PHILIPS TECHNICAL REVIEW

VOLUME 40, 1982, No. 6

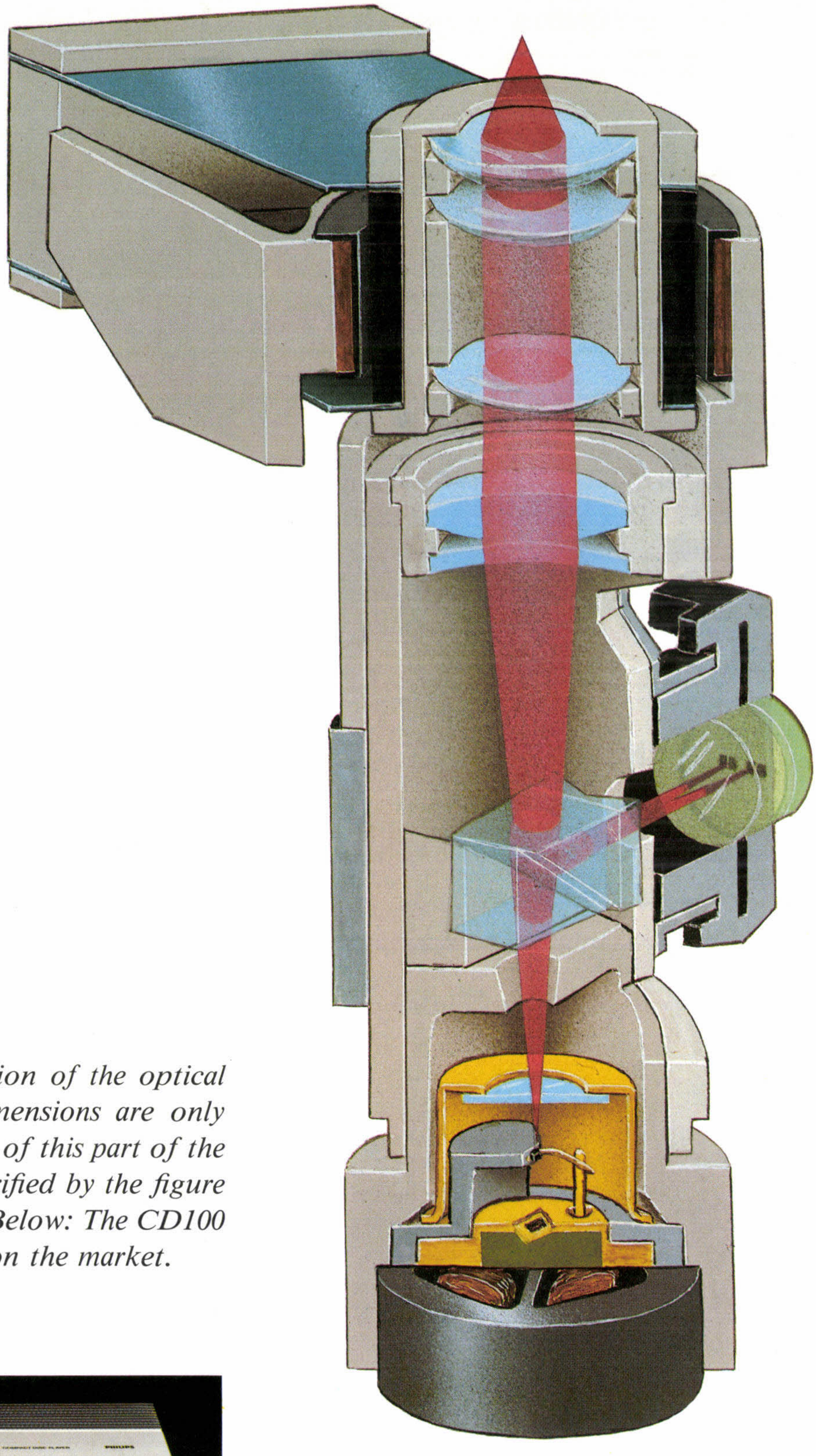
COMPACT
disc
DIGITAL AUDIO

In 1877 Edison's phonograph played the nursery rhyme 'Mary had a little lamb', after he had recorded it on the wax cylinder in his own voice: the human voice had been reproduced for the first time in history. Then came Berliner's wax disc, followed by the 78-turns-per-minute shellac disc, and eventually the modern long-play record. Now when we enjoy the music from our 'LPs' at home it is almost perfectly reproduced by our hi-fi equipment. However, the record player itself is a weak link in the chain, since damage to the vulnerable disc often introduces an unwanted accompaniment of undesirable sounds to the music. This cannot happen with the Compact Disc. It is scanned optically, so playing it cannot produce any damage, and dust and fingermarks have far less effect — because errors can in fact be corrected. Another way in which the Compact Disc differs from the conventional long-play record is that the sound is recorded on the disc in digital form. The digital processing of the audio signals will perhaps be a more far-reaching development in the history of record-playing equipment than the change from acoustic to electrical reproduction was in its time. As we shall see, digital processing brings many advantages. To make the best use of it, it is necessary to build up a complete system that extends from the record-manufacturer's equipment to the record player at home. It is clear that so extensive a system only has a chance of success if records and playing equipment from different manufacturers are all absolutely compatible.

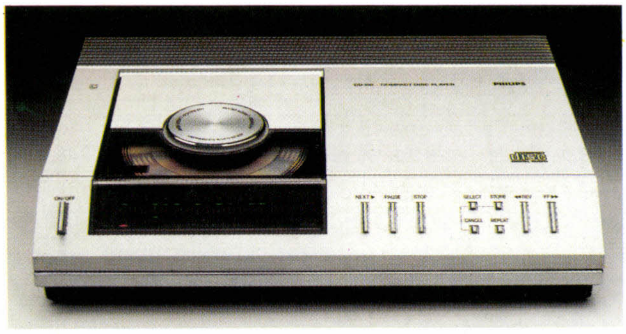
The study of the possibility of recording audio signals optically on a disc was started in 1974 at Philips Research Laboratories, in close cooperation with the Philips Audio Division. It soon became clear that a different method would have to be used from that in the LaserVision system [*]*: digital signal coding, instead of analog modulation methods. More and more people from the Research Laboratories and the Audio Division gradually became involved in the project. After an agreement 'in principle' had been reached with the Sony company in 1979, extensive technical discussions were started, mainly about the signal processing. Eventually a system standard was produced, including contributions from both companies. Then licensing agreements were made with a number of other manufacturers of audio equipment and gramophone records.*

