the amplitude, and the f_m's the maximum frequencies for the three systems ($c_1 = 1$, $c_2 = \sin 30° = 0.5$, $c_3 = \sin 15° = 0.26$, $f_{m1} = \frac{1}{2} f_d$, $f_{m2} = \frac{1}{3} f_d$, $f_{m3} = \frac{1}{4} f_d$; f_d is the data bit rate). The second set of equations follows from the first set by substituting $0.8 \times (2NA/\lambda)v$ for f_c, with $NA = 0.45$, $\lambda = 0.8$ μm, $v = f_d/\sigma$. The factor 0.8 is introduced as a rough first-order correction to the 'ideal' amplitude characteristic.

Further requirements for the modulation system

In developing the modulation system further we still had two more requirements to take into account.

In the first place it must be possible to regenerate the *bit clock* in the player from the read-out signal (the signal at Q in fig. 1). To permit this the number of pit edges per second must be sufficiently large, and in particular the 'maximum run length' T_{max} must be as small as possible.

The second requirement relates to the 'low-frequency content' of the read signal. This has to be as small as possible. There are two reasons for this. In the first place, the servosystems for track following and focusing [1] are controlled by low-frequency signals, so that low-frequency components of the information signal could interfere with the servosystems. The second reason is illustrated in *fig. 8*, in which the read signal is shown for a clean disc (*a*) and for a disc that has been soiled, e.g. by fingermarks (*b*). This causes the amplitude and average level of the signal to fall. The fall in level causes a completely

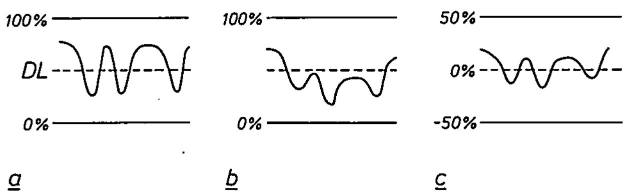

Fig. 8. The read-out signal for six pit edges on the disc, *a*) for a clean disc, *b*) for a soiled disc, *c*) for a soiled disc after the low frequencies have been filtered out. *DL* decision level. Because of the soiling, both the amplitude and the signal level decrease; the decision errors that this would cause are eliminated by the filter.

wrong read-out if the signal falls below the decision level. Errors of this type are avoided by eliminating the low-frequency components with a filter (*c*), but the use of such a filter is only permissible provided the information signal itself contains no low-frequency components. In the Compact Disc system the frequency range from 20 kHz to 1.5 MHz is used for information transmission; the servosystems operate on signals in the range 0-20 kHz.

The EFM modulation system

Fig. 9 gives a schematic general picture of the bit streams in the encoding system. The information is divided into 'frames'. One frame contains 6 sampling periods, each of 32 audio bits (16 bits for each of the two audio channels). These are divided into symbols of 8 bits. The bit stream B_1 thus contains 24 symbols per frame. In B_2 eight parity symbols have been added and one C&D symbol, resulting in 33 'data symbols'. The modulator translates each symbol into a new symbol of 14 bits. Added to these are three 'merging bits', for reasons that will appear shortly. After the addition of a synchronization symbol of 27 bits to the frame, the bit stream B_i is obtained. B_i therefore contains $33 \times 17 + 27 = 588$ channel bits per frame. Finally, B_i is converted into a control signal for the write laser. It should be noted that in B_i '1' or '0' does not mean 'pit' or 'land', as we assumed for simplicity in fig. 6, but a '1' indicates a pit edge. The information is thus completely recorded by the positions of the pit edges; it therefore makes no difference to the decoding system if 'pit' and 'land' are interchanged on the disc.

Opting for the translation of series of 8 bits following the division into symbols in the parity coding has the effect of avoiding error propagation. This is because in the error-correction system an entire symbol is always either 'wrong' or 'not wrong'. One channel-bit error that occurs in the transmission spoils an entire symbol, but — because of the correspondence between modulation symbols and data symbols — never more than one symbol. If a different modulation system is used, in which the data bits are not translated in groups of 8, but in groups of 6 or 10, say, then the bit stream B_2 is in fact first divided up into 6 or 10 bit 'modulation symbols'. Although one channel-bit error then spoils only one modulation symbol, it usually spoils two of the original 8 bit symbols.

In EFM the data bits are translated 8 at a time into 14 channel bits, with a T_{min} of 3 and a T_{max} of 11 channel bits (this means at least 2 and at the most 10 successive zeros in B_i). This choice came about more or less as follows. We have already seen that the choice of about $1\frac{1}{2}$ data bits for T_{min}, with about 16 channel bits on 8 data bits, is about the optimum for the Compact Disc system [4]. A simple calculation shows that at least 14 channel bits are necessary for the reproduction of all the 256 possible symbols of 8 data bits under the conditions $T_{min} = 3$, $T_{max} = 11$ channel bits. The choice of T_{max} was dictated by the fact that a larger choice does not make things very much easier, whereas a smaller choice does create far more difficulties.

