



PHILIPS

Service Service **Service**

**Basic course
Compact Disc**

Concern Service Centre
Supporting Services
Training Group



PHILIPS

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**Basic course
Compact Disc**



Student's manual

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Concern Service Centre
Supporting Services
Training Group

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This manual was developed by Concern Service in close cooperation with Audio Service and the Audio Division's Compact Disc Development Laboratory.

The contents of this manual provide a basic knowledge of the Compact Disc system, the way it is marketed by Philips.

Each chapter is preceded by a number of questions which refer to the subjects treated in the chapter concerned.

If you are able to answer these questions, it will suffice to only go over to the summary.

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The progressive development of technology has, in its turn, made it possible to bring the storage and reproduction of sound to greater perfection.

At present it is possible to obtain good sound reproduction from high-quality discs.

However a number of difficult problems still remain in the conventional system of mechanical scanning. These are:

- disc wear
- damage to the discs
- signal-to-noise ratio
- dynamic range
- channel separation
- wow and flutter

The Compact Disc system offers a solution to these problems.

Since the disc is read out optically there is no mechanical contact between the pick-up unit and the disc.

Thus, the occurrence of wear is prevented.

The information on the disc is covered by a relatively thick layer of transparent plastic so that scratches, dust and dirt cannot damage it. Furthermore, the laser beam that scans the disc is focused relatively far below the plastic surface so that dust, scratches and dirt will hardly (if at all) affect the ultimate audio signal.

Digitization of signal processing has resulted in a signal-to-noise ratio and a dynamic range better than 90 dB for the Compact Disc system.

Digitization has also made it possible to correct errors which can occur during signal transmission. For this purpose, the digital information on the disc is supplemented by additional codes. These codes enable the player to check and, where possible, correct the digital information.

Audio information of one channel cannot affect the information contained in the other channel when applying the Compact Disc system, because the signals are processed completely separately.

This ensures an impressively good channel separation.

Wow and flutter belong to the past; within defined limits, processing of the signals occurs independently of variations in the turntable's rotational speed, the reference being a crystal oscillator.

All these good qualities make the Compact Disc a unique device with an extremely good quality of reproduction.

General block diagram

Questions

- 1. What is an optical pick-up unit composed of?**
- 2. How is an analogue signal converted to a digital signal?**
- 3. What must be added to the digital signal to enable the correction of errors?**
- 4. How is the string of 0s and 1s recorded on the disc?**
- 5. What are the functions of**
 - DEMOD?**
 - ERCO?**
 - CIM?**
 - FIL?**
 - DAC?**
- 6. How is the radial error signal obtained?**
- 7. How is the rotational speed of the motor controlled?**
- 8. What provision has been made to keep the laser beam focussed on the reflective surface of the disc?**

Description of the general block diagram



Introduction

A distinctive feature of the Compact Disc system is that the information contained in a reflective disc is read out by means of an optical system.

The information is present in digital form, and represented by two electrical levels only, i.e. 0 volts and +5 volts.

Inside the player the information is demodulated and existing errors are, whenever possible, corrected before a digital-to-analogue converter reconverts the digital signal into a reproduction of the original analogue audio signal.

The optical system is composed of a laser, a set of lenses and light-sensitive diodes (photodiodes).

The lens system ensures correct focusing of the laser light on the disc surface, whereas the photodiodes convert the reflected light into electric signals.

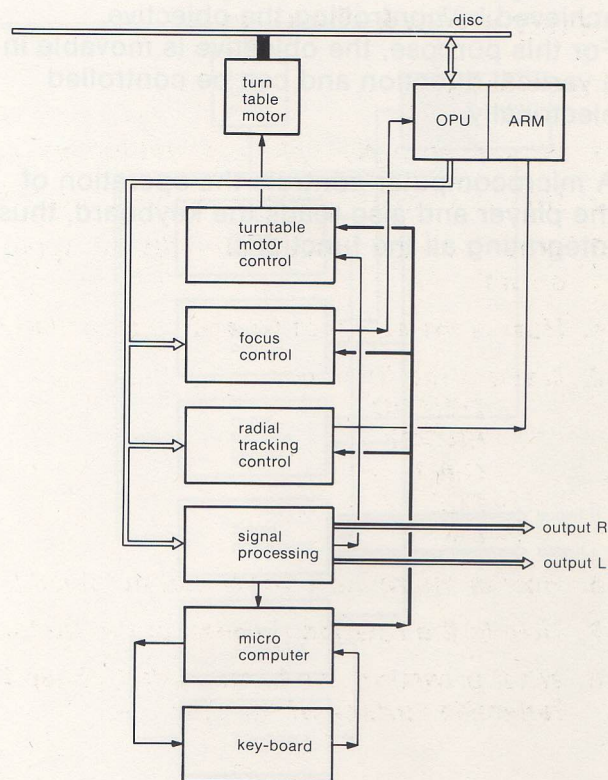
The information is present on the disc in the form of pits or depressions, pressed into the reflective surface.

Light that hits these pits is scattered in such a way that the intensity of the reflected light falling on the photodiodes is much lower than that of the light reflected back from the reflective surface between the pits.

This results in variations of the electric signal that leaves the photodiodes.

During disc read-out the pick-up unit, comprising a laser and a set of lenses, will have to follow the track with great accuracy. There exists, however, no mechanical contact between the pick-up unit and the disc to achieve correct mutual positioning.

By using more than one photodiode it becomes possible to measure the deviation of and hence correct the position of the arm holding the pick-up unit.



The speed at which the turntable is rotating also needs regular adjustment to ensure a constant rate of recovery of information. To achieve this, the speed at which the information is read from the disc is measured; if the rate is too high, the turntable is braked; too low a rate of information flow is corrected by increasing the turntable's rotational speed.

An essential requirement for the system's correct operation is that the laser light should be perfectly focused on the information track. The accuracy of focusing can be determined by means of the reflected light. If the system is out of focus, correction is achieved by controlling the objective. For this purpose, the objective is movable in a vertical direction and can be controlled electrically.

A microcomputer controls the operation of the player and also reads the keyboard, thus integrating all the functions.

The pick-up unit

The pick-up unit is composed of:

- a laser
- a prism
- a collimator lens
- an objective
- a few photodiodes

The laser generates a light beam which - via the other optical elements - finds its path to the disc, is reflected back and thus becomes the signal carrier.

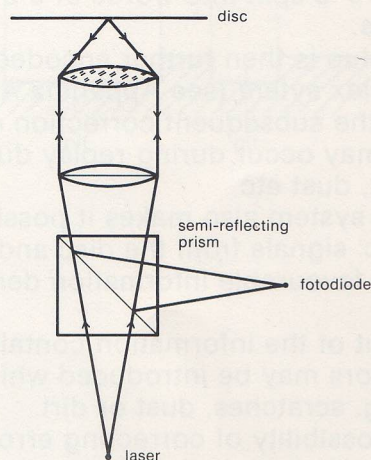
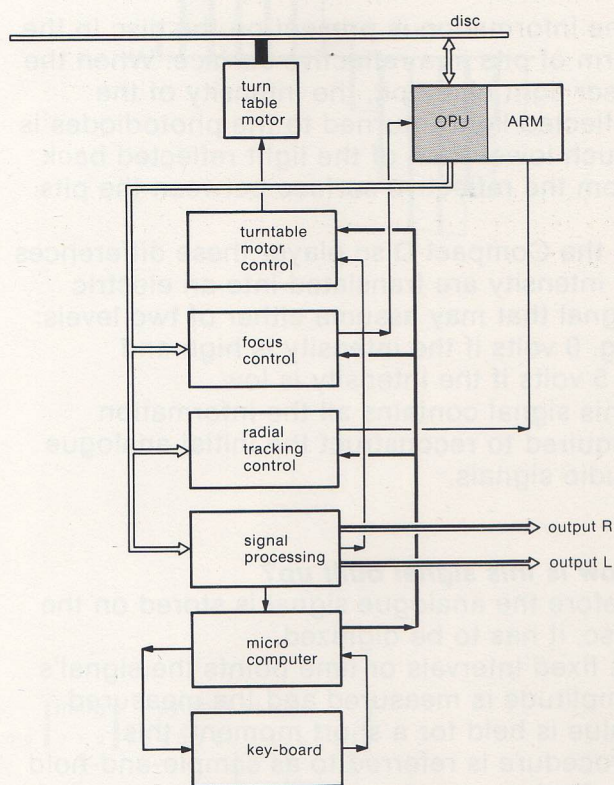
The prism offers undisturbed passage to the laser light going to the disc; the light reflected back from the disc surface is, however, deflected by the prism towards the photodiodes.

The collimator lens causes the laser beam to become a parallel light beam.

The objective's function is to focus the light beam on the disc.

The objective can be displaced in a vertical direction. If the distance between disc and objective is varying, the displacement facility makes it possible to maintain correct focusing by adjusting the objective's position.

The photodiodes convert the intensity-modulated light into an electrical signal. The player contains a total of 4 photodiodes. By suitably combining the signals from the photodiodes it is possible to obtain the audio information and to obtain also error signals to correct focus and radial tracking.



Signal processing

The information is present on the disc in the form of pits in a reflective surface. When the laser light hits a pit, the intensity of the reflected light returned to the photodiodes is much lower than of the light reflected back from the reflective surface between the pits.

In the Compact Disc player these differences in intensity are translated into an electric signal that may assume either of two levels: e.g. 0 volts if the intensity is high and + 5 volts if the intensity is low. This signal contains all the information required to reconstruct the initial analogue audio signals.

How is this signal built up?

Before the analogue signal is stored on the disc, it has to be digitized. At fixed intervals or time points the signal's amplitude is measured and the measured value is held for a short moment; this procedure is referred to as sample-and-hold method.

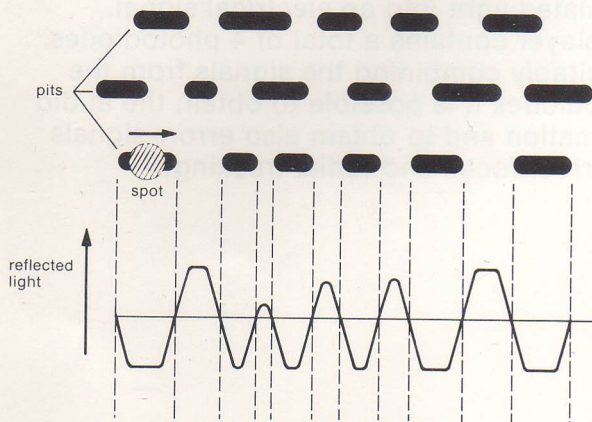
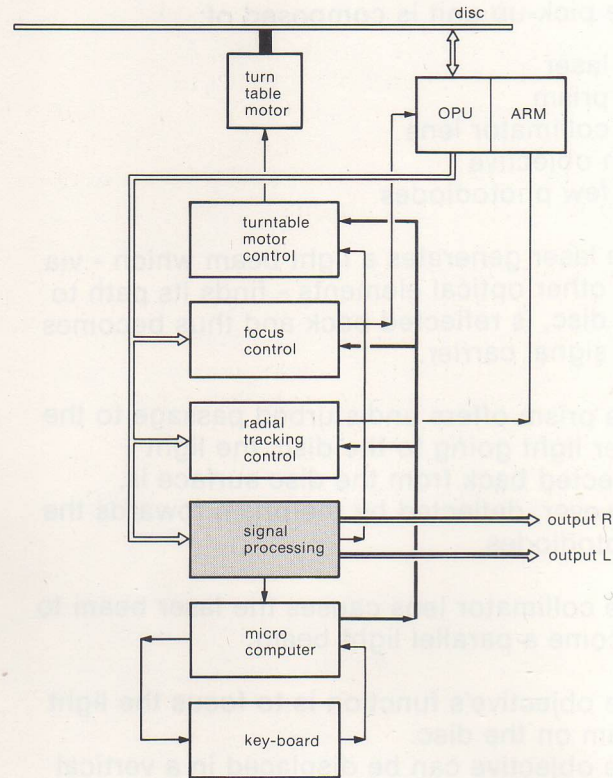
With the aid of an analogue-to-digital converter the amplitude of the sample is converted to a 16-bit binary value. The 16 bit word is split into words of 8 bits called symbols.

This binary value is then further encoded by a rather complex system (see Appendix A) which allows the subsequent correction of errors which may occur during replay due to scratches, dirt, dust etc.

The encoding system also makes it possible to obtain 'sync' signals from the disc and to obtain a more favourable information density on the disc.

During readout of the information contained in the disc errors may be introduced which result from e.g. scratches, dust or dirt.

To have the possibility of correcting errors, if any, further symbols, called parity symbols,

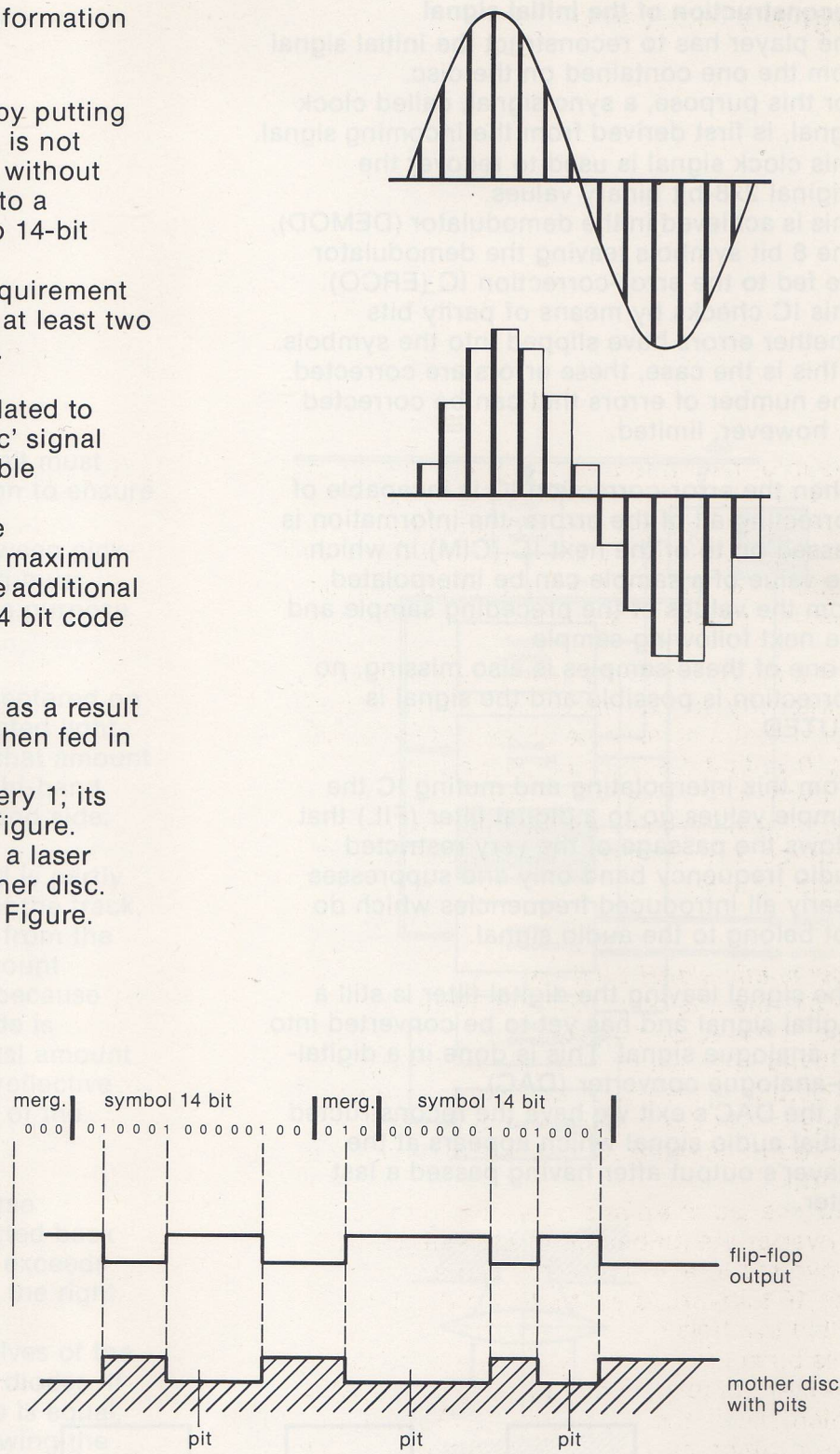


are added to the existing audio information symbols.

The string of 1s and 0s obtained by putting all the symbols after one another, is not suited to be recorded on the disc without more; these symbols are first fed to a modulator that converts them into 14-bit symbols. These 14-bit symbols meet the requirement that a ONE is always followed by at least two ZEROS and maximum 10 ZEROS.

This requirement has been formulated to enable the player to obtain a 'sync' signal from the disc and a more favourable information density on the disc. In between the symbols, the same requirement of minimum two and maximum ten zeros must be met for this three additional bits are inserted between the 14 bit code words, called merging bits.

The string of 0's and 1's obtained as a result of the above encoding system is then fed in serial form to a flip-flop. The flip-flop changes states at every 1; its output signal is as shown in the Figure. In turn, the flip-flop output drives a laser which burns the pits into the mother disc. The results are also shown in the Figure.



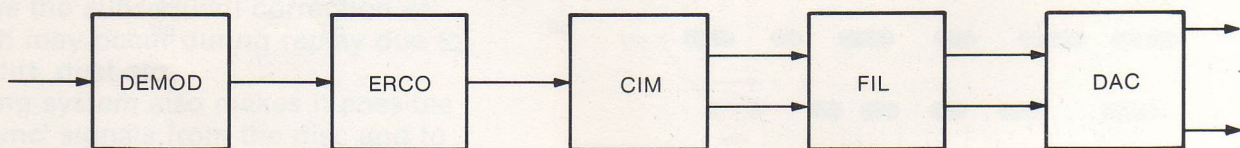
Reconstruction of the initial signal

The player has to reconstruct the initial signal from the one contained on the disc. For this purpose, a sync signal, called clock signal, is first derived from the incoming signal. This clock signal is used to recover the original 2x8-bit binary values. This is achieved in the demodulator (DEMOM). The 8 bit symbols leaving the demodulator are fed to the error-correction IC (ERCO). This IC checks by means of parity bits whether errors have slipped into the symbols. If this is the case, these errors are corrected. The number of errors that can be corrected is, however, limited.

When the error-correction IC is incapable of correcting all of the errors, the information is passed on to of the next IC (CIM) in which the value of a sample can be interpolated from the values of the preceding sample and the next following sample. If one of these samples is also missing, no correction is possible and the signal is MUTED.

From this interpolating and muting IC the sample values go to a digital filter (FIL) that allows the passage of the very restricted audio frequency band only and suppresses nearly all introduced frequencies which do not belong to the audio signal.

The signal leaving the digital filter is still a digital signal and has yet to be converted into an analogue signal. This is done in a digital-to-analogue converter (DAC). At the DAC's exit we have the reconstructed initial audio signal which appears at the player's output after having passed a last filter.



Tracking

During disc readout the pick-up unit must follow the track with great precision to ensure a reliable signal.

Since any mechanical contact between pick-up unit and disc is absent, tracking must occur by electronic means. For this purpose reflected light is used.

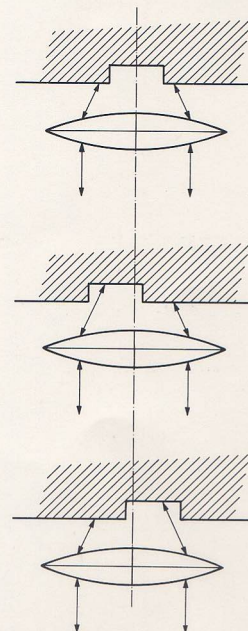
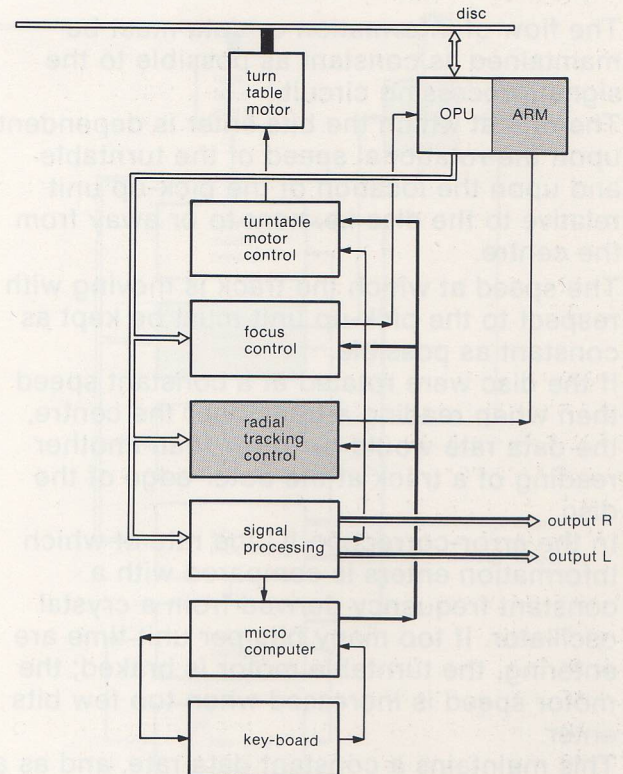
When the light beam is correctly centered on the track, the intensity of the reflected light will be equal at any point, that is, that amount of light reflected back from the right-hand side will equal that from the left-hand side.

If the beam has shifted slightly and is partly falling e.g. at the right-hand side of the track, the amount of light reflected back from the right-hand side will exceed the amount returning from the left-hand side, because part of the light at the left-hand side is scattered by the track, whereas total amount of light is reflected back from the reflective disc surface at the right-hand side of the track.