This issue contains four articles written by staff from Philips Research Laboratories and the Compact Disc Development Laboratory of the Audio Division. They give an account of various aspects of the revolutionary Compact Disc system: the complete system, the modulation of the digital signal, the method of error correction and the conversion of the digital signal into the analog signal.

[*] Formerly called the VLP system.



Right: An artist's impression of the optical pick-up, whose actual dimensions are only 45×12 mm. The operation of this part of the playback equipment is clarified by the figure and caption on page 153. Below: The CD100 player, the first to be put on the market.



The Compact Disc Digital Audio system

M. G. Carasso, J. B. H. Peek and J. P. Sinjou

Introduction

During the many years of its development the gramophone has reached a certain maturity. The availability of long-play records of high quality has made it possible to achieve very much better sound reproduction in our homes than could be obtained with the machine that first reproduced the sound of the human voice in 1877. A serious drawback of these records is that they have to be very carefully handled if their quality is to be preserved. The mechanical tracking of the grooves in the record causes wear, and damage due to operating errors cannot always be avoided. Because of the analog recording and reproduction of the sound signal the signal-to-noise ratio may sometimes be poor (< 60 dB), and the separation between the stereo channels (< 30 dB) leaves something to be desired.

For these and other problems the Compact Disc system offers a solution. The digital processing of the signal has resulted in signal-to-noise ratios and a channel separation that are both better than 90 dB. Since the signal information on the disc is protected by a 1.2 mm transparent layer, dust and surface damage do not lie in the focal plane of the laser beam that scans the disc, and therefore have relatively little effect. Optical scanning as compared with mechanical tracking means that the disc is not susceptible to damage and wear. The digital signal processing makes it possible to correct the great majority of any errors that may nevertheless occur. This can be done because error-correction bits are added to the information present on the disc. If correction is not possible because there are too many defects, the errors can still be detected and 'masked' by means of a special procedure. When a Compact Disc is played there is virtually no chance of hearing the 'tick' so familiar from conventional records.