With 14 channel bits it is possible to make up 267 symbols that satisfy the run-length conditions. Since we only require 256, we omitted 10 that would have introduced difficulties with the 'merging' of symbols under these conditions, and one other chosen at random. The dictionary 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 circuit, i.e. a circuit that contains the minimum of logic gates.

The merging bits are primarily intended to ensure that the run-length conditions continue to be satisfied when the symbols are 'merged'. If the run length is in danger of becoming too short we choose '0's for the merging bits; if it is too long we choose a '1' for one of them. If we do this we still retain a large measure of freedom in the choice of the merging bits, and we use this freedom to minimize the low-frequency content of the signal. In itself, *two* merging bits would be sufficient for continuing to satisfy the run-length conditions. A third is necessary, however, to give sufficient freedom for effective suppression of the low-frequency content, even though it means a loss of 6% of the information density on the disc. The merging bits contain no audio information, and they are removed from the bit stream in the demodulator.

Fig. 10 illustrates, finally, how the merging bits are determined. Our measure of the low-frequency content is the 'digital sum value' (DSV); this is the difference between the totals of pit and land lengths accumulated from the beginning of the disc. At the top are shown two data symbols of B_2 and their translation from the dictionary into channel symbols (B_3). From the T_{min} rule the first of the merging bits in this case must be a zero; this position is marked 'X'. In the two following positions the choice is free; these are marked 'M'. The three possible choices XMM = 000, 010 and 001 would give rise to the patterns of pits as illustrated, and to the indicated waveform of the