A beam shift to the left will of course augment the amount of light reflected back from the left of the track so that it exceeds the amount of light returning from the right.

In the Compact Disc player the halves of the reflected light fall on two different diodes. If the amount of light on either diode is equal, the pick-up unit is accurately following the track; in the event of a deviation to the right or to the left, resulting in unequal amounts of light, a difference signal is obtained by subtracting the output signal of the diodes from one another. This difference signal is referred to as **radial error signal**.

This error signal is used to control the arm holding the pick-up unit until the error has been cancelled.



Turntable motor control

The flow of information or data must be maintained as constant as possible to the signal processing circuit.

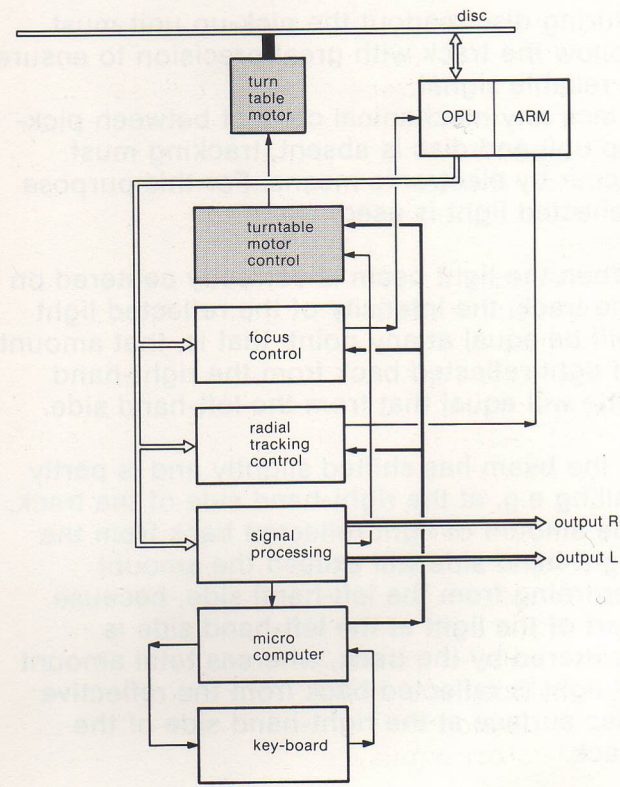
The rate at which the bits enter is dependent upon the rotational speed of the turntable and upon the location of the pick-up unit relative to the disc i.e. near to or away from the centre.

The speed at which the track is moving with respect to the pick-up unit must be kept as constant as possible.

If the disc were rotated at a constant speed then when reading a track near the centre, the data rate would be lower than another reading of a track at the outer edge of the disc.

In the error-correction IC the rate at which information enters is compared with a constant frequency derived from a crystal oscillator. If too many bits per unit time are entering, the turntable motor is braked; the motor speed is increased when too few bits enter.

This maintains a constant data rate, and as a consequence, virtually eliminates wow and flutter.



Focus control

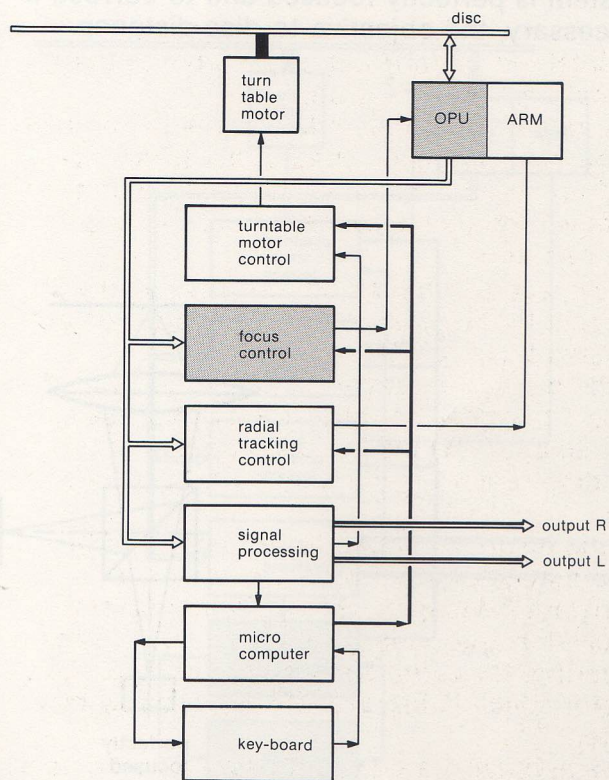
In order to obtain a reliable signal it is necessary that the light beam be properly focused on the track of the disc. Disc and turntable tolerances may cause variations in disc-to-objective spacing and, consequently, lead to incorrect focusing of the light beam on the disc, unless certain provisions are made.

To maintain a constant distance between disc and objective the objective is made movable in vertical direction.

By passing a greater or lesser amount of current in one direction through a coil, the objective can be moved e.g. down to a more or less extent.

By feeding the current in opposite direction through the coil an upward movement is generated.

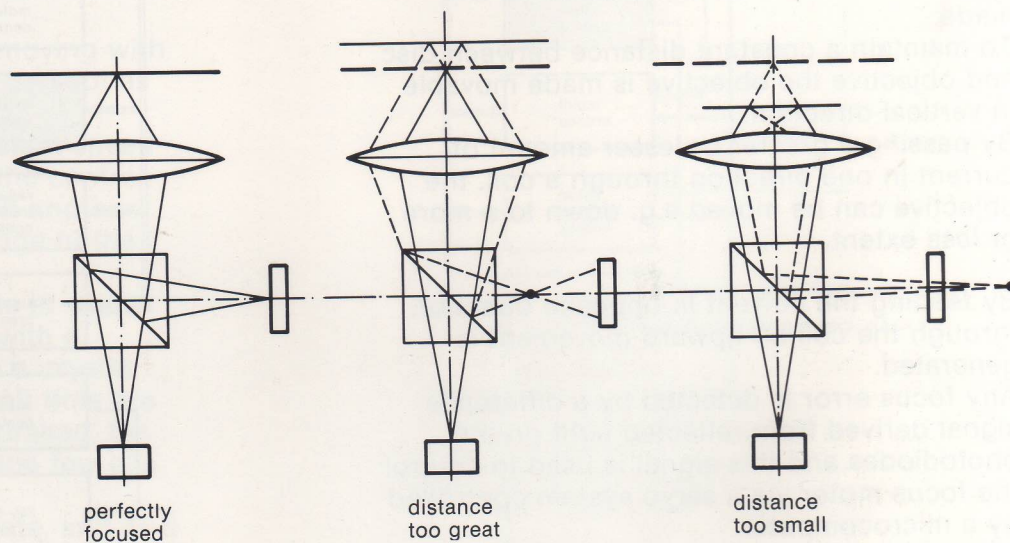
Any focus error is detected by a difference signal derived from reflected light on the photodiodes and this signal is used to control the focus motor via a servo system controlled by a microcomputer.



The information whether the distance is too great, too small or perfect can be obtained from the reflected light.

If the focal point coincides with the information plane on the disc, the focal points of the reflected light will also coincide with the diodes. Too great or too small a distance between disc and objective will bring the system out of focus and the focal points of the reflected light are in front of or behind the diodes.

The special arrangement of these diodes allows to determine whether or not the system is perfectly focused and to correct, if necessary, the objective-to-disc distance.



The microprocessor

The various different functions in the player need be correctly integrated (to one another), and also the commands entered via the keyboard must be processed.

A microcomputer is used to achieve this. It checks the various functions for correct operation and ensures that - e.g. during start-up of the player - the functions are correctly sequenced.

The conditions which the microcomputer monitors are:

Is the system perfectly focused?

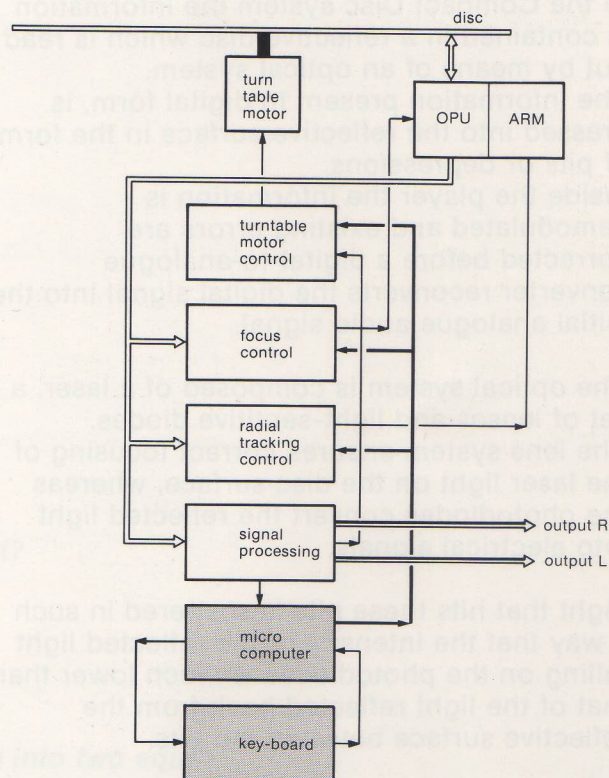
Does there exist a data flow?

Is the track correctly followed?

Has a key been depressed on the keyboard?

All these things are checked and, in the event of a deviation signals are generated to cause the equipment to respond appropriately. The cycle of the microprocessor contains a fixed programme for keyboard scanning; this programme is carried out at fixed time intervals.

If a key is depressed, the microprocessor sees to it that the command is carried out by instructing the related circuits.



Summary

In the Compact Disc system the information is contained in a reflective disc which is read out by means of an optical system.

The information present in digital form, is pressed into the reflective surface in the form of pits or depressions.

Inside the player the information is demodulated and existing errors are corrected before a digital-to-analogue converter reconverts the digital signal into the initial analogue audio signal.

The optical system is composed of a laser, a set of lenses and light-sensitive diodes.

The lens system ensures correct focusing of the laser light on the disc surface, whereas the photodiodes convert the reflected light into electrical signals.

Light that hits these pits is scattered in such a way that the intensity of the reflected light falling on the photodiodes is much lower than that of the light reflected back from the reflective surface between the pits.

During disc read-out the pick-up unit will have to follow the track with great accuracy. Using more than one photodiode makes it possible to measure the deviation and hence correct the position of the arm holding the pick-up unit.

The speed at which the turntable is rotating also needs regular adjustment. To achieve this, the rate at which the information enters is measured.

The accuracy of focusing can also be determined by means of the reflected light. If the system is out of focus, correction is achieved by controlling the objective.

A microcomputer controls the operation of the player and reads the keyboard, thus integrating all the functions.

The Compact Disc Player

Questions

The disc

- 1. What is the disc composed of?**
- 2. At what speed is the information track scanned?**
- 3. At what speed does the disc rotate?**

The optical pick-up unit

- 1. What kind of laser is applied in the CD player?**
- 2. How many light beams emerge from the laser?**
- 3. On what is the amount of emergent light dependent?**
- 4. How is the intensity controlled?**
- 5. What is the function of the**
 - . prism?**
 - . photodiodes?**
- 6. In what way is a split-up of the reflected light beam into two equal beams obtained?**
- 7. How is the objective moved up and down?**

Description of the decoder

- 1. What kind of output signals are supplied by the demodulator?**
- 2. How is the bit clock re-generated from the incoming data?**
- 3. How can DEMOD determine where the flow of data begins?**
- 4. What does modulation imply?**
- 5. Which tasks are performed by the error corrector IC?**
- 6. How is the motor speed controlled?**
- 7. How is de-interleaving obtained?**

8. What happens in the error correction decoders in case of
 - no errors?
 - one error?
 - two errors?
 - more than two errors?
9. What function has the interpolating and muting IC?
10. What processing is first performed on the data in the CIM?
11. Which actions are undertaken by the CIM if
 - a single error is flagged?
 - two or more errors are flagged?
 - after a drop-out a new signal is present?
12. What filtering system is used in the Philips Compact Disc player?
13. Which method is used in this filter?
14. How many bits can the DAC convert to an analogue signal?

Servo systems

1. Which mechanical movements in the CD player are controlled by a servo system?
2. When will a servo system become unstable?
3. How is a servo system in the Compact Disc prevented from becoming unstable?
4. How is the stability of a system affected by the gain?
5. How is the radial error signal obtained?
6. Which other factors can make the radial tracking system unstable?
7. Why have the factors d and k been introduced?
8. How are the factors d and k obtained?
9. How are the signals of the photodiodes affected by a change in the distance between disc and objective?

The Compact Disc player

The disc

The CD disc is composed of a plastic base material or carrier into which the information has been pressed in the form of pits. Next, a reflective layer is placed over the pits and a transparent protective coating is added over the reflective layer.

The total playing time of a CD disc is 60 minutes.

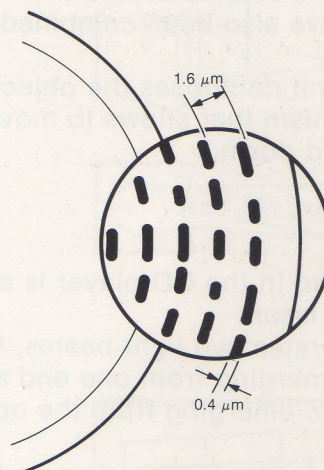
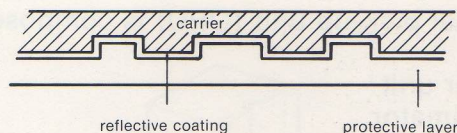
The diameter of the disc is 120 mm.

The average pitch of the information track is $1.6\text{ }\mu\text{m}$; the width of the pits is $0.4\text{ }\mu\text{m}$.

The disc is single sided.

The information track is scanned at a constant speed of 1.3 m/s . To achieve this, the rotational speed is gradually stepped down from 500 rpm at the beginning of the disc near the centre of the disc to 200 rpm near the outer edge of the disc.

This chapter will enter into more detail on the Compact Disc player, starting with the disc and the optical element and followed by the signal path and the servo systems.



The optical pick-up unit

The optical pick-up unit (OPU) is composed of:

- the laser unit
- the collimator
- the photodiodes
- the focusing unit

The laser and the prism are accommodated in the laser unit; the collimator and the photodiodes have also been combined into a single unit.

The focusing unit comprises the objective and the mechanism that allows to move the objective up and down.

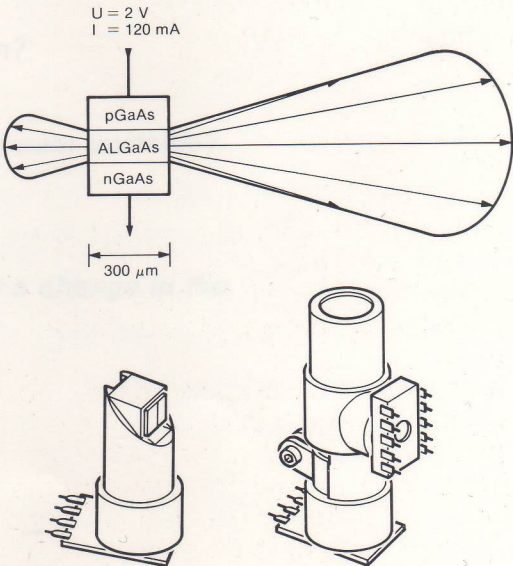
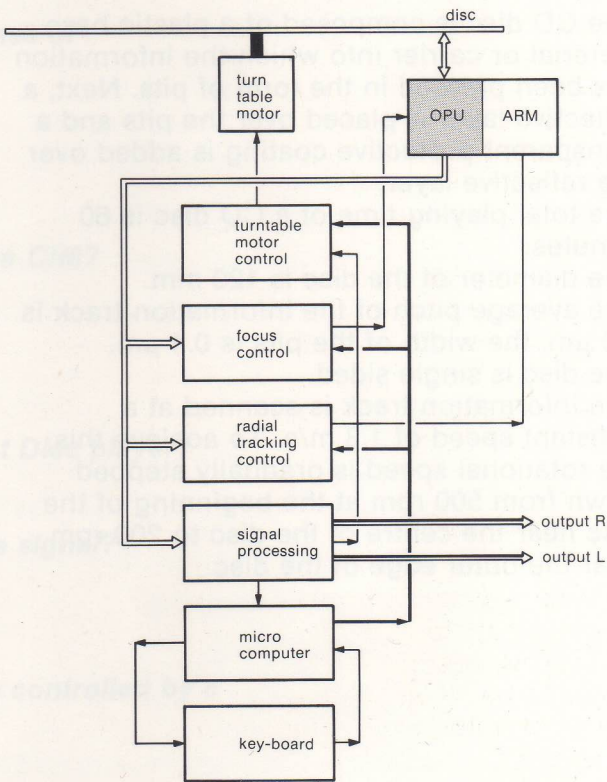
The laser

The laser applied in the CD player is a semiconductor laser.

This laser generates two light beams, that is, a main beam emerging from one end and a secondary beam emerging from the opposite end.

The main beam goes via the prism and the set of lenses to the disc; the secondary beam falls on a photodiode that has been incorporated in the laser casing.

There exists a fixed relationship between the intensities of the main and secondary light beams. An important factor for good operation of the player is that the amount of light falling on the disc is kept as constant as possible with time. The intensity is, however, dependent on the current fed through the laser and on the temperature of the diode. Consequently, the intensity must be kept constant by controlling the laser current. The light of the secondary beam is taken as measure for the intensity of the main beam.



The amount of light of the secondary beam is measured with a photodiode installed below the laser. The signal of this diode goes to the laser supply and thus determines the current through the laser and, consequently, the intensity.

If the intensity is too low, the diode signal is too small and the current will be increased; if the intensity is too high, the diode signal will be too strong and the current through the laser is reduced.

Prism and photodiodes

The light that leaves the laser goes to a semi-reflecting prism; this prism lets pass the light coming from the laser and deflects the light beam returning from the disc to the photodiodes.

The photodiodes convert the reflected light into electric signals. A total of four diodes have been mounted slightly apart in groups of two.

The light belonging to one half of the reflected light beam falls on one diode pair, that belonging to the other falls on the other diode pair. To obtain a split-up of the light beam into two equal beams, one side of the prism has been ground to wedge shape.

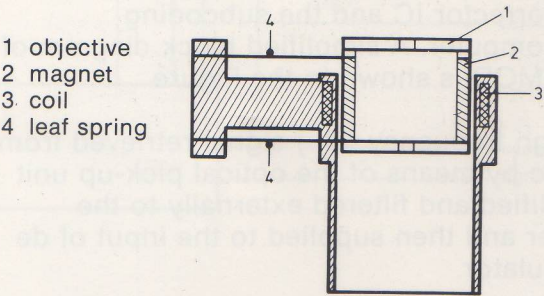
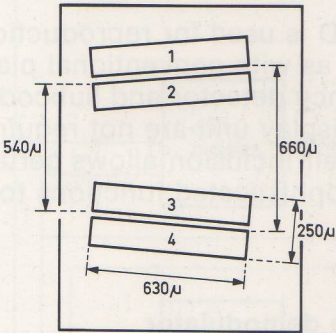
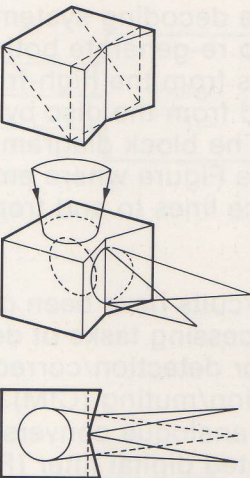
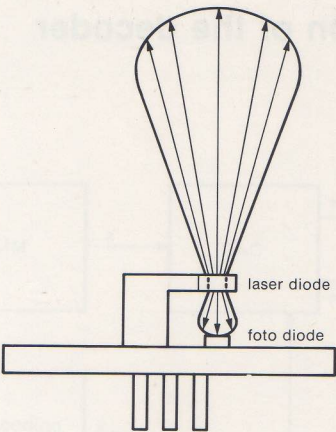
The collimator

After the laser beam has passed the prism, it goes to the collimator where the diverging beam is transformed into a parallel beam. The advantage of a parallel beam is that the distance between collimator and the next lens does not influence the functioning of the set of lenses.

This feature is required because the objective that follows the collimator has to move in a vertical direction to maintain perfect focus with a disc moving in a vertical direction due to tolerances.

The objective

The objective is installed on a ring of magnetic material; the latter is attached to a housing by means of two leaf springs. This housing contains a coil which produces a magnetic field when an electric current is fed through it. This magnetic field being proportional to the current traversing the coil, exerts a force on the magnetic material onto which the objective has been mounted, thus making the objective move up or down, depending on the polarity and magnitude of the coil current.



Description of the decoder

Introduction

The task of the decoding system within a CD player is to re-generate both analogue audio channels from the high-frequency data being retrieved from the disc by the optical pick-up unit. The block diagram of a decoder is shown in the Figure where emphasis is put on the interface lines to and from the servo systems.