With its high information density and a playing time of an hour, the outside diameter of the disc is only 120 mm. Because the disc is so compact, the dimensions of the player can also be small. The way in which the digital information is derived from the analog music signal gives a frequency characteristic that is flat from 20 to 20 000 Hz. With this system the well-known wow and flutter of conventional players are a thing of the past.

Another special feature is that 'control and display' information is recorded, as 'C&D' bits. This includes first of all 'information for the listener', such as playing time, composer and title of the piece of music. The number of a piece of music on the disc is included as well. The C&D bits also contain information that indicates whether the audio signal has been recorded with pre-emphasis and should be reproduced with de-emphasis [1]. In the Compact Disc system a pre-emphasis characteristic has been adopted as standard with time constants of 15 and 50 μs. In some of the versions of the player the 'information for the listener' can be presented on a display and the different sections of the music on the disc can be played in the order selected by the user.

In the first article of a series of four on the Compact Disc system we shall deal with the complete system, without going into detail. We shall consider the disc, the processing of the audio signal, reading out the signal from the disc and the reconstitution of the audio signal. The articles that follow will examine the system aspects and modulation, error correction and the digital-to-analog conversion.

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[1] See F. W. de Vrijer, Modulation, Philips tech. Rev. 36, 305-362, 1976, in particular pages 323 and 324.

The disc

In the LaserVision system [2], which records video information, the signal is recorded on the disc in the form of a spiral track that consists of a succession of pits. The intervals between the pits are known as 'lands'. The information is present in the track in analog form. Each transition from land to pit and vice versa marks a zero crossing of the modulated video signal. On the Compact Disc the signal is recorded in a similar manner, but the information is present in the track in digital form. Each pit and each land represents a series of bits called channel bits. After each land/pit or pit/land transition there is a '1', and all the channel bits in between are '0'; see *fig. 1*.

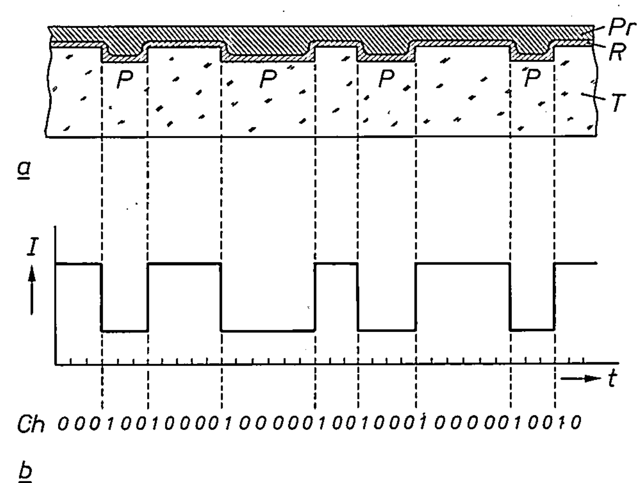


Fig. 1. *a*) Cross-section through a Compact Disc in the direction of the spiral track. *T* transparent substrate material, *R* reflecting layer, *Pr* protective layer. *P* the pits that form the track. *b*) *I* the intensity of the signal read by the optical pick-up (see fig. 2), plotted as a function of time. The signal, shown in the form of rectangular pulses, is in reality rounded and has sloping sides [3]. The digital signal derived from this waveform is indicated as a series of channel bits *Ch*.

The density of the information on the Compact Disc is very high: the smallest unit of audio information (the audio bit) covers an area of 1 μm² on the disc, and the diameter of the scanning light-spot is only 1 μm. The pitch of the track is 1.6 μm, the width 0.6 μm and the depth 0.12 μm. The minimum length of a pit or the land between two pits is 0.9 μm, the maximum length is 3.3 μm. The side of the transparent carrier material *T* in which the pits *P* are impressed — the upper side during playback if the spindle is vertical — is covered with a reflecting layer *R* and a protective layer *Pr*. The track is optically scanned from below the disc at a constant velocity of 1.25 m/s. The speed of rotation of the disc therefore varies, from about 8 rev/s to about 3.5 rev/s.