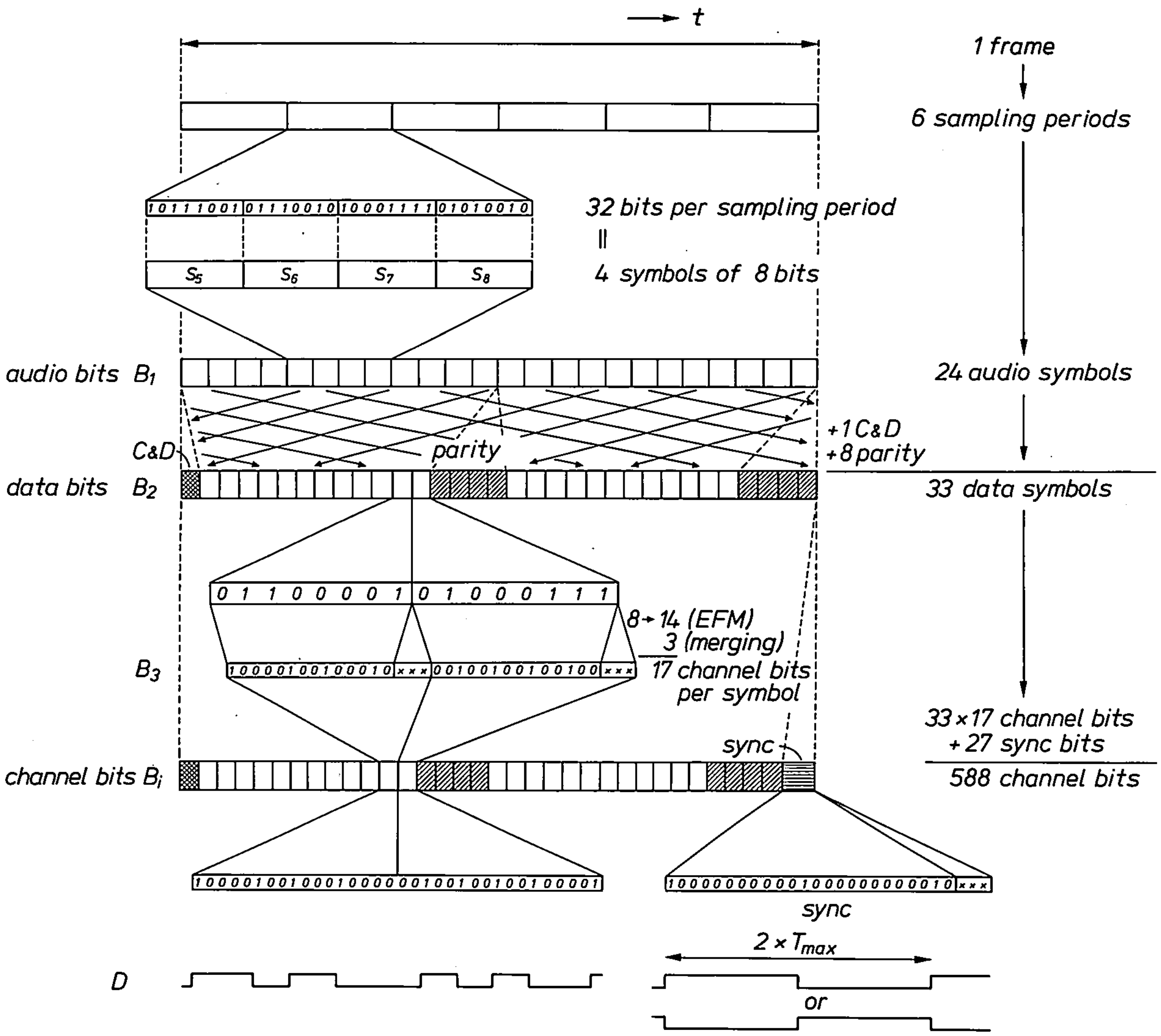

Fig. 9. Bit streams in the encoding system (fig. 2). The information is divided into frames; the figure gives one frame of the successive bit streams. There are six sampling periods for one frame, each sampling period giving 32 bits (16 for each of the two audio channels). These 32 bits are divided to make four symbols in the 'audio bit stream' B_1. In the 'data bit stream' B_2 eight parity and one C&D symbols have been added to the 24 audio symbols. To scatter possible errors, the symbols of different frames in B_1 are interleaved, so that the audio signals in one frame of B_2 originate from different frames in B_1. The modulation translates the eight data bits of a symbol of B_2 into fourteen channel bits, to which three 'merging bits' are added (B_3). The frames are marked with a synchronization signal of the form illustrated (bottom right); the final result is the 'channel bit stream' (B_i) used for writing on the master disc, in such a way that each '1' indicates a pit edge (D).

[4] A more detailed discussion is given in K. A. Immink, Modulation systems for digital audio discs with optical readout, Proc. IEEE Int. Conf. on Acoustics, speech and signal processing, Atlanta 1981, pp. 587-589.

DSV, on the assumption that the DSV was equal to 0 at the beginning. The system now opts for the merging combination that makes the DSV at the end of the second symbol as small as possible, i.e. 000 in this case. If the initial value had been −3, the merging combination 001 would have been chosen.

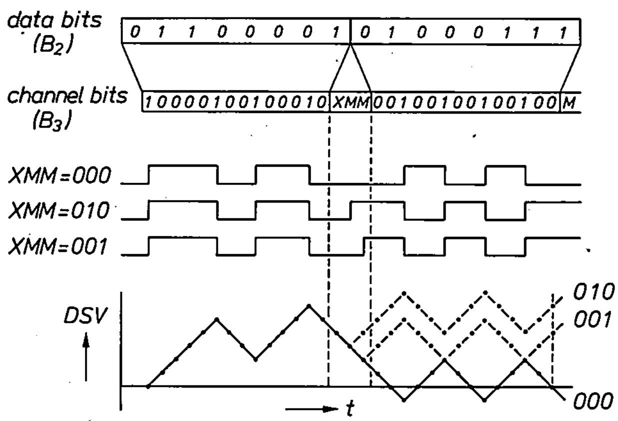

Fig. 10. Strategy for minimizing the digital sum value (DSV). After translation of the data bits into channel bits, the symbols are merged together by means of three extra bits in such a way that the run-length conditions continue to be satisfied and the DSV remains as small as possible. The first run-length rule (at least two zeros one after the other) requires a zero at the first position in the case illustrated here, while the choice remains free for the second and third positions. In this case there are thus three merging alternatives: 000, 010 and 001. These alternatives give the patterns of pits shown in the diagram and the illustrated DSV waveform. The system chooses the alternative that gives the lowest value of DSV at the end of the next symbol. The system looks 'one symbol ahead'; strategies for looking further ahead are also possible in principle.

When this strategy is applied, the noise in the servo-band frequencies (< 20 kHz) is suppressed by about 10 dB. In principle better results can be obtained, within the agreed standard for the Compact Disc system, by looking more than one symbol ahead, since minimization of the DSV in the short term does not always contribute to longer-term minimization. This is not yet done in the present equipment.

Summary. The Compact Disc system can be considered as a transmission system that brings sound from the studio into the living room. The sound encoded into data bits and modulated into channel bits is sent along the 'transmission channel' consisting of write laser — master disc — user disc — optical pick-up. The maximum information density on the disc is determined by the diameter d of the laser light spot on the disc and the 'number of data bits per light spot'. The effect of making d smaller is to greatly reduce the manufacturing tolerances for the player and the disc. The compromise adopted is $d \approx 1$ µm, giving very small tolerances for objective and disc tilt, disc thickness and defocusing. The basic idea of the modulation is that, while maintaining the minimum length for 'pit' and 'land' (the 'minimum run length') required for satisfactory transmission, the information density can be increased by increasing the number of possible positions per unit length for pit edges (the bit density). Because of clock regeneration there is also a maximum run length, and the low-frequency content of the transmission channel must be kept as low as possible. With the EFM modulation system used each 'symbol' of eight data bits is converted into 14 channel bits with a minimum run length of 3 and a maximum run length of 11 bits, plus three merging bits, chosen such that, when the symbols are merged together, the run-length conditions continue to be satisfied and the low-frequency content is kept to the minimum.

This prototype player, which will be put on the market later, will display 'information for the listener' such as title, composer, 'track number' and playing time of the piece of music. The different sections of the music on the disc can also be played in the order selected by the user — the numbers on the far right.