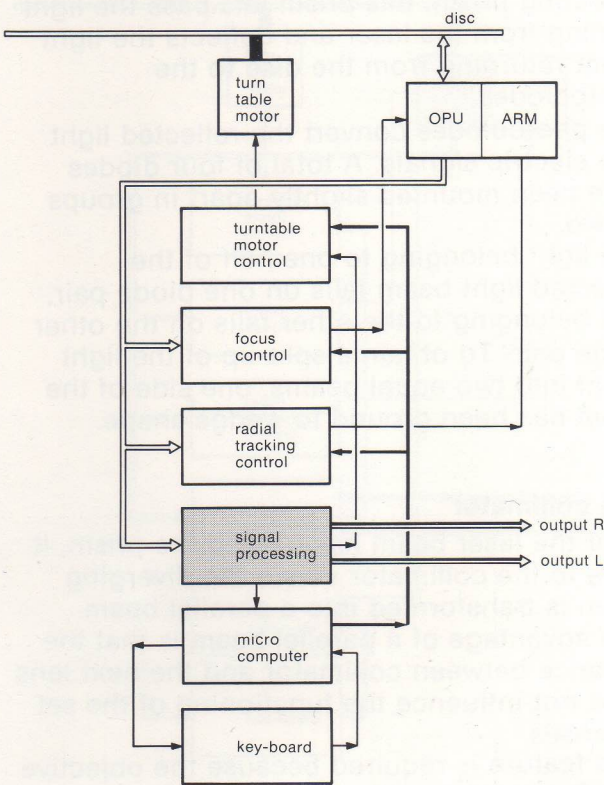
Special LSI circuits have been developed for the signal processing tasks of demodulation (DEMOM) error detection/correction (ERCO) and interpolation/muting (CIM). The digital-to-analogue conversion is based upon a dedicated digital filter (FIL) and a 14-bit D/A converter (DAC) which, when combined, give the performance of a 16-bit system with the additional benefits of a linear phase characteristic and reduced distortion.

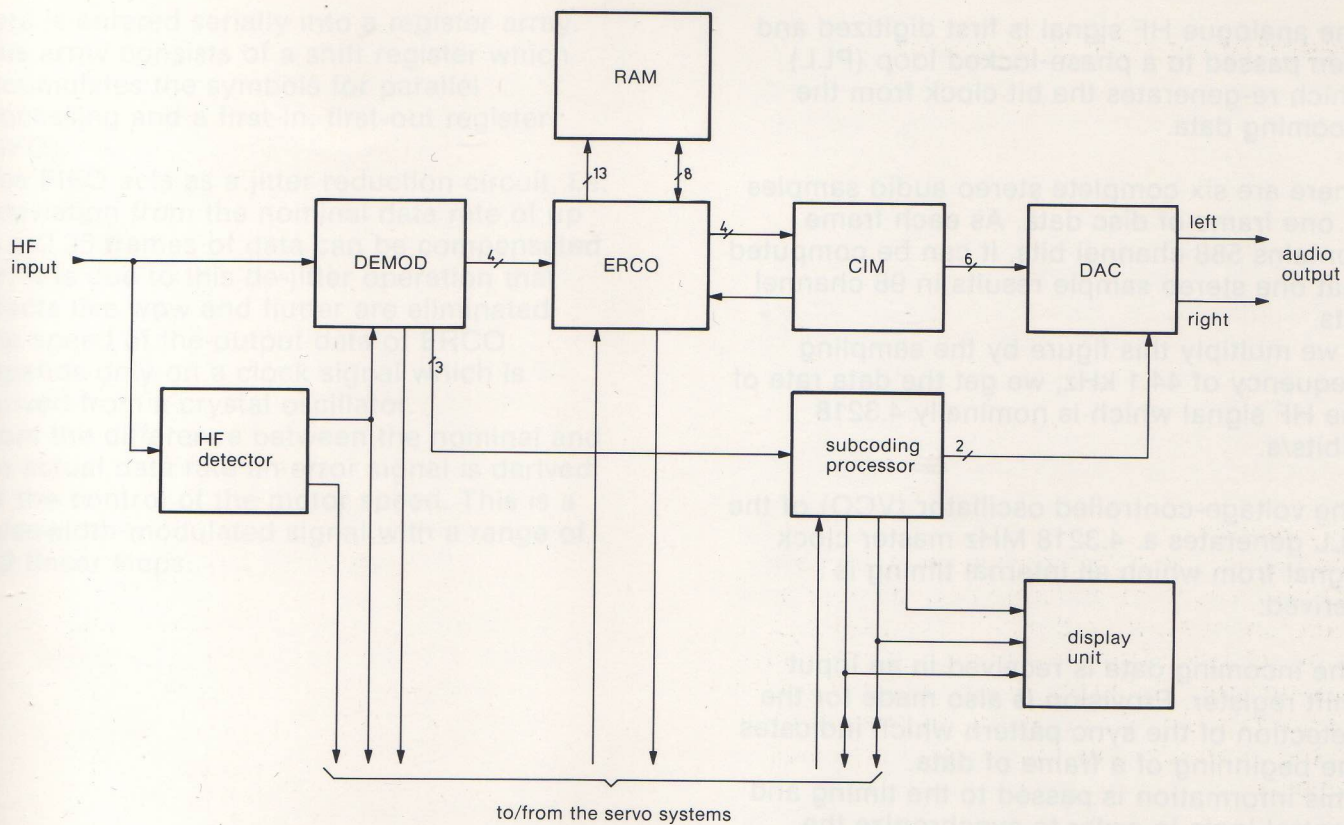
When the CD is used for reproduction of sound only, as with conventional players, the high frequency detector and subcoding processor display unit are not required. However, their inclusion allows certain additional sophisticated functions to be realised.

DEMOM, the demodulator

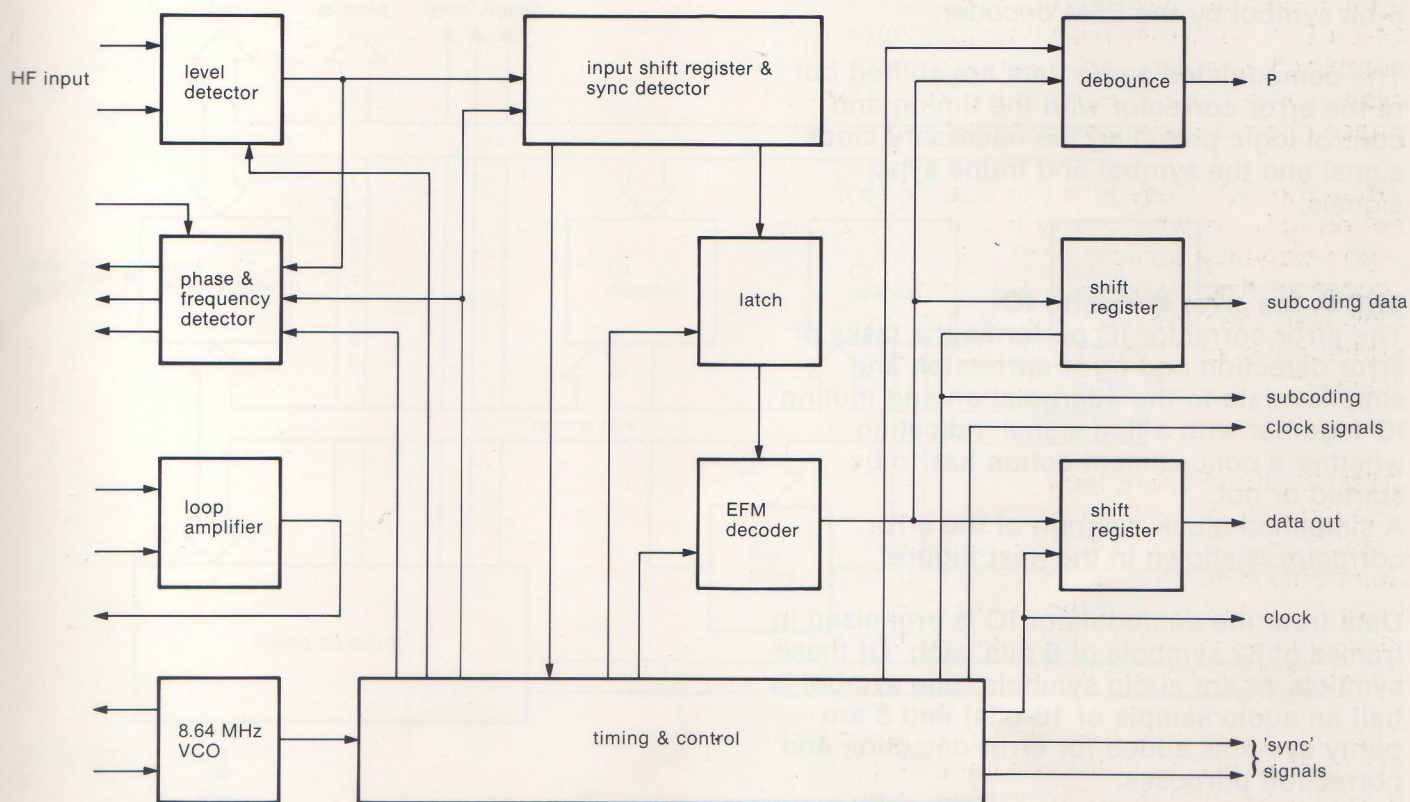
The demodulator IC forms the front-end of the CD decoding system and supplies demodulated data and timing signals to the error corrector IC and the subcoding microcomputer. A simplified block diagram of the DEMOM is shown in the Figure.

The high frequency (HF) signal retrieved from the disc by means of the optical pick-up unit is amplified and filtered externally to the decoder and then supplied to the input of demodulator.





DECODER



DEMOD

The analogue HF signal is first digitized and then passed to a phase-locked loop (PLL) which re-generates the bit clock from the incoming data.

There are six complete stereo audio samples in one frame of disc data. As each frame contains 588 channel bits, it can be computed that one stereo sample results in 98 channel bits.

If we multiply this figure by the sampling frequency of 44.1 kHz, we get the data rate of the HF signal which is nominally 4.3218 Mbits/s.

The voltage-controlled oscillator (VCO) of the PLL generates a 4.3218 MHz master clock signal from which all internal timing is derived.

The incoming data is received in an input shift register. Provision is also made for the detection of the sync pattern which indicates the beginning of a frame of data.

This information is passed to the timing and control logic in order to synchronize the demodulation process to the received data.

Provided the timing generator is locked to the HF signal, each received 14-bit word is stored in a latch and is then demodulated into an 8-bit symbol by the EFM decoder.

The demodulated audio data are shifted out to the error corrector with the timing and control logic providing the necessary clock signal and the symbol and frame sync signals.

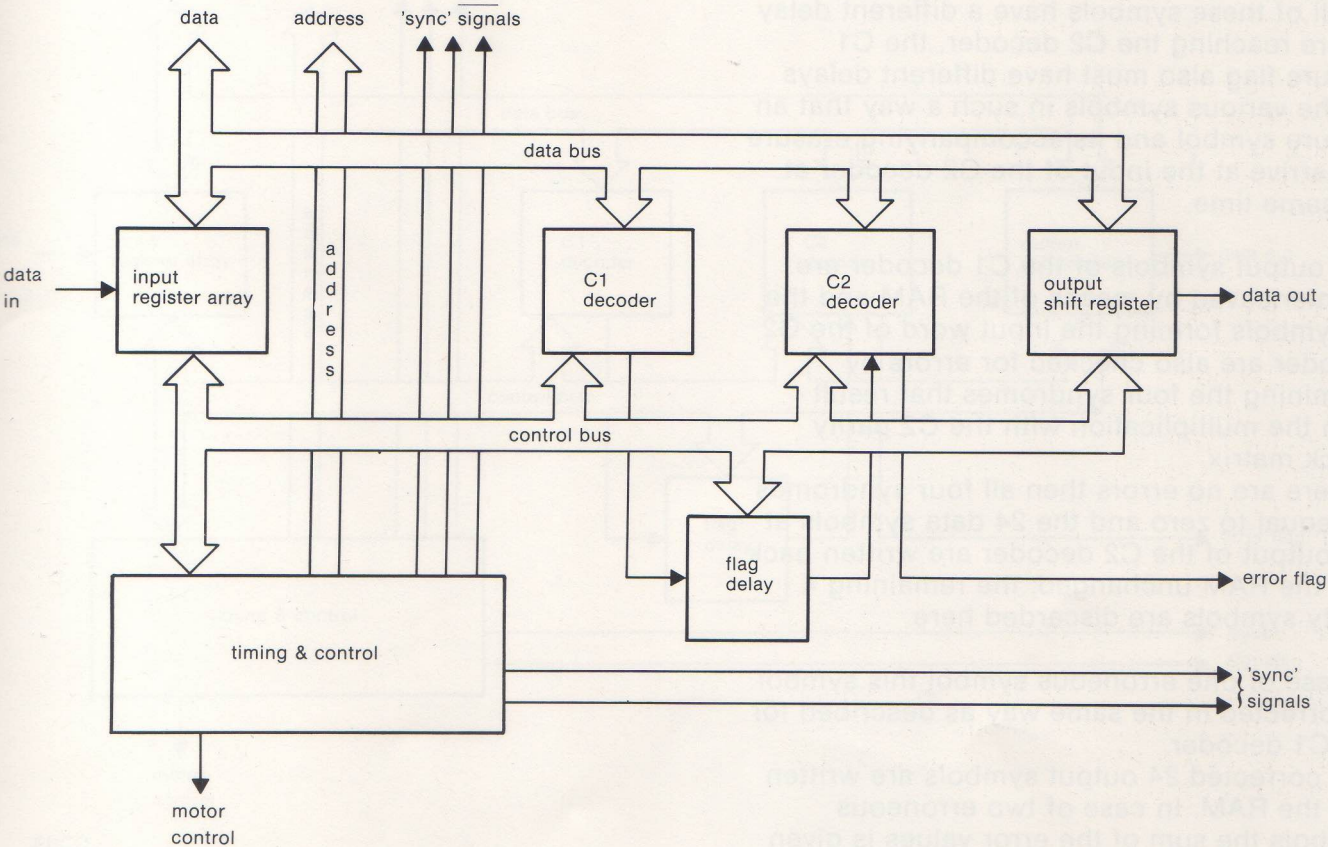
ERCO, the error corrector IC

The error corrector IC performs the tasks of error detection and error correction and supplies data to the interpolation and muting IC together with a flag signal indicating whether a concealment action has to be started or not.

A simplified block diagram of the error corrector is shown in the next Figure.

Data from the demodulator IC is organized in frames of 32 symbols of 8 bits each. Of these symbols 24 are audio symbols (one symbol is half an audio sample of 16-bits) and 8 are parity symbols added for error detection and correction purposes.

Data is entered serially into a register array. This array consists of a shift register which accumulates the symbols for parallel processing and a first-in, first-out register (FIFO). This FIFO acts as a jitter reduction circuit, i.e. a deviation from the nominal data rate of up to ± 2.25 frames of data can be compensated for. It is due to this de-jitter operation that effects like wow and flutter are eliminated. The speed of the output data of ERCO depends only on a clock signal which is derived from a crystal oscillator. From the difference between the nominal and the actual data rate an error signal is derived for the control of the motor speed. This is a pulsewidth-modulated signal with a range of 142 linear steps.



ERCO

The interleaving of data symbols (being a part of the CIRC error correction encoding procedure, see Appendix A) is compensated for by suitable de-interleaving operations in the error corrector.

The de-interleaving is performed on data symbols prior to entering the C1 and C2 decoders by temporarily storing these symbols in an external RAM.

The data symbols from the demodulator are de-interleaved by means of the RAM and the 32 symbols forming the input word to the C1 decoder are checked for errors.

By multiplying the input word by the parity check matrix, four syndromes are obtained. If there are no errors then all four syndromes are equal to zero and the 28 data symbols at the output of the C1 decoder (4 of the parity symbols are discarded here) are written back into the RAM unchanged.

In case of one erroneous symbol this symbol is corrected and the 28 corrected output symbols are written into the RAM. In case of two or more erroneous symbols the 28 output symbols are written into the RAM unchanged and a flag is set to mark these 28 symbols as being unreliable.

If an erroneous C1 word results in a C1 flag then all of the symbols in the C1 output word are unreliable.

As all of these symbols have a different delay before reaching the C2 decoder, the C1 erasure flag also must have different delays for the various symbols in such a way that an erasure symbol and its accompanying erasure flag arrive at the input of the C2 decoder at the same time.

The output symbols of the C1 decoder are de-interleaved by means of the RAM and the 28 symbols forming the input word of the C2 decoder are also checked for errors by examining the four syndromes that result from the multiplication with the C2 parity check matrix.

If there are no errors then all four syndromes are equal to zero and the 24 data symbols at the output of the C2 decoder are written back into the RAM unchanged; the remaining 4 parity symbols are discarded here.

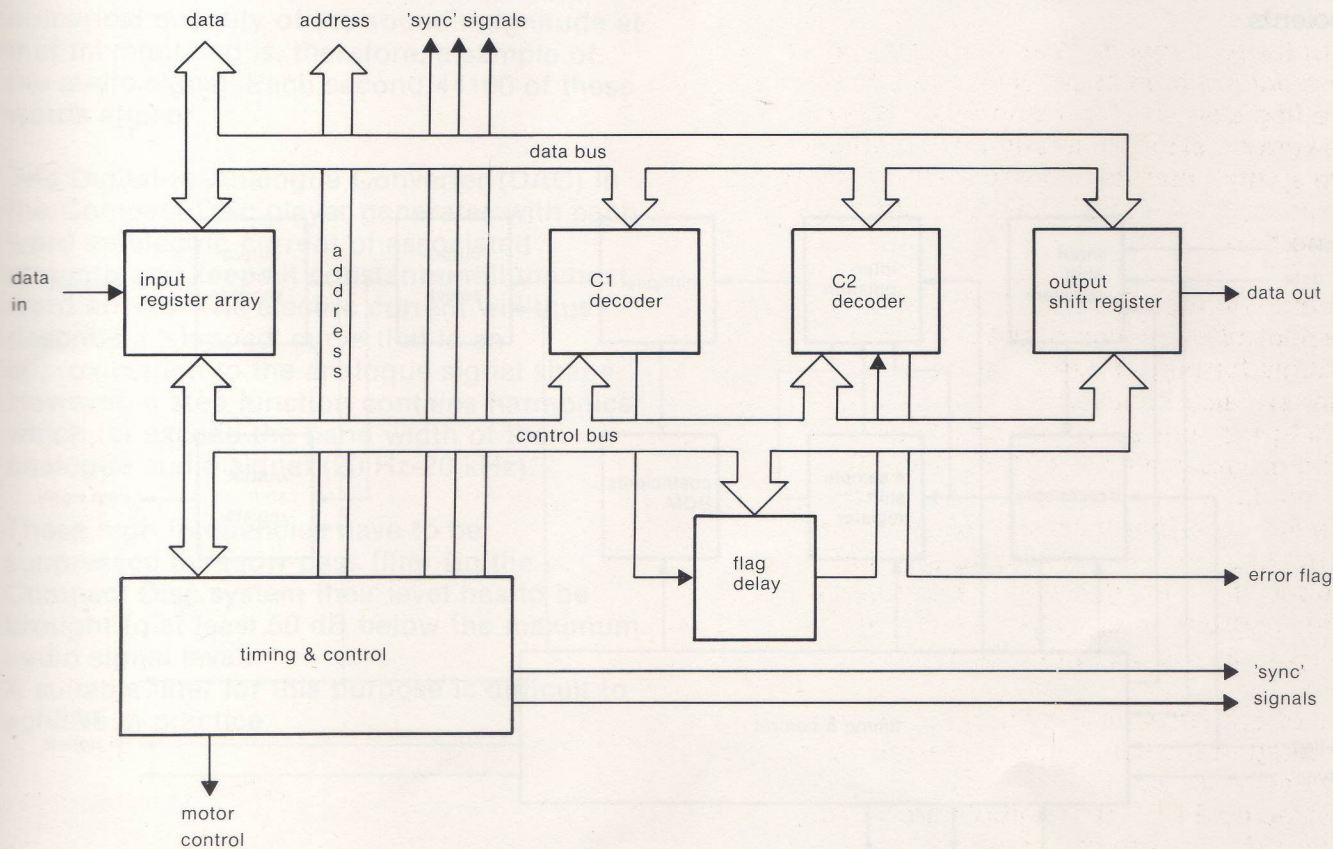
In case of one erroneous symbol this symbol is corrected in the same way as described for the C1 decoder.

The corrected 24 output symbols are written into the RAM. In case of two erroneous symbols the sum of the error values is given by one of the syndromes.

Information about the position of the erasure symbols is available due to the erasure flags. With this additional information both errors are corrected and the 24 corrected output symbols are written into the RAM. In case of more than two erroneous symbols in a C2 word all 24 symbols are written back into the RAM unchanged and a C2 flag is set to mark these 24 symbols as being unreliable.

The unreliable data signal provides two different kinds of information. If the flag is output simultaneously with a data symbol it marks that symbol. If it is output in the gap between two data samples it marks that sample that will reach the output 60 sample times later. This advanced flag is needed for the interpolation and muting circuit.

The output data are shifted out serially together with the data clock and the frame sync signal.



ERCO

CIM, the interpolation and muting IC

The interpolation and muting IC is used to reduce audible annoyance that could be caused by uncorrected erroneous data samples from the error corrector IC. A simplified block diagram is shown in the Figure.

Data from ERCO is entered serially with the shift clock.