Processing of the audio signal

For converting the analog signal from the microphone into a digital signal, pulse-code modulation (PCM) is used. In this system the signal is periodically sampled and each sample is translated into a binary number. From Nyquist's sampling theorem the frequency of sampling should be at least twice as high as the highest frequency to be accounted for in the analog signal. The number of bits per sample determines the signal-to-noise ratio in the subsequent reproduction.

In the Compact Disc system the analog signal is sampled at a rate of 44.1 kHz, which is sufficient for reproduction of the maximum frequency of 20 000 Hz. The signal is quantized by the method of uniform quantization; the sampled amplitude is divided into equal parts. The number of bits per sample (these are called audio bits) is 32, i.e. 16 for the left and 16 for the right audio channel. This corresponds to a signal-to-noise ratio of more than 90 dB. The net bit rate is thus $44.1 \times 10^3 \times 32 = 1.41 \times 10^6$ audio bits/s. The audio bits are grouped into 'frames', each containing six of the original samples.

Successive blocks of audio bits have blocks of parity bits added to them in accordance with a coding system called CIRC (Cross-Interleaved Reed-Solomon Code) [4]. This makes it possible to correct errors during the reproduction of the signal. The ratio of the number of bits before and after this operation is 3:4. Each frame then has C&D (Control and Display) bits, as mentioned earlier, added to it; one of the functions of the C&D bits is providing the 'information for the listener'. After the operation the bits are called data bits.

Next the bit stream is modulated, that is to say the data bits are translated into channel bits, which are suitable for storage on the disc; see fig. 1*b*. The EFM code (Eight-to-Fourteen Modulation) is used for this: in EFM code blocks of eight bits are translated into blocks of fourteen bits [5]. The blocks of fourteen bits are linked by three 'merging bits'. The ratio of the number of bits before and after modulation is thus 8:17.

For the synchronization of the bit stream an identical synchronization pattern consisting of 27 channel bits is added to each frame. The total bit rate after all these manipulations is 4.32×10^6 channel bits/s.

[2] See Philips tech. Rev. 33, 187-193, 1973.
[3] See fig. 3 of the article by J. P. J. Heemskerk and K. A. Schouhamer Immink, on p. 159 of this issue.
[4] See H. Hoeve, J. Timmermans and L. B. Vries, Error correction and concealment in the Compact Disc system, this issue, p. 166.
[5] See J. P. J. Heemskerk and K. A. Schouhamer Immink, Compact Disc: system aspects and modulation, this issue, p. 157.
[6] J. C. J. Finck, H. J. M. van der Laak and J. T. Schrama, Philips tech. Rev. 39, 37, 1980.

Table I. Names of the successive signals, the associated bit rates and operations during the processing of the audio signal.

Name	Bit rate in 10^6 bits/s	Operations
Audio signal		PCM (44.1 kHz)
Audio bit stream	1.41	CIRC (+ parity bits) Addition of C&D bits
Data bit stream	1.94	EFM Addition of merging bits Addition of synchronization patterns
Channel bit stream	4.32	

disc (called the 'master'). A pattern of pits is produced on this disc by means of a photographic developing process. After the surface has been coated with a thin silver layer, an electroplating process is applied to produce a nickel impression, called the 'metal father'. From this 'father disc' impressions called 'mother discs' are produced in a similar manner. The impressions of the mother discs, called 'sons' or 'stampers', are used as tools with which the pits P are impressed into the thermoplastic transparent carrier material T of the disc; see fig. 1.

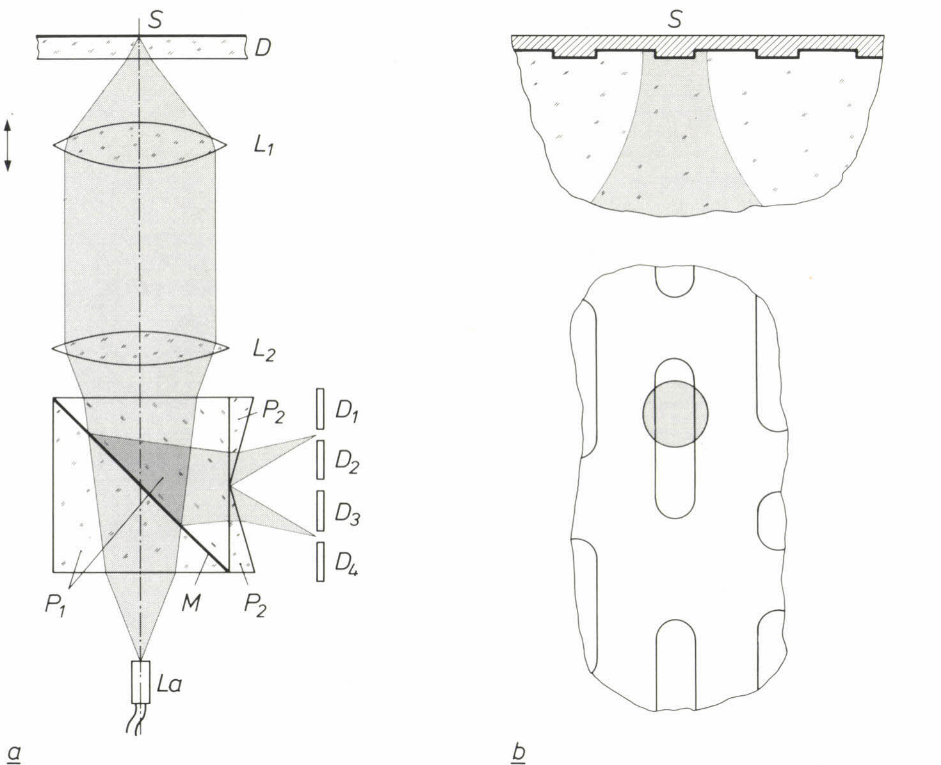


Fig. 2. *a*) Diagram of the optical pick-up. D radial section through the disc. S laser spot, the image on the disc of the light-emitting part of the semiconductor laser La. L_1 objective lens, adjustable for focusing. L_2 lens for making the divergent laser beam parallel. M half-silvered mirror formed by a film evaporated on the dividing surface of the prism combination P_1. P_2 beam-splitter prisms. D_1 to D_4 photodiodes whose output currents can be combined in various ways to provide the output signal from the pick-up and also the tracking-error signal and the focusing-error signal. (In practice the prisms P_2 and the photodiodes D_1 to D_4 are rotated by 90° and the reflection at the mirror M does not take place in a radial plane but in a tangential plane.) *b*) A magnified view of the light spot S and its immediate surroundings, with a plan view. It can clearly be seen that the diameter of the spot (about 1 μm) is larger than the width of the pit (0.6 μm).

Table I gives a survey of the successive operations with the associated bit rates, with their names. From the magnitude of the channel bit rate and the scanning speed of 1.25 m/s it follows that the length of a channel bit on the disc is approximately 0.3 μm.

The signal produced in this way is used by the disc manufacturer to switch on and off the laser beam that illuminates the light-sensitive layer on a rotating glass

Read-out from the disc

As we have seen, the disc is optically scanned in the player. This is done by the AlGaAs semiconductor laser described in an earlier article in this journal [6]. *Fig. 2* shows the optical part of the 'pick-up'. The light from the laser La (wavelength 800 nm) is focused through the lenses L_2 and L_1 on to the reflecting layer of the disc. The diameter of the light spot S is about

1 μm. When the spot falls on an interval between two pits, the light is almost totally reflected and reaches the four photodiodes D_1-D_4 via the half-silvered mirror M. When the spot lands on a pit — the depth of a pit is about $\frac{1}{4}$ of the wavelength in the transparent substrate material — interference causes less light to be reflected and an appreciably smaller amount reaches the photodiodes. When the output signals from the four photodiodes are added together the result is a fairly rough approximation [3] to the rectangular pulse pattern present on the disc in the form of pits and intervals.