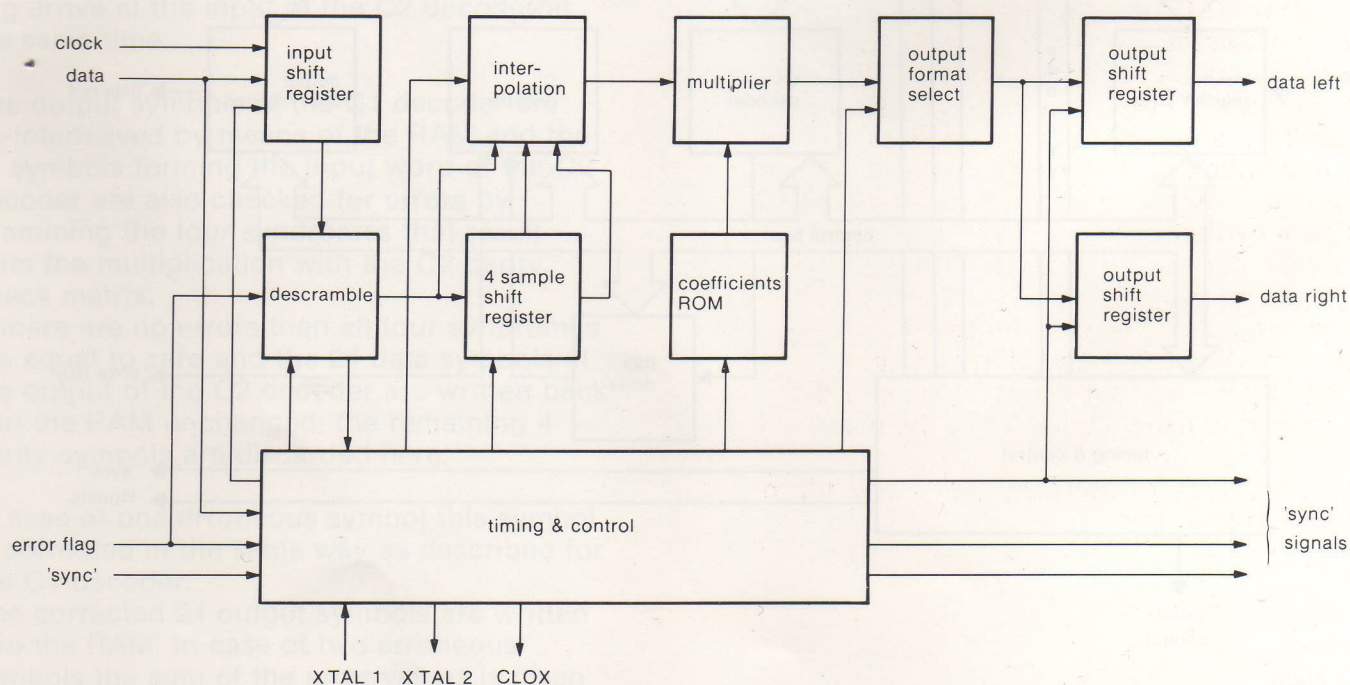
After that, data are descrambled (see Appendix A) and separated into left and right channel samples.

performed on the unreliable data flag. When no error flags occur, the data value is not affected through CIM.

If for either left or right channel a single 'error' is flagged between two 'good' samples then linear interpolation is used to replace the erroneous value. If two or more adjacent samples are flagged then the samples in error will be muted.

Beginning 30 samples before the consecutive errors, the value of the samples will be attenuated smoothly to zero level, using a (0 to π) cosine-shaped curve.

The coefficients of that curve are stored in a ROM and the attenuation is performed by a multiplication of the sample values with these coefficients.



CIM

After the error burst the next 30 samples are processed with a (π to 2π) cosine-shaped curve after which the full level is reached again. This muting or attenuation occurs to data of the left and right channel simultaneously regardless of the source of the mute.

A crystal oscillator is used to generate the internal timing signals as well as the clock signal CLOX for the error corrector and the digital filter.

By using the frame sync signal for internal timing reset, CIM automatically synchronizes to the ERCO output.

Digital-to-Analogue Conversion

The last section in the sequence of processings to which the signal in the Compact Disc system is subjected, is the re-transformation of the digital code into an analogue signal that has the same shape as the original acoustic vibration that was captured by the microphone.

The digital signal takes - after decoding and error correction - the form of a series of 16-bit words.

Each word represents in binary form the numerical quantity of the sound magnitude at that moment and is, therefore, a sample of the audio signal. Each second 44100 of these words appear.

The Digital-to-Analogue Converter (DAC) in the Compact Disc player generates with each word an electric current of associated strength and keeps it constant until the next word arrives. The electric current will thus describe a 'stepped' curve that is an approximation to the analogue signal shape. However, a step function contains harmonics which far exceed the band width of the analogue audio signal (20 Hz-20 kHz).

These high frequencies have to be suppressed by a low-pass filter (in the Compact Disc system their level has to be brought to at least 50 dB below the maximum audio signal level).

A suitable filter for this purpose is difficult to achieve in practice.



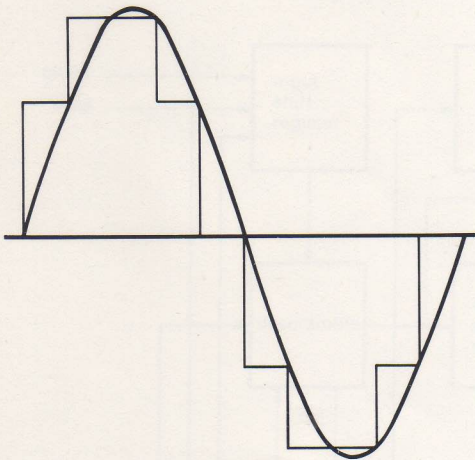
Digital filter

In order to allow mitigation of the low-pass filter requirements, the approach selected in the Philips Compact Disc player is to perform a preliminary filtering operation during the digital phase.

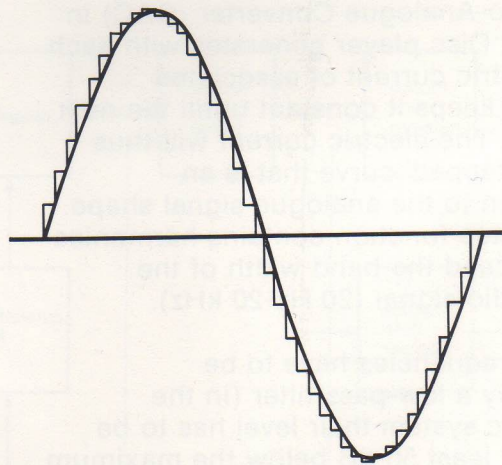
For this, a fourfold oversampling is performed, that is, a digital filter operating at four times the sampling frequency ($4 \times 44.1 \text{ kHz} = 176.4 \text{ kHz}$) provides signal values at this raised frequency, thus refining the stepped curve and facilitating the filtering out of the high frequencies. As a result, it will suffice to use a relatively simple third-order low-pass filter after the digital-to-analogue conversion section.

To convert the 16-bit words into an analogue signal the Philips Compact Disc player uses a 14-bit digital-to-analogue converter available in IC form; this DAC is designed to operate at the high sampling frequency of 176.4 kHz.

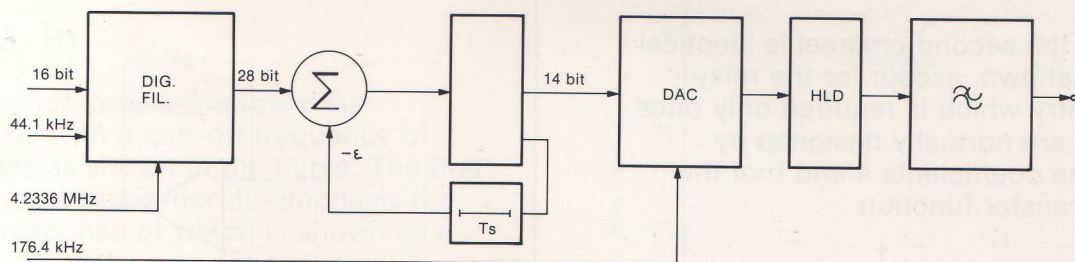
The fourfold oversampling on the one hand and a feedback in phase opposition of the rounding errors on the other hand allows the rounding-off to 14-bits without degrading the signal-to-noise ratio (SNR). This SNR is maintained at about the 96 dB level associated with 16-bit quantization.



$f_s = 44.1 \text{ kHz}$



$f_s = 176.4 \text{ kHz}$



$$4.2336 \text{ MHz} = 96 \times 44.1 \text{ kHz}$$

The set of operations belonging to the digital-to-analogue conversion are illustrated in the block diagram.

For stereo reproduction through two channels the complete chain is duplicated. Oversampling takes place in the digital filter to which the input signal is fed. The next step consists in rounding off to 14-bits; the rounding error is fed back in phase opposition in the noise shaper (NS). The digital filter and the noise shaper are accommodated in a single IC. Then follow the D/A converter and a hold circuit. Finally, the analogue signal passes through a low-pass filter.

Digital-to-Analogue Converter (DAC)

The DAC is the device that reconverts the binary codes which represent the amplitudes of the audio samples into an analogue signal. Each binary code has its corresponding current.

Example

code 0000 provides a current of 0 mA, 0001 a current of 1 mA, 0010 2 mA, 0011, 3 mA, etc.

In the CD player the codes to be converted do not comprise four bits like in the example, but 14 bits; the number of different currents is $2^{14} = 16\,384$.

The 14-bit codes which enter the DAC control a number of switches which enable or inhibit the flow of these currents to the output.

In practice this will mean an increment of $0.6104 \mu\text{A}$.

For a more detailed description of the DAC used in the Philips Compact Disc see Appendix B.

The Bessel filter

The application of the Bessel filter is shown in the Figure.

The filter for the second channel is identical with the one shown, except for the relay control circuitry which is required only once. Bessel filters are normally designed by looking up the coefficients a and b of the normalized transfer function

$$A(s) = \frac{1}{1 + S \cdot a/\omega_c + S^2 \cdot b/\omega_c^2}$$

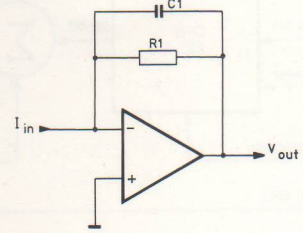
in a table and then finding suitable component values.

ω_c is the desired cut-off frequency of the complete Bessel filter which is equal to $2\pi \cdot 30 \text{ kHz}$.

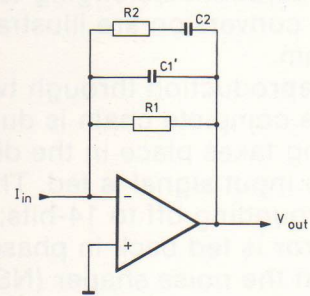
The coefficients for a third-order Bessel filter are found to be $a_1 = 0.7560$ and $b_1 = 0.0000$ for the one-pole section and $a_2 = 0.9996$ and $b_2 = 0.4772$ for the two-pole section.

A de-emphasis characteristic is easily implemented in the one-pole section of the filter. This is achieved by closing the contact 1a.

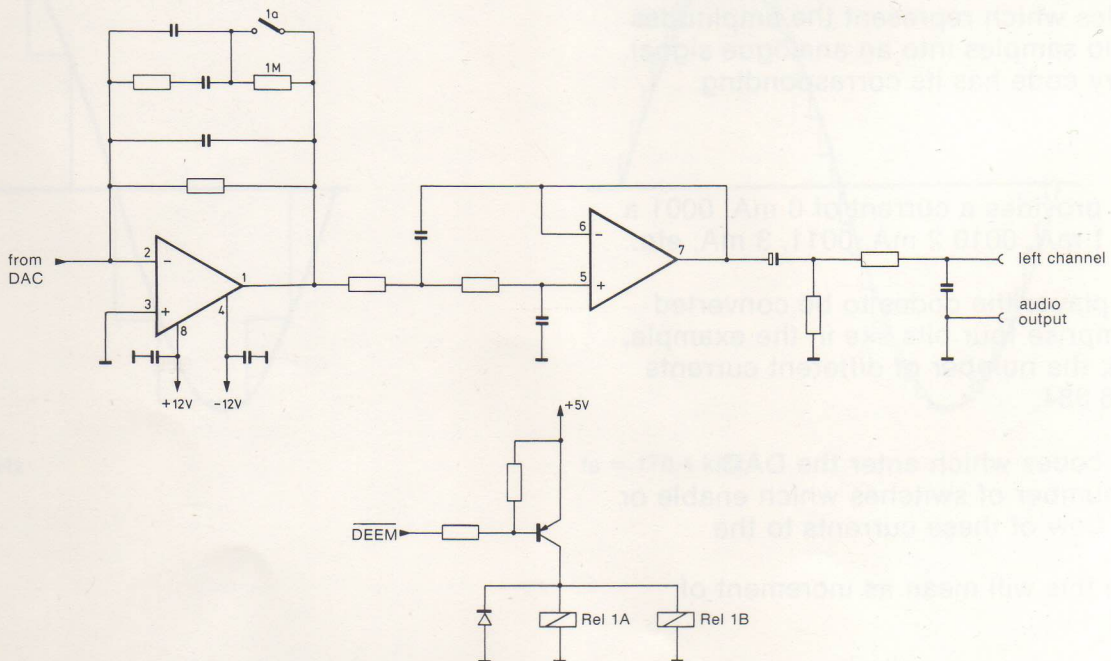
If it is open the $1 \text{ M}\Omega$ resistor provides a discharge path for the capacitor but can be neglected for the calculation of the transfer characteristic of the filter. In the Figure the first part of the filter circuit is shown with and without a de-emphasis network.



without de-emphasis



with de-emphasis



Besides converting an input current I_{in} from the DAC into an output voltage

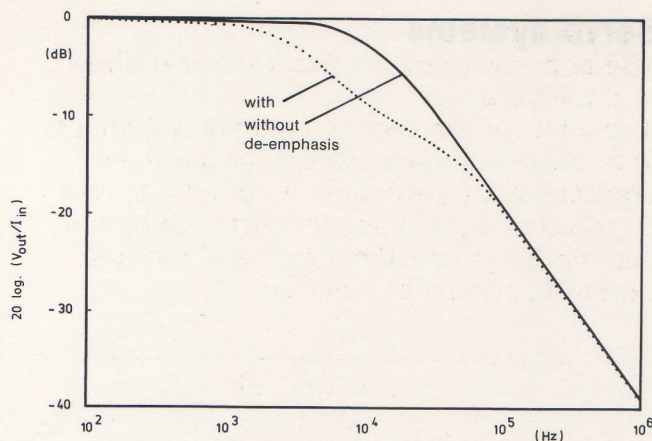
$$V_{out} = -I_{in} \cdot R1$$

the filter section provides a low-pass characteristic with a cut-off frequency of about 40 kHz as shown in the curve. The first filter provides a first-order de-emphasis the frequency response of which is shown as a dotted line.

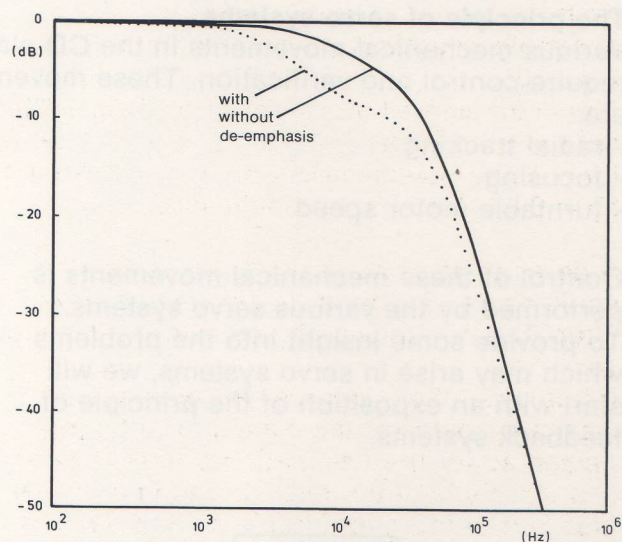
The frequency response of the complete Bessel filter is shown in the other curve. The lowest unwanted frequency in the output spectrum of DAC is 156.4 kHz which is 4 times the sampling frequency of 44.1 kHz minus 20 kHz. At this frequency the filter provides an attenuation of 33.8 dB.

A further 18 dB are added due to the $\frac{\sin x}{x}$

effect of the D/A converter, so that the 176.4 kHz \pm 20 kHz lobe is attenuated by more than 50 dB.



Frequency response of the first filter section



Frequency response of the Bessel filter

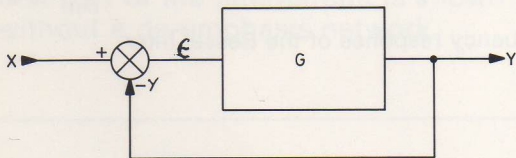
Servo systems

The principle of servo systems

Various mechanical movements in the CD player require control and verification. These movements are:

- radial tracking
- focusing
- turntable motor speed

Control of these mechanical movements is performed by the various servo systems. To provide some insight into the problems which may arise in servo systems, we will start with an exposition of the principle of feedback systems.



X = desired value (reference)
Y = actual value
ε = error signal
G = system properties

$$\epsilon = X - Y$$

$$Y = \epsilon \cdot G$$

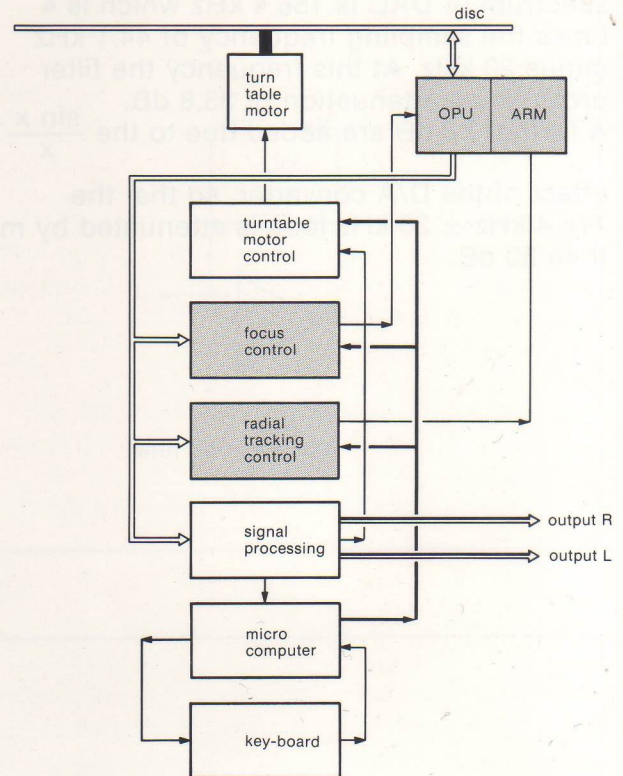
$$\epsilon = X - Y \} Y = (X - Y) G$$

$$(1 + G) Y = X \cdot G$$

$$\frac{Y}{X} = \frac{G}{1 + G}$$

This formula leads to the conclusion that the system will become unstable when $G = -1$,

because $\frac{Y}{X}$ will then be indeterminate.



$G = -1$ is satisfied when $G = 1$, while at the same time a phase shift of 180 degrees is taking place.

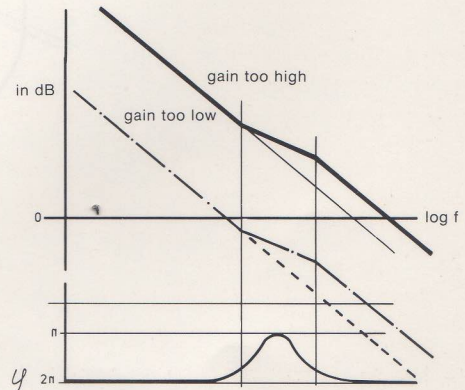
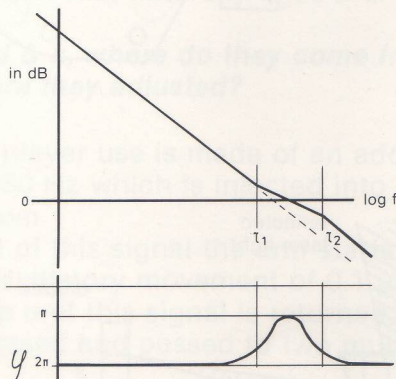
To prevent the system from becoming unstable, care must be taken that no 180° phase shift occurs when $G = 1$.

The transfer function of a system comprising a motor or something reacting like a motor, may be described as a 2nd order function with a roll-off of -12 dB per octave. However, a 2nd order system presents a phase shift of 180 degrees at $G = 1$. If no precautions are taken such a system will consequently become unstable each time the gain becomes 1. For this reason such systems are implemented with a so-called lead network that brings about a 90° phase shift with a roll-off of -6 dB/octave.

The time constants of the lead network must be chosen so, that the 90° phase-shift occurs when $G = 1$.

All the servo systems built into the CD player contain a lead network to keep them stable. The problem is, however, that the gain of the systems depends on so many factors that a change in these factors may result in finding the unity gain ($G = 1$) outside the crossover points of the lead network as shown in the Figure below.

Therefore it is advisable to keep the gain as constant as possible. This can be achieved by controlling the gain as is done for radial tracking (refer to following chapter).



The radial tracking system

The pick-up is attached to an arm that is capable of moving in a horizontal plane. The movement of the arm is based on the moving-coil principle as used in measuring instruments.

By passing more/less current through the coil, the arm can be brought to a defined position.

Information about the position of the unit relative to the track on the disc is obtained from the amount of light falling on the light-sensitive diodes.