The optical pick-up shown in fig. 2 is very small (about 45 × 12 mm) and is mounted in a pivoting arm that enables the pick-up to describe a radial arc across the disc, so that it can scan the complete spiral track. Around the pivotal point of the arm is mounted a 'linear' motor that consists of a combination of a coil and a permanent magnet. When the coil is energized the pick-up can be directed to any required part of the track, the locational information being provided by the C&D bits added to each frame on the disc. The pick-up is thus able to find independently any particular passage of music indicated by the listener. When it has been found, the pick-up must then follow the track accurately — to within ± 0.1 μm — without being affected by the next or previous track. Since the track on the disc may have some slight eccentricity, and since also the suspension of the turntable is not perfect, the track may have a maximum side-to-side swing of 300 μm. A tracking servosystem is therefore necessary to ensure that the deviation between pick-up and track is smaller than the permitted value of ± 0.1 μm and in addition to absorb the consequences of small vibrations of the player.

The tracking-error signal is delivered by the four photodiodes D_1 to D_4. When the spot S, seen in the radial direction, is situated in the centre of the track, a symmetrical beam is reflected. If the spot lies slightly to one side of the track, however, interference effects cause asymmetry in the reflected beam. This asymmetry is detected by the prisms P_2, which split the beam into two components. Beyond the prisms one component has a higher mean intensity than the other. The signal obtained by coupling the photodiodes as $(D_1 + D_2) - (D_3 + D_4)$ can therefore be used as a tracking-error signal.

As a result of ageing or soiling of the optical system, the reflected beam may acquire a slowly increasing, more or less constant asymmetry. Owing to a d.c. component in the tracking-error signal, the spot will then always be slightly off-centre of the track. To compensate for this effect a second tracking-error signal is generated. The coil that controls the pick-up arm is therefore supplied with an alternating voltage at 600 Hz, with an amplitude that corresponds to a radial displacement of the spot by ± 0.05 μm. The output sum signal from the four photodiodes — which is at a maximum when the spot is in the centre of the track — is thus modulated by an alternating voltage of 600 Hz. The amplitude of this 600 Hz signal increases as the spot moves off-centre. In addition the sign of the 600 Hz error signal changes if the spot moves to the other side of the track. This second tracking-error signal is therefore used to correct the error signal mentioned earlier with a direct voltage. The output sum signal from the photodiodes, which is processed in the player to become the audio signal, is thus returned to its maximum value.

The depth of focus of the optical pick-up at the position of S (see fig. 2) is about 4 μm. The axial deviation of the disc, owing to various mechanical effects, can have a maximum of 1 mm. It is evident that a servosystem is also necessary to give correct focusing of the pick-up on the reflecting layer. The objective lens L_1 can therefore be displaced in the direction of its optical axis by a combination of a coil and a permanent magnet, in the same way as in a loudspeaker. The focusing-error signal is also provided by the row of photodiodes D_1 to D_4. If the spot is sharply focused on the disc, two sharp images are precisely located between D_1 and D_2 and between D_3 and D_4. If the spot is not sharply focused on the disc, the two images on the photodiodes are not sharp either, and have also moved closer together or further apart. The signal obtained by connecting the photodiodes as $(D_1 + D_4) - (D_2 + D_3)$ can therefore be used for controlling the focusing servosystem. The deviation in focusing then remains limited to ± 1 μm.

Reconstitution of the audio signal

The signal read from the disc by the optical pick-up has to be reconstituted to form the analog audio signal.

Fig. 3 shows the block diagram of the signal processing in the player. In *DEMOD* the demodulation follows the same rules that were applied to the EFM modulation, but now in the opposite sense. The information is then temporarily stored in a buffer memory and then reaches the error-detection and correction circuit *ERCO*. The parity bits can be used here to correct errors, or just to detect errors if correction is found to be impossible [4]. These errors may originate from defects in the manufacturing process, damage during use, or fingermarks or dust on the disc. Since the information with the CIRC code is 'interleaved' in time, errors that occur at the input of *ERCO* in one frame are spread over a large number of frames during decoding in *ERCO*. This increases the probability that the maximum number of correctable errors per frame will not be exceeded. A flaw such as a scratch can often produce a train of errors, called an error burst. The error-correction code used in *ERCO* can correct a burst of up to 4000 data bits, largely because the errors are spread out in this way.