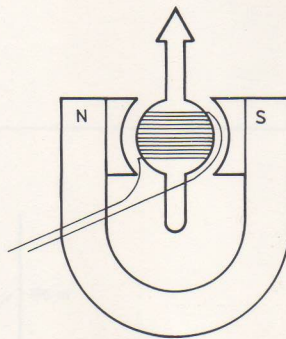
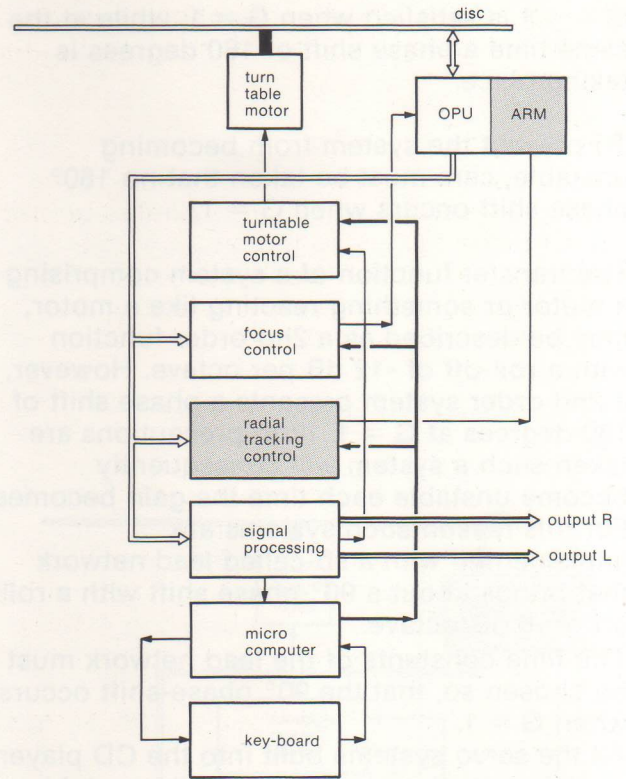
The diodes have been placed apart; in order to make the light fall on the diodes, the side of the prism where the reflected light emerges has been ground to a wedge shape. As a result of the wedge shape the reflected light is split into two equal beams. Each beam falling on a diode.

In reality we have diode pairs, but since for radial tracking the sum of the signals from each pair of diodes is used, we may as well speak of single diodes.

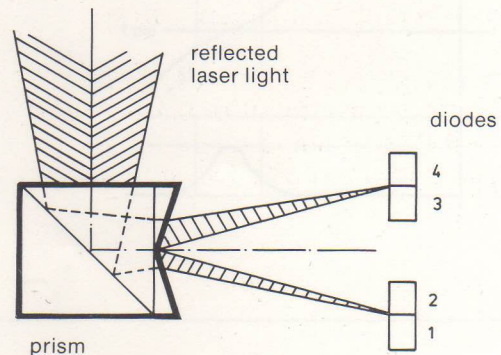
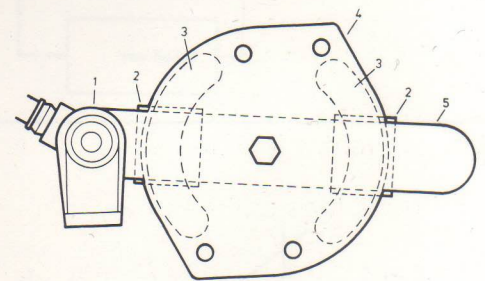
$(i_1 + i_2)$ is the signal from the first pair and $(i_3 + i_4)$ the signal from the second pair.

The radial error (RE) signal consists of the difference between the two signals:

$$RE = (i_3 + i_4) - (i_1 + i_2).$$



- 1 objective + lightpen
- 2 coil
- 3 magnet
- 4 chassis
- 5 arm
- 6 diodes



The signal may, however, be affected by a number of factors that could make the system unstable.

The factors are:

- a possible asymmetry of the spot;
- a possible deviation from the angle the disc must make relative to the pick-up unit;
- the arm's angle relative to the track (90 degrees in the centre of the disc and 45 degrees at the edge of the disc);
- the intensity of the spot;
- the reflectivity of the disc surface.

An asymmetric spot, resulting from a deviation in the angle between disc and pick-up unit, results in the signals ($i_1 + i_2$) and ($i_3 + i_4$) being unequal, even when the pick-up unit is below the centre of the track. This error can be corrected by introducing a factor, called factor d (to be explained shortly).

As a result of the introduction of this factor d the radial error signal will become:

$$RE = 2d (i_1 + i_2 + i_3 + i_4) - 2 (i_1 + i_2).$$

Compensation with factor d is not yet sufficient, for the system's gain is also affected by these factors.

If the laser produces little light, and the disc is reflecting well or badly etc., this may influence the gain by a factor of up to 6. Such a variation of gain can make the system unstable; to prevent this, the gain must be controlled.

The radial error signal will then become:

$$RE = k.d. (i_1 + i_2 + i_3 + i_4) - k (i_1 + i_2),$$

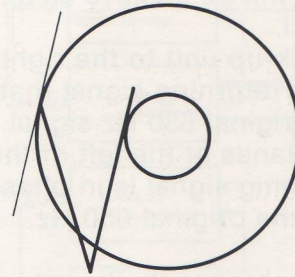
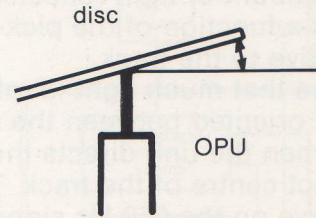
where factor k (also to be explained) and, consequent, the gain can be adjusted.

The questions that naturally arise are:

What are d & k, where do they come from, and how are they adjusted?

In the CD player use is made of an additional signal of 650 Hz which is injected into the radial system.

As a result of this signal the arm starts to make an oscillatory movement of $0.1 \mu\text{m}$. Via the pick-up unit this signal is returned; it is then processed and passed to two multipliers.



Factor d

The curve as in the Figure opposite represents the amount of light reflected back from the disc as a function of the pick-up unit's place relative to the track.

This curve shows that much light is reflected when the unit is oriented between the tracks and little light when the unit directs the laser beam to the exact centre of the track. The effect to this curve on the 650 Hz signal that makes the arm oscillate, is shown in the Figure below.

When the laser light hits the centre of the track, the signal will be positive as a result of the 650 Hz signal.

A shift of the pick-up unit to the right of the track results in a returning signal that is in phase with the original 650 Hz signal.

If the laser light lands at the left of the track centre, the returning signal is in phase opposition with the original 650 Hz.

By detecting synchronously the returning signal, a control signal is obtained that determines the factor d.

The factor d co-determines the radial error signal that in turn sets the arm into movement so that the laser spot is correctly directed to the centre of the track.

Factor k

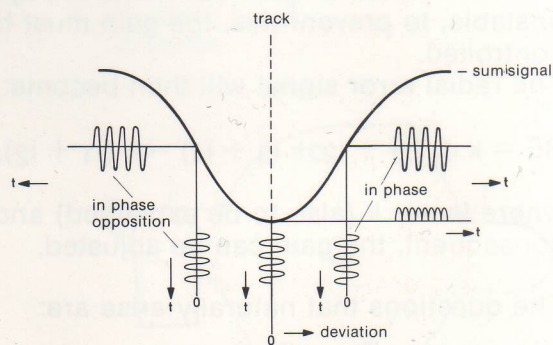
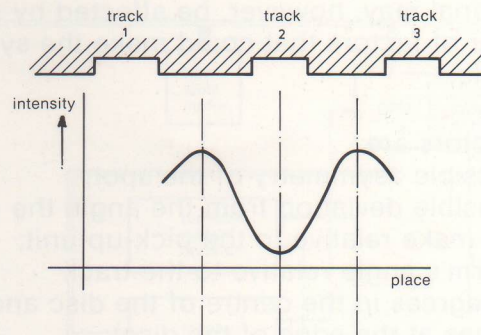
For the adjustment of factor k use is made once again of the 650 Hz signal.

It appears that in the radial tracking system the phase of the 650 Hz signal is strongly influenced by the gain.

If the gain increases or decreases the phase shift between the original signal and the returning signal will also increase or decrease.

Both signals are compared with each other in a phase detector and the resultant signal is proportional to the phase shift and, consequently, to the magnitude of the gain.

This resultant signal is used to adjust factor k, thus maintaining the gain of the entire system as constant as possible.



The focus servo system

In the description of the general block diagram it was said that a position of the disc above or below the objective's focal point will make the focal point of the reflected light land behind or in front of the diodes.

The shape of the prism causes the reflected light to be split up into two equal beams, each beam falling on a diode pair.

If the system is correctly focused, that is, the distance between objective and reflective surface is nominal, the spot lands at the centre between the two diodes of the diode pair.

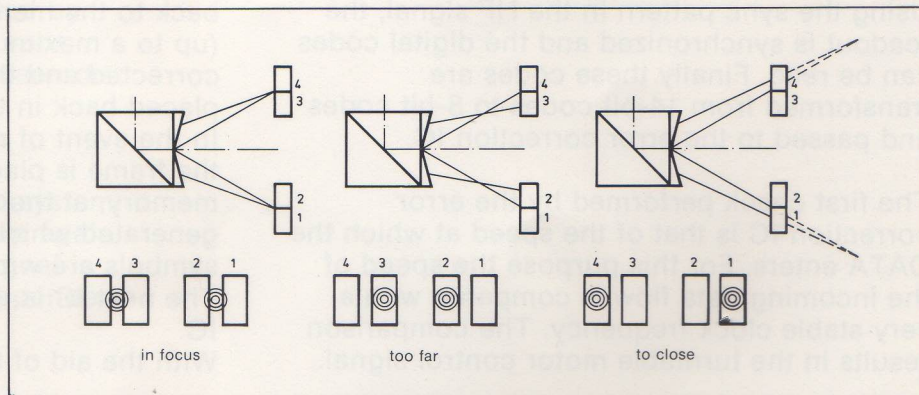
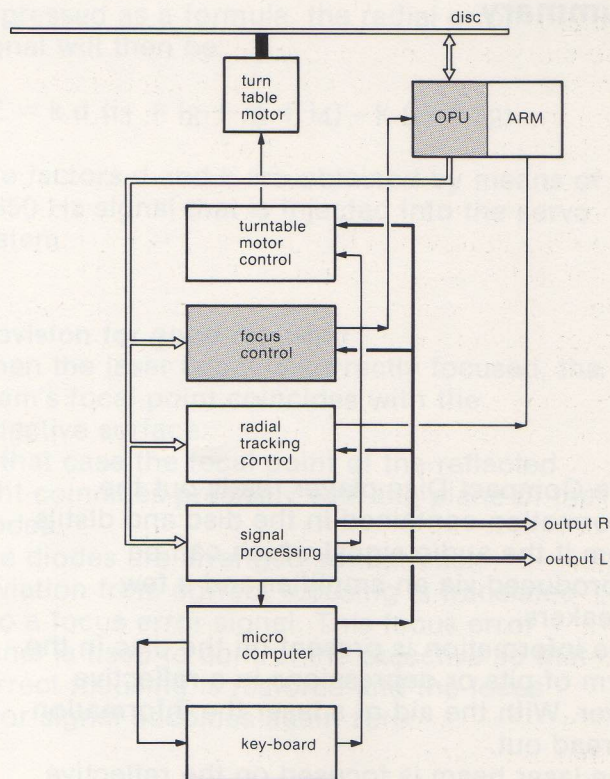
In this case both diodes receive an equal amount of light. If, on the contrary, the distance between objective and disc is too great, so that the focal point of the reflected light moves to a point in front of the diodes, the light spots will land on the diodes 2 and 3. In the opposite case, that is, when the distance is too little, the focal points will be somewhere behind the diodes, and the diodes 1 and 4 will receive the greatest quantity of light. By summing the signals from the diodes 2 and 3 and those from the diodes 1 and 4, and by subtracting the results, the focus error signal is found.

$$FE = (i_1 + i_4) - (i_2 + i_3).$$

From the outcome of this formula the following conclusions can be drawn:

- FE = 0: the system is correctly focused.
- FE < 0: distance between disc and objective too great
- FE > 0: distance between disc and objective too small

The focus error signal (FE) is, via an amplifier, fed to the coil that causes the objective to move up or down until the FE signal is again 0 and the system correctly focused.



Summary

The Compact Disc player reads out the information contained in the disc and distils from it the audio signal which can be reproduced via an amplifier and a few speakers.

The information is present on the disc in the form of pits or depressions in a reflective layer. With the aid of a laser the information is read out.

The laser beam is focused on the reflective layer that reflects back all of the beam. If the laser beam falls on a pit, the intensity of the reflected light will be low. In this way the reflected beam is intensity-modulated by the information on the disc.

This light is via a prism projected on light-sensitive diodes which in turn convert the intensity variations into electric signals.

The signals from the diodes are used for various purposes i.e.:

- to be converted into audio signals;
- for focusing;
- for track following.

Conversion of the diode signals into audio signals

The diode signal (HF signal) enters the demodulator IC.

A PLL circuit allows to process this signal to have a sync signal for information readout at the correct moments of time.

Using the sync pattern in the HF signal, the readout is synchronized and the digital codes can be read. Finally these codes are transformed from 14-bit codes to 8-bit codes and passed to the error correction IC.

The first check performed by the error correction IC is that of the speed at which the DATA enters. For this purpose the speed of the incoming data flow is compared with a very stable clock frequency. The comparison results in the turntable motor control signal.

Next, the 8-bit code words which contain the audio information, are stored in a memory (RAM) before being passed to the error correction decoders. This memory makes it possible to present the codes or symbols to the decoder in the order necessary to accomplish the decoding and error correction process.

In the first decoder the 32 symbols of a frame are split up into 28 information symbols and 4 parity symbols.

The following step is to multiply the parity symbols matrixwise by the information symbols. If no errors are found, the 28 symbols return to the memory.

If not more than one error is found, this error is corrected and the corrected frame is returned to the memory.

If more than one error occurs, the 28 symbols are placed back uncorrected in the memory; however, error signals (flags) are generated which inform the second decoder that these symbols are wrong. Since in this situation a de-interleaving action must be undertaken, the symbols arrive in a different order in the second decoder.

Four out of the 28 symbols presented are parity symbols, the remaining 24 contain audio information.

Again matrixwise multiplications are carried out.

If no errors are found, the 24 symbols go back to the memory. If wrong symbols occur (up to a maximum of two), these are corrected and the corrected frame is then placed back in the memory.

In the event of more than two wrong symbols, the frame is placed back unchanged in the memory; at the same time error signals are generated which instruct the next IC which symbols are wrong.

The next IC is the interpolating and muting IC.

With the aid of the flag from the error

correction IC the interpolating and muting IC is capable of determining whether an erroneous sample can be interpolated or should be muted.

In cases where the values of the preceding and following samples are known, linear interpolation makes it possible to compute a value for the missing sample. If one of these sample values is absent, muting is applied. A further task of the interpolating and muting IC is to separate the information for the right and left channels so that two outputs are available.

Each channel signal is then fed to a digital filter that sees to it that the audio frequencies only are let through to the digital-to-analogue converter (DAC).

The DAC generates an analogue current which corresponds with the binary word at the input. This current changes each time a different binary code appears at the input. In this way the original analogue audio signal is reconstructed from the binary codes.

Finally, an analogue filter is used to further suppress any interfering frequencies.

Radial tracking

When the laser beam is incorrectly centered on the track, the intensity of one half of the reflected beam will be lower than that of the other beam. The split up of the reflected beam into two equal parts and the falling of each half on a separate diode makes it possible to detect whether the track is followed correctly.

If an equal amount of light falls on either diode, we know that the track is correctly followed. If the light beam is incorrectly centered on the information track, the light on one diode will be less than on the other diode.

By taking up the arm that bears the pick-up unit into a servo system it is possible to control the arm in a way that maintains equal amounts of light falling on each diode.

The spot intensities are also affected by factors such as: asymmetry of the spot, deviation from the ideal angle between disc and pick-up unit.

The introduction of factor d allows compensation for these influences.

To ensure the stability of the servo system, it is also necessary to control the loop again.

For this purpose the radial error signal is multiplied by a factor k which is dependent on the gain.

Expressed as a formula, the radial error signal will then be:

$$RE = k \cdot d (i_1 + i_2 + i_3 + i_4) - k (i_1 + i_2).$$

The factors d and k are obtained by means of a 650 Hz signal that is injected into the servo system.

Provision for good focusing

When the laser beam is correctly focused, the beam's focal point coincides with the reflective surface.

In that case the focal point of the reflected light coincides precisely with the plane of the diodes.

The diodes are arranged so that each deviation from correct focusing is translated into a focus error signal. This focus error signal is used to control the objective so that correct focusing is restored and the focus error signal becomes again zero.

Appendix A

2. Give two reasons for introducing margin bits?

4. Which encoding procedure, used in the Compact Disc, meets the modulation system's requirements?

3. What are the consequences of the light spot's finite dimensions?

2. In what way is regeneration of the clock frequency achieved mostly?

1. Why?

Modulation

1. How does the frame look, the way it is recorded on the disc?
2. What are the consequences of scrambling on the location of the symbols on the disc?
3. What is meant by scrambling?
4. What is meant by interspersal in the Compact Disc field?
5. What is the effect of interspersal on the error encoding system?
6. How is interspersal brought about?
7. What is the result of interspersal with regard to the symbols on the disc?

Scrambling and interspersal

encoding system?

2. How many parity symbols are added to the audio symbols by the

frames?

3. What formulae apply for error detection and correction of 16 errors?

4. In how many positions must valid code words differ to enable

2. What is meant by the minimum distance (d_{min})?

detection of 16 errors?

3. What formulae apply for error detection and correction of 16 errors?

Error detection and correction principles

2. What is a synchronization pattern and what is it needed for?

3. Where are the necessary sections for error correction performed?

4. What takes place in the ADC?

5. How is quantization noise created?

6. What happens in the Sample-and-Hold circuit?

7. Why is a sharp cut-off low-pass filter required?

System basics

Compact Disc encoding

Questions

System basics

1. *Why is a sharp cut-off low-pass filter required?*
2. *What happens in the Sample-and-Hold circuit?*
3. *How is quantization noise created?*
4. *What takes place in the ADC?*
5. *Where are the necessary actions for error correction performed?*
6. *What is a synchronization pattern and what is it needed for?*

Error detection and correction principles

1. *What are redundancies and what are they needed for?*
2. *In how many positions must valid code words differ to enable detection of t_1 errors?*
3. *What is meant by the minimum distance (d_m)?*
4. *In how many positions must valid code words differ to enable correction of t_2 errors?*
5. *What formulas apply for error detection and correction of symbols in frames?*
6. *How many parity symbols are added to the audio symbols by the encoding system?*

Scrambling and interleaving

1. *What is the result of interleaving with regard to the symbols on the disc?*
2. *How is interleaving brought about?*
3. *What is the effect of interleaving on the error encoding system?*
4. *What is meant by interpolation in the Compact Disc field?*
5. *What is meant by scrambling?*
6. *What are the consequences of scrambling on the locations of the symbols on the disc?*
7. *How does the frame look, the way it is recorded on the disc?*

Modulation

1. *What requirements must the modulation system for the Compact Disc fulfil?*
2. *In what way is regeneration of the clock frequency achieved mostly?*
3. *What are the consequences of the light spot's finite dimensions?*
4. *Which encoding procedure, used in the Compact Disc, meets the modulation system's requirements?*
5. *Give two reasons for introducing merging bits?*

Compact Disc encoding



Introduction

This Appendix is dealing with the way in which the analogue audio signals are processed prior to recording on the disc. Ample attention will be given to the principles of error correction so as to enhance the understanding of the error correction IC.

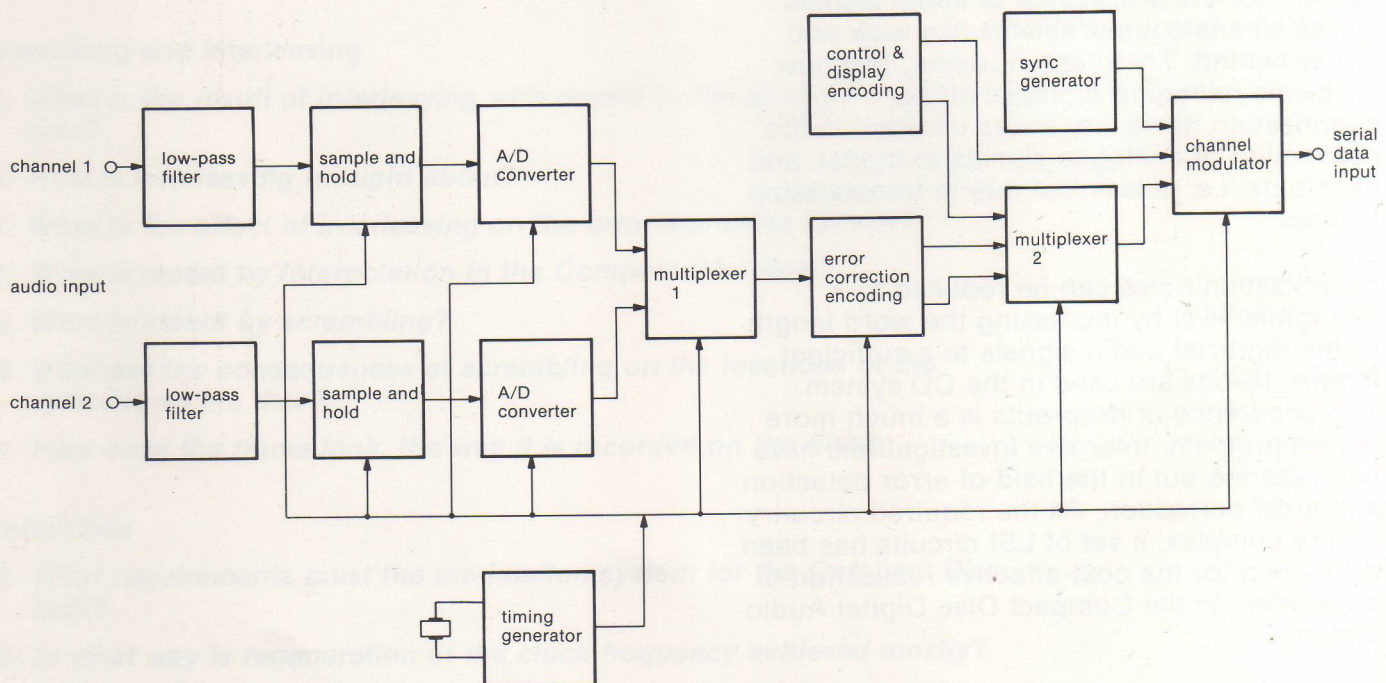
The Compact Disc (CD) is a storage medium for the reproduction of highest-quality audio signals. Sound is digitized and recorded on a disc as groups of binary numbers represented as pits and flat areas along a helical track.

The change from the analogue to the digital domain for the processing of audio signals leaves all analogue problems like wow and flutter behind. There are, however, two new problems related to digitized audio: quantization noise, i.e. errors inherent in the conversion of analogue signals to digital, and drop-outs, i.e. pulses lost due to transmission failures.

Quantization noise can be reduced to a negligible level by increasing the word length of the digitized audio signals to a sufficient length, 16-bits are used in the CD system. The occurrence of drop-outs is a much more severe problem. Intensive investigations have been carried out in the field of error detection and error correction. As the required circuitry is very complex, a set of LSI circuits has been developed for the cost-effective realisation of a decoder for the Compact Disc Digital Audio system.

System basics

The CD carries digitized audio information for a playing time of up to one hour in stereo. For the production of the disc, the audio signals have to be digitized and encoded. The block diagram of this encoding system is shown in the Figure.



The low-pass filters

Before any other processing takes place, the analogue input signals are passed through a sharp cut-off low-pass filter in order to limit their bandwidth to a maximum frequency f_m equal to or less than one-half the sampling frequency of $f_s = 44.1$ kHz. This is to ensure conformity with the Nyquist theorem.

Otherwise, intermodulation distortion could occur due to frequency fold-over.

In an impulse-sampled system, the frequency spectrum simply is the audio signal spectrum repeated periodically with a scaling factor of f_s .

Sample-And-Hold

Before the stereo audio signal is recorded on the disc, it has to be converted into a digital signal.

One of the first conversion steps is to sample the analogue audio signal at fixed intervals or time points.

At each of these time points (also referred to as sampling times) the signal's amplitude is measured and the measured value is held for a moment to enable its processing, like - for instance - the conversion of the measured value into a binary code.

This way of sampling is referred to as Sample-And-Hold technique.

The procedure of this Sample-And-Hold technique is very much similar to the charge/discharge procedure of a capacitor via a switch.

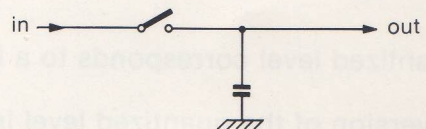
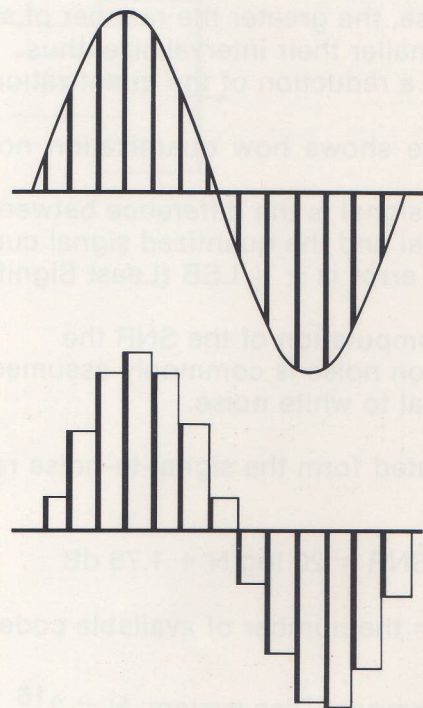
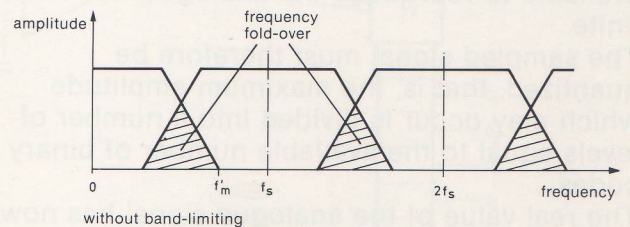
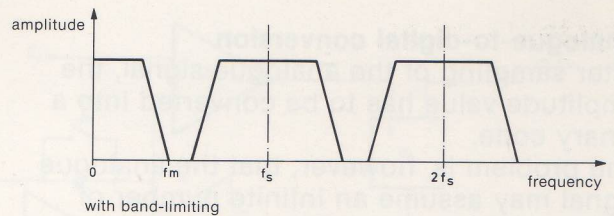
The switch is actuated by a pulse of a relatively short duration. This pulse is named 'sampling pulse'.

When the analogue signal comes in at the Input and the switch is closed by the sampling pulse, the capacitor will be charged to the signal's amplitude at the moment (= instantaneous value).

If the switch is then re-opened, the capacitor will retain the instantaneous value until the switch is closed again. In this new situation the capacitor will take over the signal's present instantaneous value.

The maximum frequency that can be sampled in this way is one-half the frequency of sampling.

In the CD system the sampling frequency is 44.1 kHz and, consequently, amply sufficient for a signal frequency of 20 kHz.



Analogue-to-digital conversion

After sampling of the analogue signal, the amplitude value has to be converted into a binary code.

The problem is, however, that the analogue signal may assume an infinite number of levels, whereas the number of binary codes available to reproduce the analogue levels is finite.

The sampled signal must therefore be quantized, that is, the maximum amplitude which may occur is divided into a number of levels equal to the available number of binary codes.

The real value of the analogue signal has now to be rounded to a quantized value that comes closest to the analogue value.

The difference between the signal's real value and the quantized signal is experienced as noise and is referred to as quantization noise. The greater the number of codes available for the purpose, the greater the number of steps and the smaller their interval size, thus leading to a reduction of the quantization noise.

The Figure shows how quantization noise is created.

The error signal is the difference between the input signal and the quantized signal curve. Maximum error is $\pm \frac{1}{2}$ LSB (Least Significant Bit).

For the computation of the SNR the quantization noise is commonly assumed to be identical to white noise.

In formulated form the signal-to-noise ratio (SNR) is:

$$\text{SNR} = 20 \log N + 1.76 \text{ dB}$$

where N = the number of available codes.

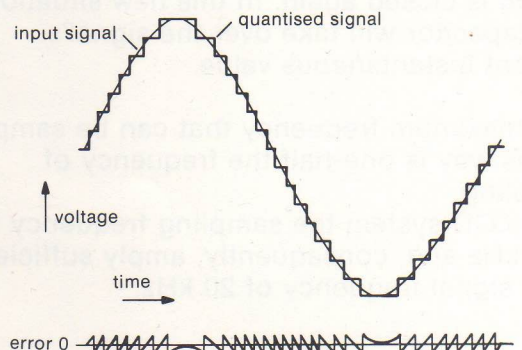
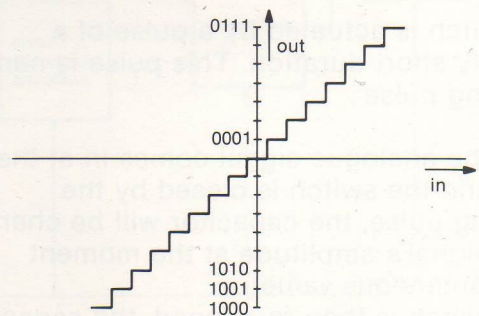
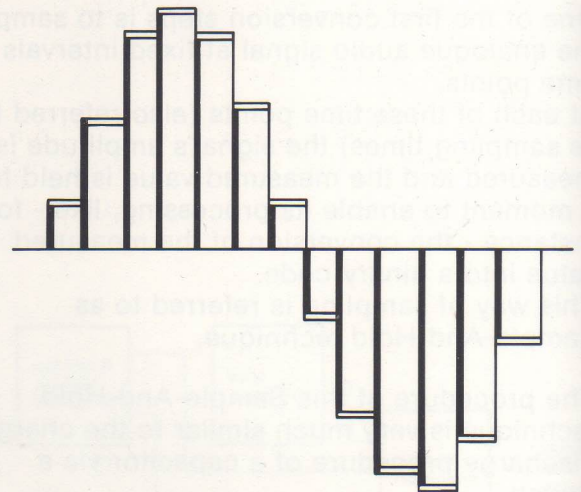
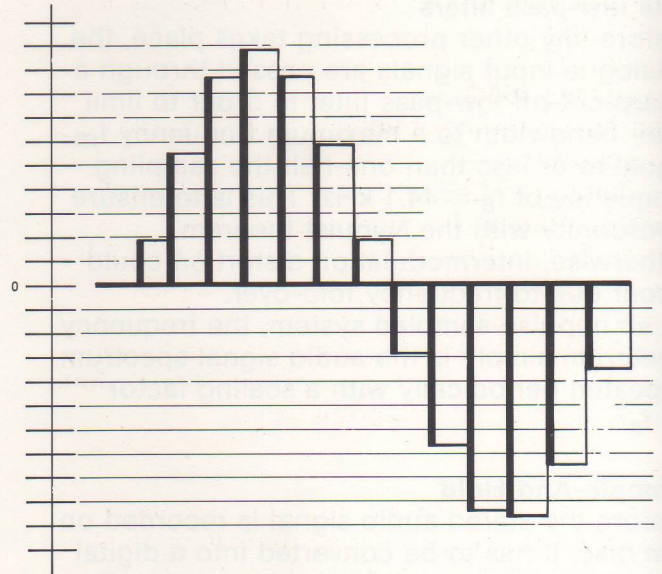
For the Compact Disc system: $N = 2^{16}$, since a code consists of 16 binary digits or 16 bits.

Thus dividing the amplitude into 65536 equal intervals gives

$$\text{SNR} = 20 \log 2^{16} + 1.76 = 98 \text{ dB}.$$

Each quantized level corresponds to a binary code.

The conversion of the quantized level into a binary code is carried out in an analogue-to-digital converter (ADC).



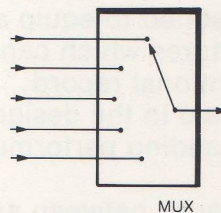
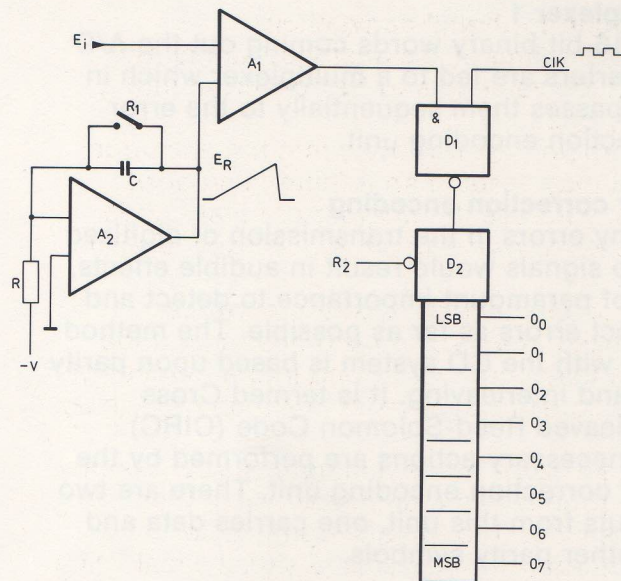
The principle of operation of an ADC is as following:

One-slope ADC

The analogue input voltage E_i is presented to comparator A1. A1 compares E_i with E_r (E_r is the sawtooth voltage coming from integrator (A2)). When E_i and E_r are equal, the output of A1 switches over and blocks the clock pulses Clk via NAND gate D1. The binary counter D2 stops and the binary content of this counter represents the digital conversion of an analogue value.

Multiplexer

Sampling, quantization and conversion to a binary code are operations performed separately for the right and left channels. However, both codes should ultimately be stored on the disc, one after the other in a single track. For this, use is made of a multiplexer; the multiplexer is a digital circuit that first passes on the information of one channel and then the information of the second channel, etc.



Multiplexer 1

The 16-bit binary words coming out the A/D converters are fed to a multiplexer which in turn passes them sequentially to the error correction encoding unit.

Error correction encoding

As any errors in the transmission of digitized audio signals would result in audible effects, it is of paramount importance to detect and correct errors as far as possible. The method used with the CD system is based upon parity bits and interleaving. It is termed Cross Interleaved Reed-Solomon Code (CIRC).

The necessary actions are performed by the error correction encoding unit. There are two outputs from this unit, one carries data and the other parity symbols.

Control and Display encoding

As the CD offers a new dimension in sound quality, it seems to be justified to equip a CD player with operating features which cannot easily be found in conventional record players. This will further aid in the design of CD players with an outstanding performance.

An identification of the pause between any two successive pieces of music can be used for the implementation of search and repeat functions, an identification whether the recording was made with pre-emphasis or not could be used for switching the de-emphasis in the player on and off automatically, recording of timing information could be used for the display of elapsed time or of the time of play or a piece of music, to mention just a few potential features.

As this control and display information is non-audible, it has to be encoded separately. This is performed by the control and display encoding unit which outputs 8-bit wide symbols, permitting the implementation of eight different information channels two of which have been specified so far.

The control and display information is sometimes referred to as subcoding information.

As the subcoding information is generated and recorded bit-serially (seen per channel), there must be some provision for the recognition of the beginning of a string of bits.

There are two outputs from this encoding unit, one carries data and the other is used for the synchronization of the subcoding data blocks.

Multiplexer 2

The 8-bit symbols from the error correction encoding unit which are either data or parity symbols and from the control and display encoding unit are passed through a second multiplexer which in turn feeds them in a certain sequence to the channel modulator.

Sync generation

As the data are organized in blocks, some means for recognizing the beginning of these blocks must be provided. The sync generator creates a unique pattern which is not contained in the normal data. It is passed to the channel modulator which adds it to its output data on request from the timing unit.

Channel modulator

The output signals of the error correction encoding and control and display encoding units are both represented in a non-return-to-zero (NRZ) format.

These NRZ data may not be recorded on the disc.

They have to be changed to another code, an action which is performed by the channel modulator.

Timing

As we have seen, many different tasks have to be carried out for the encoding of a CD.

These tasks are related to each other and therefore the timing unit supplies control information to most of the functional blocks in order to ensure proper operation.

The stability of the timing generator is based on a crystal oscillator.

Error detection and correction principles

While playing a disc, disturbances may occur which cause distortion of the audio information. These disturbances manifest themselves in the form of a tick. They may be the result of scratches, dust or dirt on the disc.

Analogue systems barely (if at all) allow correction of these disturbances.

Digital systems, however, offer the possibility of correction in a relatively simple way.

To enable correction it is necessary to take certain measures prior to recording the codes on the disc.

Single error detection

To create the possibility of detection and/or correction of errors, redundancies must be introduced.

When a code word e.g. consists of 4-bits, there are $2^4 = 16$ possible code word values. If all these code word values are used for information transmission, no value is left to detect whether the code word received is containing erroneous bits.

The addition of one bit, a so called parity bit, to the 4-bit words, allows to discover the occurrence of a single error in the code word.

Suppose we are in an even parity scheme. This implies that, in case the number of 1s in the 4-bit word is even, the parity bit added is 0 (the parity bit will be 1 when the number of 1s is odd). Thus, 5-bit words with an even number of 1s are created. Only words with even parity are valid code words.

If - at the receiver side - a word containing an odd number of 1s is received, there must have slipped in an error, because the total number of 1s (including the parity bit) should be even.

Example

4-bit word A A: 1100
 4321

4-bit word B B: 1000
 4321

The number of 1s in word A is even. Using even parity, the parity bit to be added to this word is 0.

A': 11000
 43210

So, A' is a 5-bit word.

The number of ones of word B is odd. If a parity bit is added to word B, it will be 1.

B': 10001
 43210

A' and B' have each an even number of ones and have been derived from the code words A and B.

Multiple error detection

To have the possibility of detecting more than one error in a code word, the addition of one single parity bit will not suffice to allow unambiguous detection, for in case two zeros in code word B have flipped to two ones, B' will still contain an even number of ones.

What will happen when e.g. two errors are introduced in a given code word A ?

A 000111 → A' 010101

In that case code word A' will differ from A in two positions.

A 000111
A' 010101

Each code word A'_n with two errors relative to A will differ from A in two positions. All code words A'_n are wrong code words and are therefore not allowed to belong to the valid code words.

0000	0
0001	1
0010	1
0011	0
0100	1
0101	0
0110	0
0111	1
1000	1
1001	0
1010	0
1011	1
1100	0
1101	1
1110	1
1111	0
parity symbol	

Which criteria should be met by valid code words to ensure unambiguous detection when two errors have slipped into these words?

When two errors occur in A, the received code word A'_n differs from A in two positions.

When all valid code words differ from A in three positions, a code word A'_n will never equal one of the valid code words. If all valid code numbers differ from one another in three positions, it will in no case be possible to obtain a valid code word after the introduction of two errors in a given code word.

What applies to two errors will also apply to three errors, on the condition however that the valid code words differ in four positions.

In general applies:

To be capable of detecting t_1 errors in a code word, all of the admitted code words should differ in t_1+1 positions.

The code words A' and B' of the example of single error detection comply with this rule for detecting one error, because differing from one another in two positions, viz. bit 0 and bit 3.

If, during transmission, three errors occur in code words which differ in four positions a decoder should be capable of determining unambiguously that the received code word became corrupted, without mistaking it for another code word.

Example

If, during transmission, three errors are introduced in code word A, the received code word A' should never be identical to B, C or D, because A, B, C and D differ in four positions.

Code word A: 000111 B: 011001
C: 101010 D: 110100

All possible code words A'_n containing 3 errors relative to A, differ from A in three positions.

All these code words A'_n differ both from A and from B, C and D; thus, none of the code words A'_n represents a valid code word.

A: 00 01 11
B: 01 10 01
C: 10 10 10
D: 11 01 00

A: 00 01 11 B: 01 10 01
B: 01 10 01 C: 10 10 10

A: 00 01 11 B: 01 10 01
C: 10 10 10 D: 11 01 00

A: 00 01 11 C: 10 10 10
D: 11 01 00 D: 11 01 00

A, B, C and D differ at least in 4 positions.

A: 00 01 11 $d_m \geq 4$
B: 01 10 01
C: 10 10 10
D: 11 01 00

A'_1 : 11 11 11
 A'_2 : 11 00 11
 A'_3 : 11 01 01
 A'_4 : 11 01 10
 A'_5 : 10 10 11
 A'_6 : 01 10 11
 A'_7 : 00 10 01
 A'_8 : 00 10 10
 A'_9 : 00 11 00
 A'_{10} : 00 00 00
 A'_{11} : 01 01 00
 A'_{12} : 10 01 00
 A'_{13} : 10 11 01
 A'_{14} : 10 00 01
 A'_{15} : 10 11 10
 A'_{16} : 10 00 10
 A'_{17} : 01 11 01
 A'_{18} : 01 00 01
 A'_{19} : 01 11 10
 A'_{20} : 01 00 10

All possible code words A'_n

If a maximum of three errors is introduced in a code word and the minimum distance between the valid code words is greater than or equal to four, a decoder is capable of determining unambiguously by means of comparison whether or not a received code word is wrong.
To the code words A, B, C and D applies that $d_m = 4$.

$$d_m = t_1 + 1$$

By the minimum distance (d_m) for a number of code words is meant: the minimum number of positions where all valid code words differ from each other.

Error correction

There exists also a possibility to correct errors. This requires, however, that the code words differ from each other in more positions than required for error detection.

To be capable of correcting an error the error-correction circuit should know which code words are correct ones.
The circuit is containing, as it were, a list of correct code words.
Each time a code word enters, it is compared with those contained in the list. If the received word does not figure in the list, the code must be wrong.
After having found that the received code word is wrong, a further comparison is performed to see in how many positions the received code is differing from the correct code words.
The correct code word that differs from the received one in the lowest number of positions is considered to be the word that was transmitted.

The requisite for a proper correction is that a wrong code is unambiguously related, that is, it belongs to one single correct code.
Wrong code A' of correct code A is not allowed to be equal to wrong code B' of correct code B. The wrong code A' and B' must differ from each other in at least one position.
When A' and B' are created after the introduction of 1 error in A and B, A' and B' differ in one position from A and B respectively.
To create the possibility of correction, A and B must differ from each other in three position, for A' will then differ from B in two positions, whereas B' is differing from B in one position only.

A :	00 00 00
A1' :	00 00 01
A2' :	00 00 10
A3' :	00 01 00
A4' :	00 10 00
A5' :	01 00 00
A6' :	10 00 00
B :	00 11 01
B1' :	00 11 00
B2' :	00 11 10
B3' :	00 10 01
B4' :	00 01 01
B5' :	01 11 01
B6' :	10 11 01

If A' is differing from B at least in two positions and B' is differing from B in one position, A' and B' will differ from each other in at least one position.

In the Figure A1', A3' and A4' differ from B in two positions.
 While: A1' differs in one position from B3' and B4'
 A3' differs in one position from B1' and B4'
 and A4' differs in one position from B1' and B3'.

In general applies:
 To be capable of correcting t2 errors, the minimum distance should comply with:

$$d_m = 2t_2 + 1$$

To correct one error dm should at least be 3.

Example

A 6-bit code A (00 00 00) is transmitted; A is one of the valid 6-bit codes with a minimum distance of 3. Due to some disturbance, the following code appears at the receiver side:

A': 00 10 00

If we compare this code with D and H we find that they differ from each other in three positions;
 when compared with C and G in four positions;
 when compared with B and F we find a difference in two positions
 when compared with E we find a difference in five positions.