ERCO are synchronized by a clock generator *C* controlled by a quartz crystal.

Fig. 3 also illustrates the control of the disc speed n_D. The bit stream leaves the buffer memory at a rate synchronized by the clock generator. The bit stream enters the buffer memory, however, at a rate that depends on the speed of revolution of the disc. The extent to which n_D and the sampling rate are matched

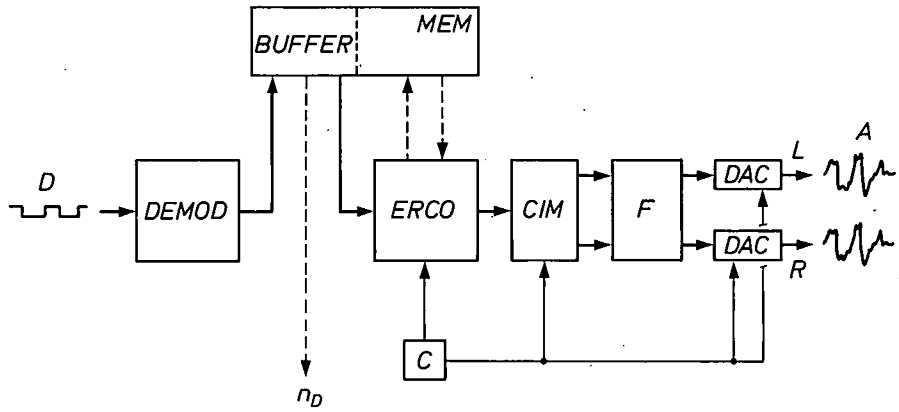


Fig. 3. Block diagram of the signal processing in the player. *D* input signal read by the optical pick-up; see fig. 2. *A* the two output analog audio signals from the left (*L*) and the right (*R*) audio channels. *DEMOD* demodulation circuit. *ERCO* error-correction circuit. *BUFFER* buffer memory, forming part of the main memory *MEM* associated with *ERCO*. *CIM* (Concealment: Interpolation and Muting) circuit in which errors that are only detected since they cannot be corrected are masked or 'concealed'. *F* filters for interpolation. *DAC* digital-to-analog conversion circuits. Each of the blocks mentioned here are fabricated in VLSI technology. *C* clock generator controlled by a quartz crystal. The degree to which the buffer memory capacity is filled serves as a criterion in controlling the speed of the disc.