However, when compared with A: 00 00 00, the code received differs in one position only. A must therefore be the code that has been transmitted. So the erroneous code must be replaced by code A, which corrects the error.

A:	00 00 00	
B:	00 11 01	
C:	01 01 10	
D:	01 10 11	
E:	10 01 11	
F:	10 10 10	$d_m \geq 3$
G:	11 00 01	
H:	11 11 00	

A:	00 00 00	
A':	00 10 00	
A:	00 00 00	$d = 1$
A':	00 10 00	
B:	00 11 01	$d = 2$
A:	00 10 00	
C:	01 01 10	$d = 4$
A:	00 10 00	
D:	01 10 11	$d = 3$
A:	00 10 00	
E:	10 01 11	$d = 5$
A:	00 10 00	
F:	10 10 10	$d = 2$
A:	00 10 00	
G:	11 00 01	$d = 4$
A:	00 10 00	
H:	11 11 00	$d = 3$
A:	00 10 00	

(d = difference)

If the valid code words only differ in two positions and an error is introduced during transmission, it is possible to detect this error.

The received code word will then differ from the transmitted code word in one position, but there will be several code words that differ from the received word in one position only.

The following code words differ from each other in at least two positions ($d_m \geq 2$).

If code word C is the word that is transmitted, but - due to a disturbance - received with one erroneous bit, e.g. \hat{C} : 1101, it is quite easy to detect that \hat{C} is an incorrect code; however, it is not possible to establish unambiguously which one of the bits is wrong, for \hat{C} differs in one bit position from code words C, E, G and H.

If \hat{C} is received, the decoder is not able to determine whether the correct word should be C, E, G or H.

Conclusion, correcting one erroneous bit in code words with a minimum distance $d_m \geq 2$ is not possible.

A:	00 00	
B:	00 11	
C:	01 01	
D:	01 10	
E:	10 01	
F:	10 10	$d_m \geq 2$
G:	11 00	
H:	11 11	
C:	01 01	
\hat{C} :	11 01	
A:	00 00	$d = 3$
\hat{C} :	11 01	
B:	00 11	$d = 3$
\hat{C} :	11 01	
C:	01 01	$d = 1$
\hat{C} :	11 01	
D:	01 10	$d = 3$
\hat{C} :	11 01	
E:	10 01	$d = 1$
\hat{C} :	11 01	
F:	10 10	$d = 3$
\hat{C} :	11 01	
G:	11 00	$d = 1$
\hat{C} :	11 01	
H:	11 11	$d = 1$
\hat{C} :	11 01	

Detection and correction of symbols in frames

In the preceding chapters we learned that it is possible to detect and correct errors in a binary word consisting of a number of bits.

To allow detection the following formula applies:

$$d_m = t_1 + 1$$

To enable correction the formula to be applied is:

$$d_m = 2t_2 + 1$$

In the Compact Disc system, however, the samples are converted into 16-bit words. Each 16-bit word is split up into two symbols of 8-bits prior to being passed on to the encoders.

In the encoder 24 of such symbols are combined in one frame.

Since in the Compact Disc system frames are formed by symbols, it is quite logical to examine the possibility of detecting and correcting wrong symbols in a frame, instead of looking at the different bits and detecting or correcting wrong bits.

This possibility exists.

In order to detect t_1 wrong symbols in one frame, in all valid frames at least $t_1 + 1$ symbols must differ.

So: $d_{ds} = t_1 + 1$
(d_{ds} = distance to detect wrong symbols)

To correct t_2 wrong symbols, the frames should differ in at least $2t_2 + 1$ positions,

so: $d_{cs} = 2t_2 + 1$
(d_{cs} = distance to correct wrong symbols)

The same equations that were derived in the previous sections appear to have their validity in this case also.

To comply with these equations it suffices to add parity symbols.

In the Compact Disc system, every batch of 24 audio symbols is complemented with 8 parity symbols in the encoder, thus bringing the number of symbols in 1 frame to 32.

In fact, the error-correction symbols are added to the audio symbols in two distinct encoders; the first adds 4 parity symbols which are related to 24 symbols, whereas the second encoder adds 4 parity symbols which

are related to the 28 symbols coming from the first encoder.

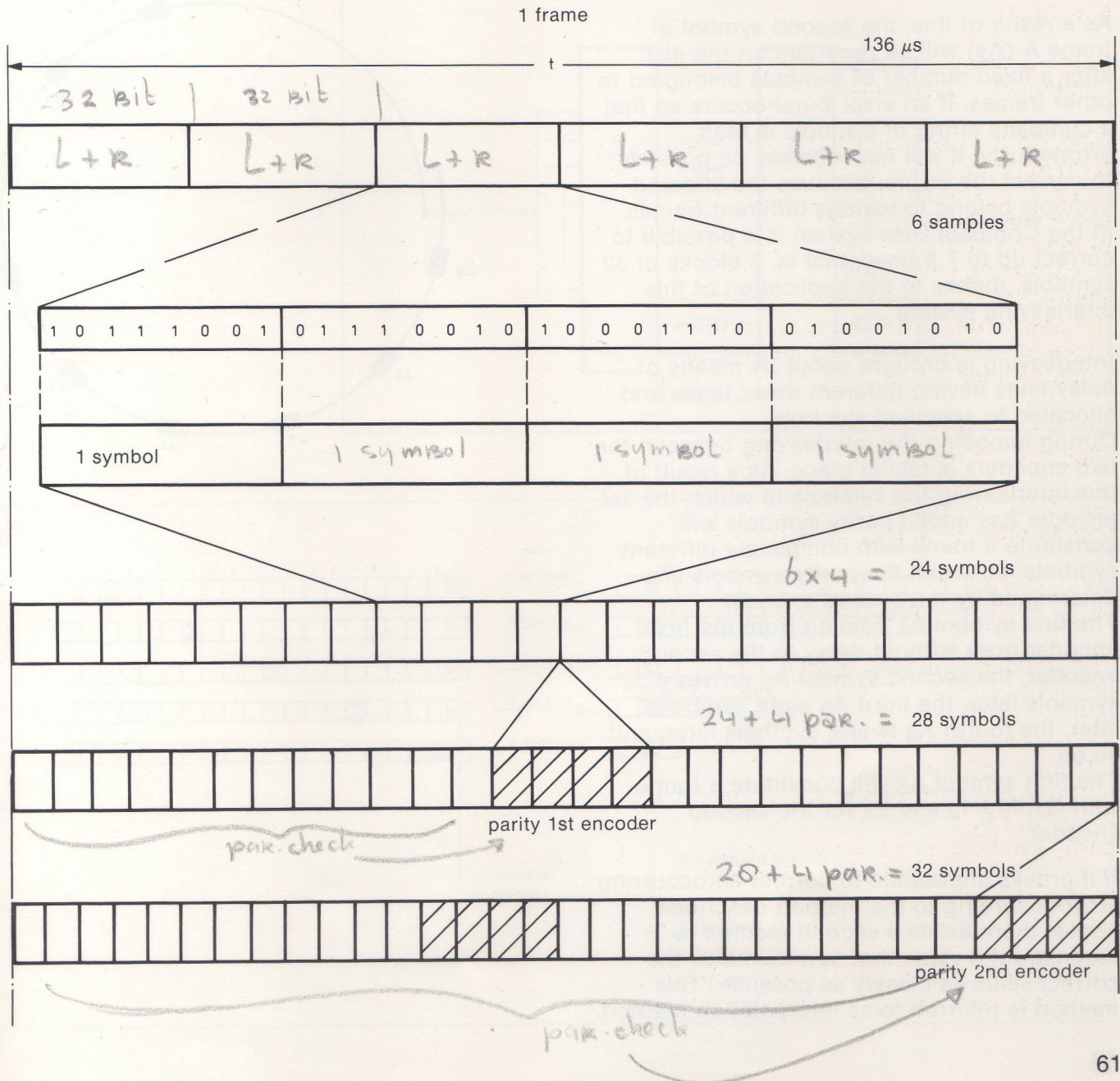
The minimum distance for the frames will thus be: $d_{ms} \geq 5$ (both for the 24 symbols and the 28 symbols).

(d_{ms} = minimum distance for symbols)

In the Compact Disc player's error-correction IC the parity symbols are placed in a matrix and the information of a frame is multiplied by this matrix.

The result of these multiplications consists of four equations, called syndromes.

These equations make it possible to detect and correct errors.



Scrambling and interleaving

The majority of errors which may occur during playback of a disc will result from scratches, dust and dirt.

Consequently, each time such a defect occurs a number of adjacent symbols on the disc will be read erroneously.

If all of the affected symbols belonged to the same frame, a great many multi-position errors would occur inside each word and, consequently, correction would be out of the question.

This situation can be avoided by giving up grouped recording on the disc of the symbols belonging to the same code word and by adopting a spread recording by weaving them according to a fixed pattern with symbols from other code words; this process is referred to as interleaving of symbols.

As a result of this, the second symbol of frame A (A_2) will be recorded on the disc after a fixed number of symbols belonging to other frames. If an error burst occurs so that a complete string of symbols is read erroneously, it will nevertheless be possible to correct the errors, because the affected symbols belong to various different frames. In the Compact Disc system it is possible to correct up to 7 frames, that is, 7 blocks of 32 symbols, thanks to the application of this interleaving system.

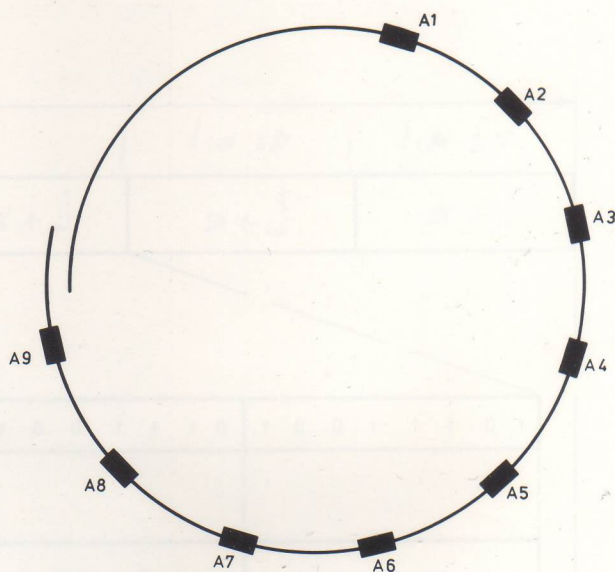
Interleaving is brought about by means of delay lines having different delay times and allocated to specified symbols.

During encoding the interleaving between the two encoders is taking place. As a result of this interleaving the symbols to which the 1st encoder has added parity symbols will constitute a frame with completely different symbols for which the parity symbols are determined by the second encoder.

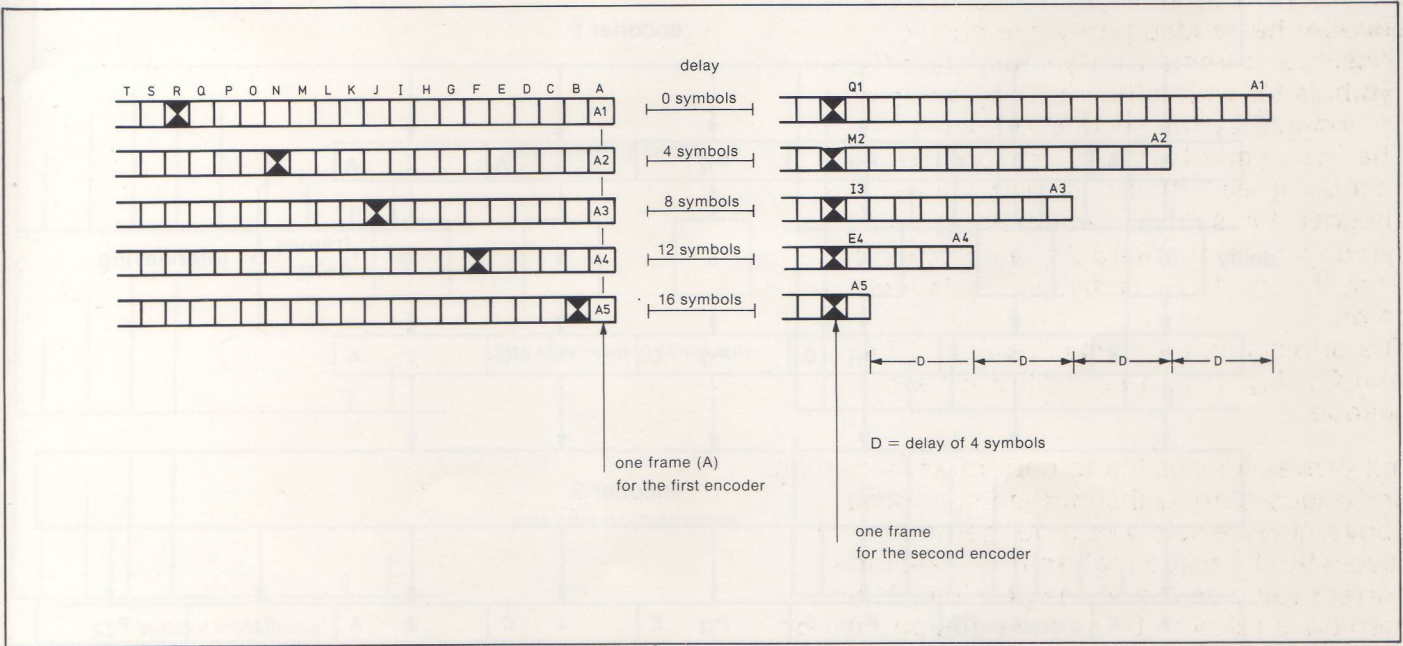
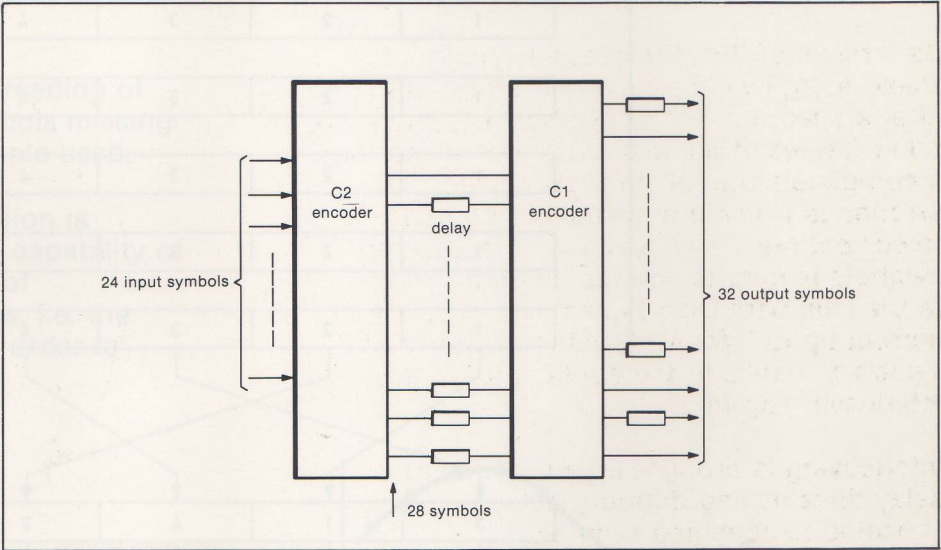
The first symbol A_1 coming from the first encoder goes without delay to the second encoder; the second symbol A_2 arrives 4 symbols later, the third A_3 eight symbols later, the fourth A_4 twelve symbols later, and so on.

The fifth symbol A_5 will constitute a frame with Q_1 , M_2 , I_3 and E_4 for the second encoder.

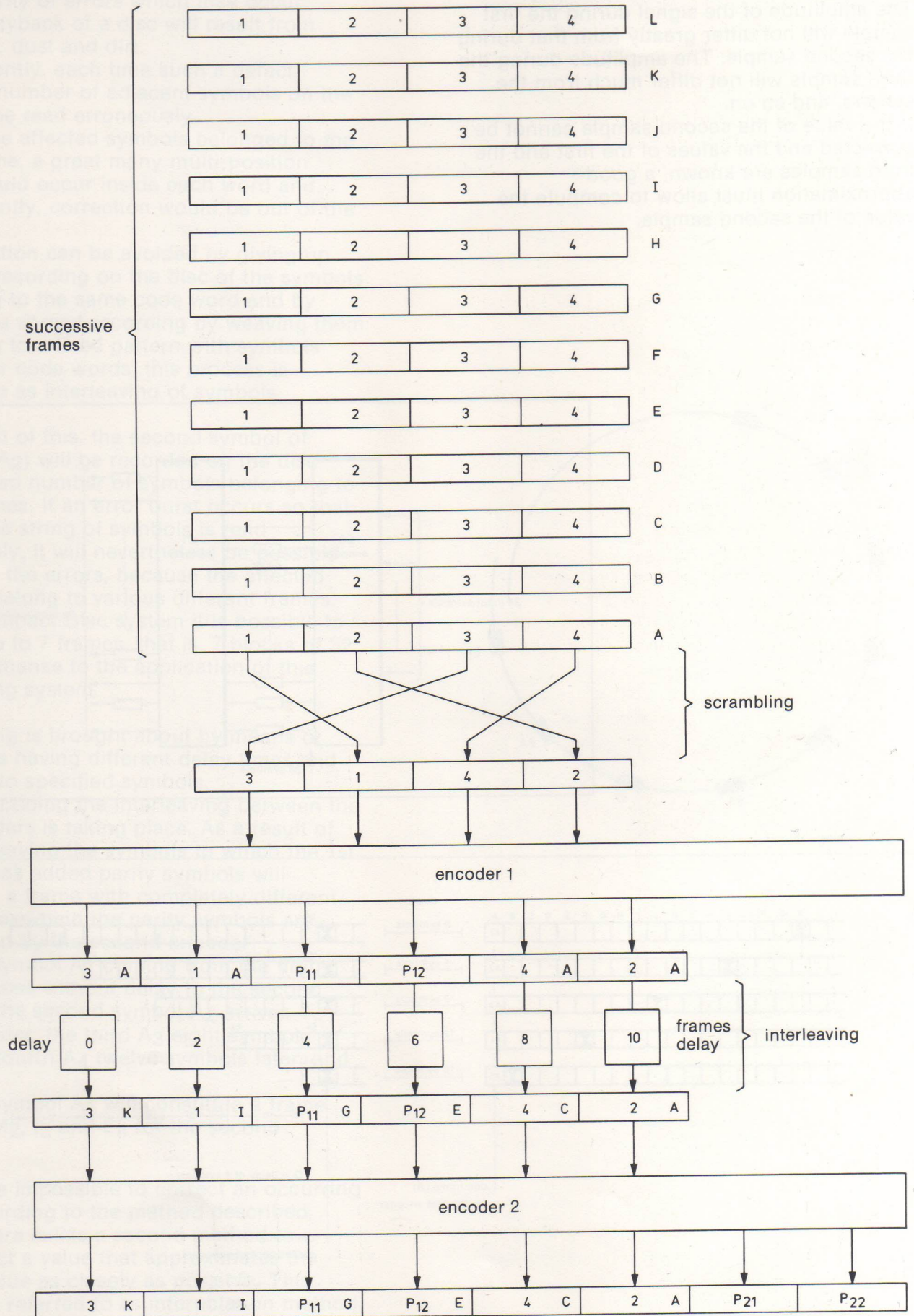
If it proves impossible to correct an occurring error according to the method described above, there exists a second method to reconstruct a value that approximates the correct value as closely as possible. This method is referred to as interpolation method.



Analogue signals are continuous signals which will usually not change abruptly. The amplitude of the signal during the first sample will not differ greatly from that during the second sample. The amplitude during the third sample will not differ much from the second, and so on. If the value of the second sample cannot be corrected and the values of the first and the third samples are known, a good approximation must allow to compute the value of the second sample.



Example of scrambling and interleaving for 4 symbols



Note: P₁₁, P₁₂ parity symbols encoder 1
P₂₁, P₂₂ parity symbols encoder 2

To enable the application of this method, the code word symbols are interchanged crosswise so that reading of symbol A₁ (of code word A) is followed by e.g. A₃ (instead of A₂) and then A₅; this is called: Scrambling of symbols.

If a disturbance causes erroneous reception of A₃ and A₅ and correction is impossible, we still have the symbols A₁, A₂, A₄ and A₆ which allow to compute interpolated value of A₃ and A₅.