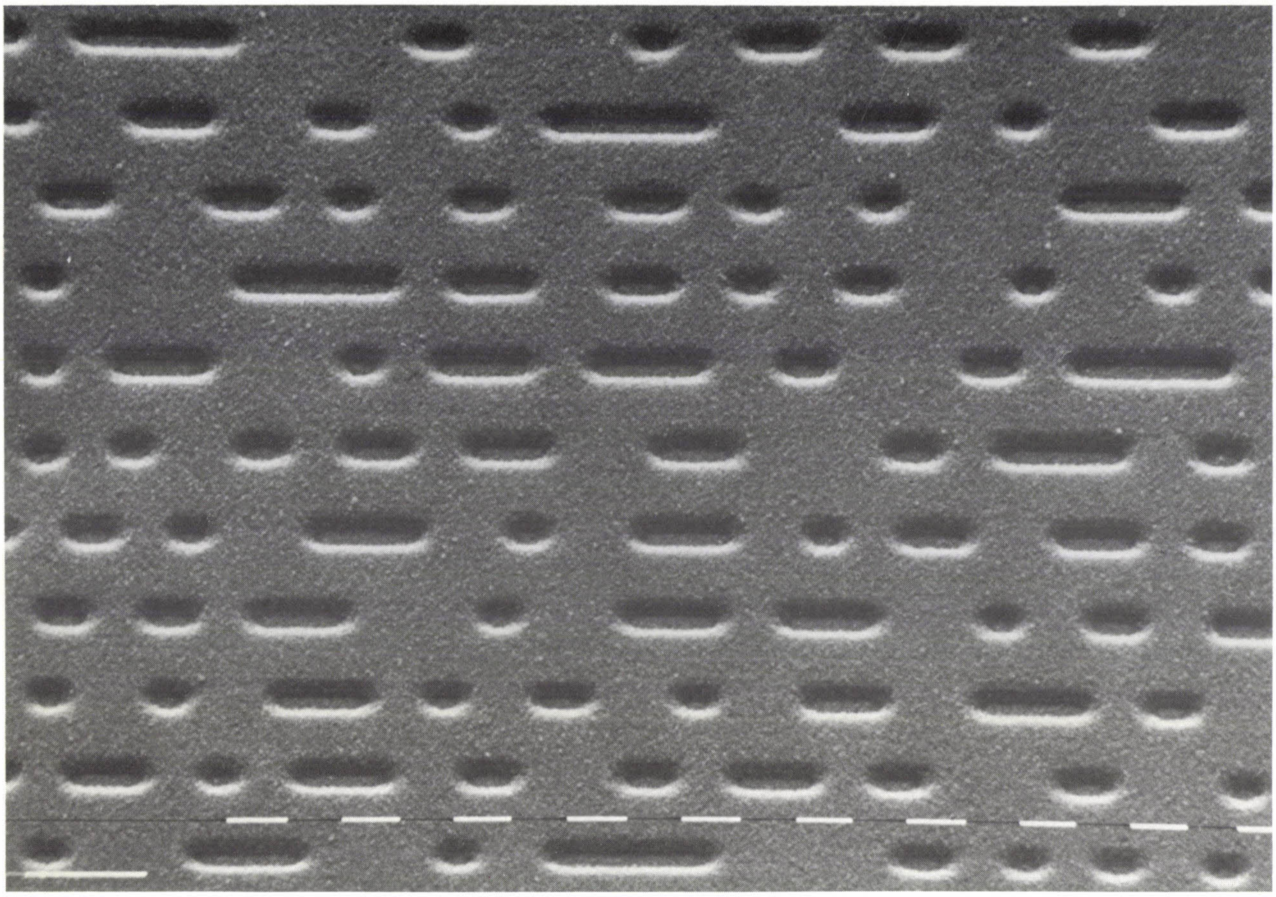
If more errors than the permitted maximum occur, they can only be detected. In the *CIM* block (Concealment: Interpolation and Muting) the errors detected are then masked. If the value of a sample indicates an error, a new value is determined by linear interpolation between the preceding value and the next one. If two or more successive sample values indicate an error, they are made equal to zero (muting). At the same time a gradual transition is created to the values preceding and succeeding it by causing a number of values before the error and after it to decrease to zero in a particular pattern.

In the digital-to-analog converters *DAC* [7] the 16 bit samples first pass through interpolation filters *F* and are then translated and recombined to recreate the original analog audio signal *A* from the two audio channels *L* and *R*. Since samples must be recombined at exactly the same rate as they are taken from the analog audio signal, the DACs and also *CIM* and

determines the 'filling degree' of the buffer memory. The control is so arranged as to ensure that the buffer memory is at all times filled to 50% of its capacity. The analog signal from the player is thus completely free from wow and flutter, yet with only moderate requirements for the speed control of the disc.

[7] See D. Goedhart, R. J. van de Plassche and E. F. Stikvoort, Digital-to-analog conversion in playing a Compact Disc, this issue, p. 174.

Summary. Digital processing of the audio signal and optical scanning in the Compact Disc system yield significant advantages: insensitivity to surface damage of the disc, compactness of disc and player, excellent signal-to-noise ratio and channel separation (both 90 dB) and a flat response over a wide range of frequencies (up to 20 000 Hz). The Compact Disc, with a diameter of only 120 mm, gives a continuous playing time of an hour or more. The analog audio signal is converted into a digital signal suitable for transcription on the disc. After the digital signal has been read from the disc by an optical 'pick-up' the original audio signal is recreated in the player.



The information on the Compact Disc is recorded in digital form as a spiral track consisting of a succession of pits. The pitch of the track is 1.6 μm, the width 0.6 μm and the depth of the pits 0.12 μm. The length of a pit or the land between two pits has a minimum value of 0.9 and a maximum value of 3.3 μm. The scale at the bottom indicates intervals of 1 μm.