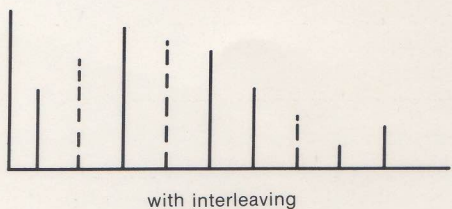
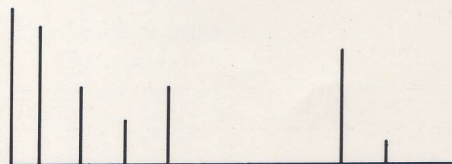
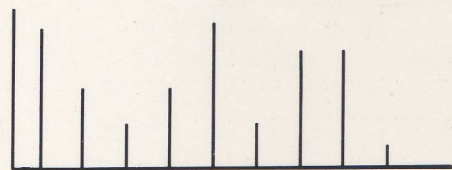
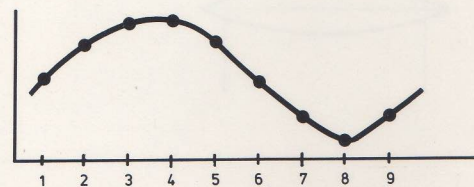
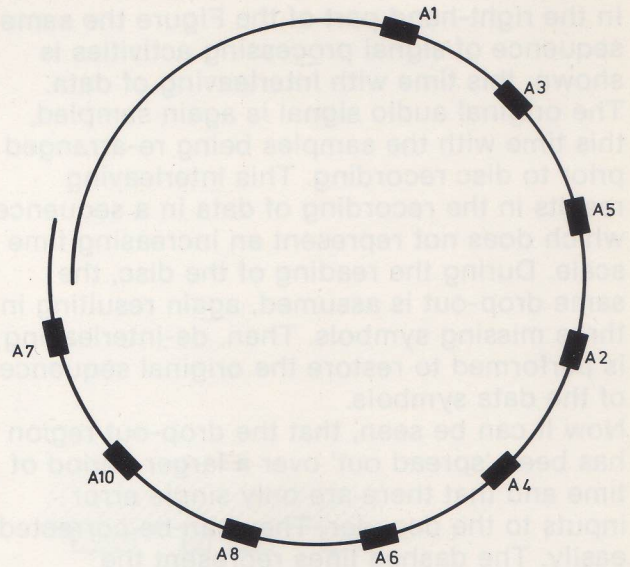
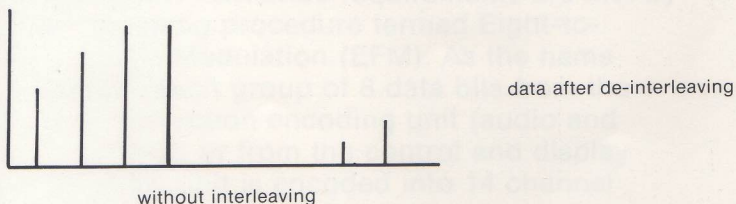
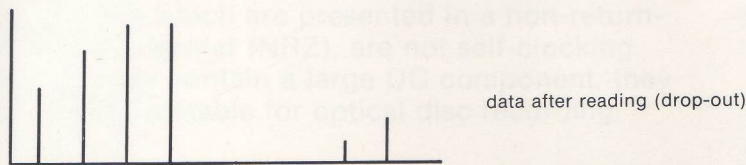
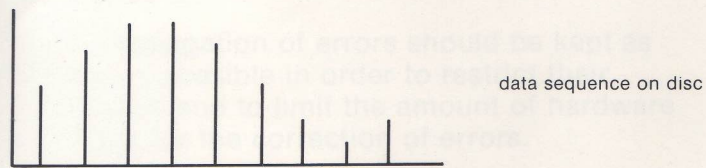
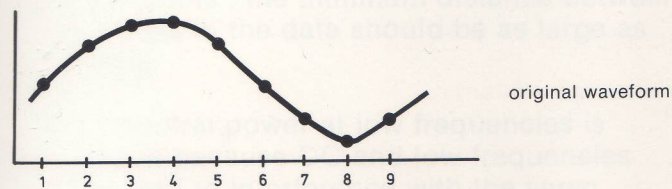
The principle of interleaving is illustrated in the Figure.

In the left-hand part, the sequence of signal processing tasks is shown without interleaving.

The audio signal is first sampled at time points 1, 2, 3, etc. than digitized and recorded on the disc.

If there is a drop-out during the reading of the disc, there will be some symbols missing in the received data. In the example used, three symbols are missing.

As the length of the drop-out region is greater than the error correction capability of the decoder, there is no chance of reconstructing the missing values, i.e. the audio output has to be muted in order to avoid clicks.



In the right-hand part of the Figure the same sequence of signal processing activities is shown, this time with interleaving of data. The original audio signal is again sampled, this time with the samples being re-arranged prior to disc recording. This interleaving results in the recording of data in a sequence which does not represent an increasing time scale. During the reading of the disc, the same drop-out is assumed, again resulting in three missing symbols. Then, de-interleaving is performed to restore the original sequence of the data symbols. Now it can be seen, that the drop-out region has been 'spread out' over a larger period of time and that there are only single error inputs to the decoder. They can be corrected easily. The dashed lines represent the recovered data.



Modulation

A modulation system for an optical audio disc must fulfil certain requirements:

- self-clocking ability
- read-out at high information density
- low spectral power at low frequencies
- small error propagation

As the bit clock has to be re-generated from the data after the reading of the disc, self-clocking ability is mandatory. This is normally achieved by making the maximum distance between high-to-low and low-to-high transitions in the data as small as possible.

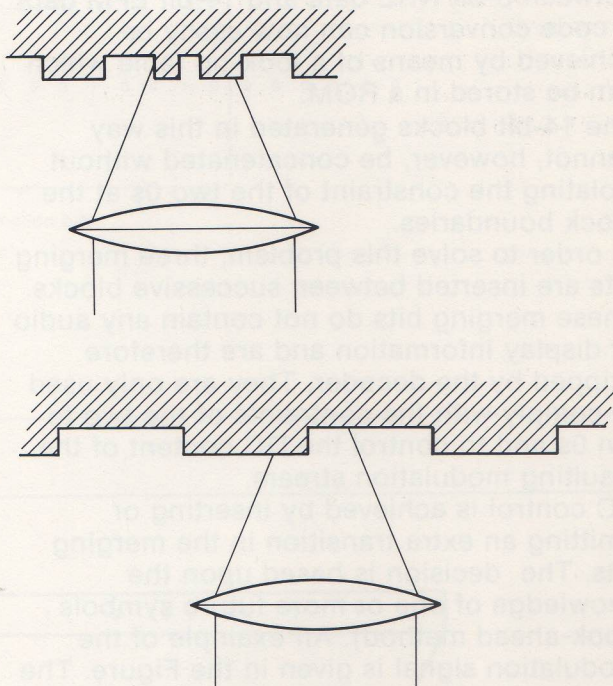
The light spot which is used for the reading of the disc has finite dimensions. This could lead to inter-symbol interference because of the high information density used on the disc. To avoid this, the minimum distance between transitions in the data should be as large as possible.

Low spectral power at low frequencies is desirable because DC and low frequencies could lead to interference with the servo systems.

The propagation of errors should be kept as small as possible in order to restrict their influence and to limit the amount of hardware needed for the correction of errors.

As data which are presented in a non-return-to-zero format (NRZ), are not self-clocking and may contain a large DC component, they are not suitable for optical disc recording.

The afore-mentioned requirements are met by an encoding procedure termed Eight-to-Fourteen Modulation (EFM). As the name implies, each group of 8 data bits from the error correction encoding unit (audio and parity bits) or from the control and display encoding unit is encoded into 14 channel bits.



In order to ensure self-clocking ability as well as to permit the reading of high-density information, there are always at least two 0's (zeros) between successive 1s (ones) and no more than ten zero's in a run.

There exist 277 different 14-bit sequences which satisfy the constraint of at least two zero's and maximum ten.

Deleting the 21 patterns with the longest run-length leaves 256 sequences which perfectly match the $2^8 = 256$ possibilities of the 8-bit wide input data.

Thus, there is a one-to-one correspondence between 8-bit NRZ data and 14-bit EFM data. A code conversion can now easily be achieved by means of a look-up table which can be stored in a ROM.

The 14-bit blocks generated in this way cannot, however, be concatenated without violating the constraint of the two 0s at the block boundaries.

In order to solve this problem, three merging bits are inserted between successive blocks. These merging bits do not contain any audio or display information and are therefore skipped by the decoder. They are only used to comply with the constraints of the two to ten 0s and to control the DC content of the resulting modulation stream.

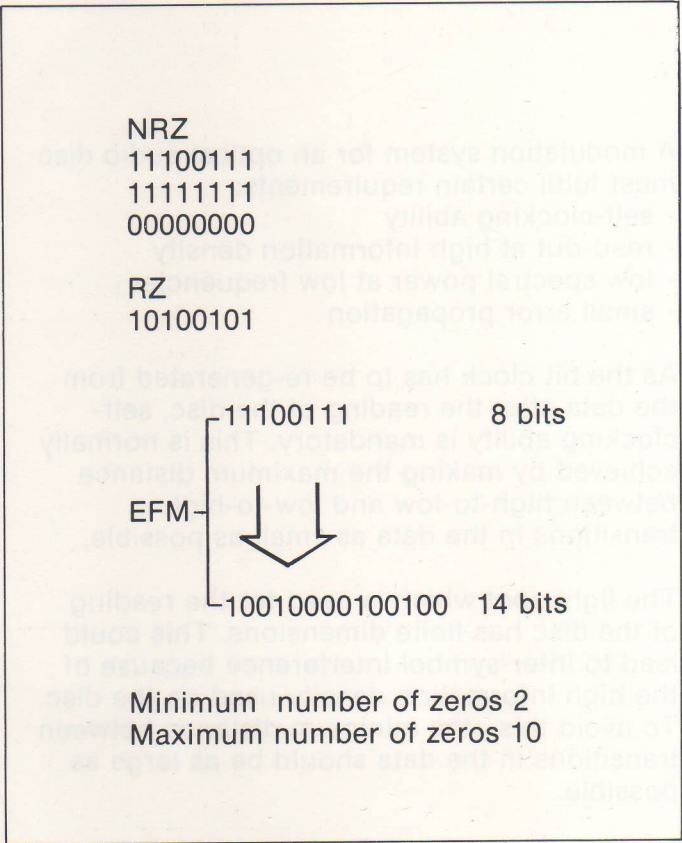
DC control is achieved by inserting or omitting an extra transition in the merging bits. The decision is based upon the knowledge of one or more future symbols (look-ahead method). An example of the modulation signal is given in the Figure. The transmitted information is contained in the position of the transitions.

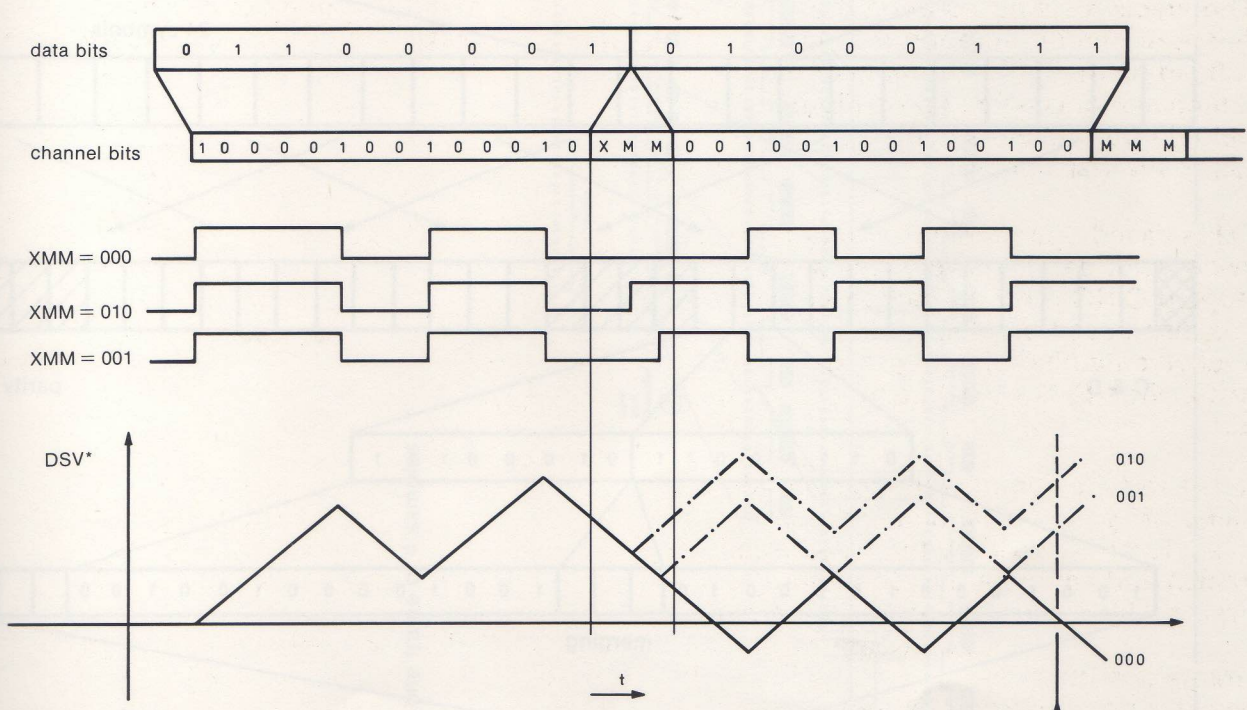
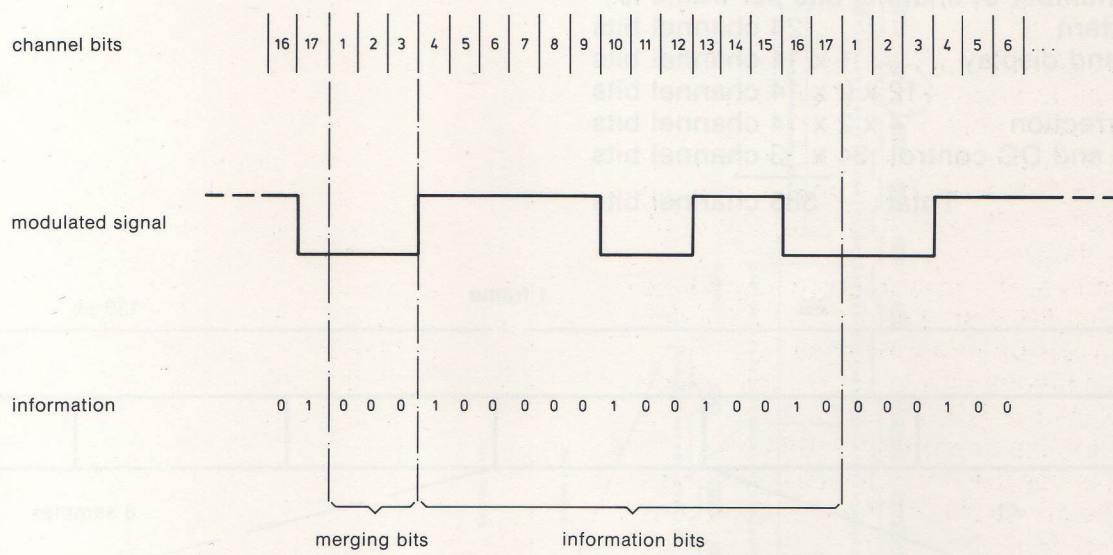
As EFM is based upon a block structure of 8-bit wide input data, it is very well suited for the adopted CIRC error correction system which is also based on blocks of 8 consecutive data bits. The propagation of errors is limited to the 8 data bits forming a symbol.

Due to the requirement of self-clocking ability some kind of synchronization is necessary. This is achieved by dividing the data stream into frames and adding a unique pattern to each frame which does not occur in the data.

Each frame contains:

- a special synchronization pattern
- 12 data words of 16-bits each
- 4 error-correction parity words of 16 bits each
- a control and display symbol of 8-bits

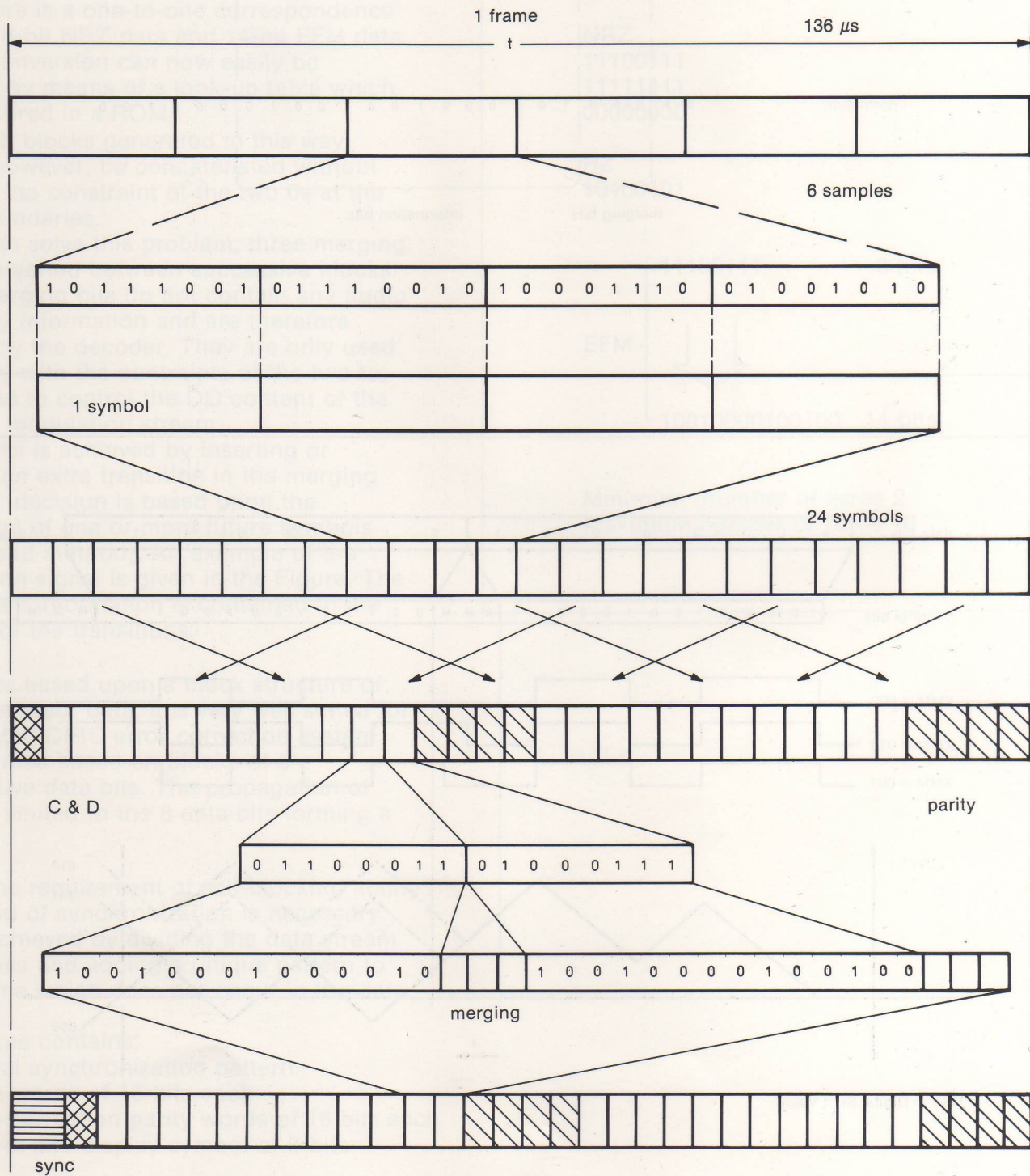




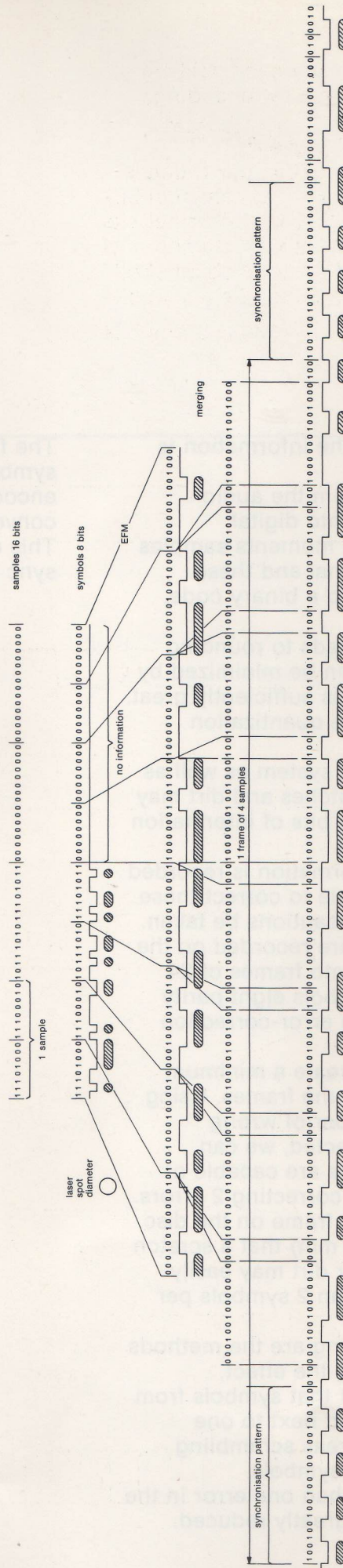
*DSV = Digital Sum Value

The data and parity words are each split-up into two 8-bit blocks before EFM encoding takes places.

- The total number of channel bits per frame is:
- sync pattern 24 channel bits
 - control and display 1 x 14 channel bits
 - data 12 x 2 x 14 channel bits
 - error correction 4 x 2 x 14 channel bits
 - merging and DC control 34 x 3 channel bits
- Total 588 channel bits



Example of one frame of 4 samples



Summary

This Appendix tells how the information is recorded on the disc.

In the Compact Disc system the audio information is converted into digital information. At fixed time moments samples are taken of the audio signal and these samples are converted into a binary code.

This kind of processing leads to rounding errors, but these errors can be minimized by taking the number of codes sufficiently great. The rounding errors cause quantization noise.

Errors in the transmission system as well as the presence of dust, scratches and dirt may cause drop-outs by which bits of information get lost.

Owing to the fact the information is recorded in digital form, it is possible to correct these errors, provided some precautions be taken. Before the binary codes are recorded on the disc, they are combined into frames of 24 symbols. To these 24 symbols eight parity symbols are added in two error-correction encoders.

These 8 parity symbols create a minimum distance $d_m = 5$ between the frames. Using the formulas for the number of wrong symbols that can be corrected, we can compute that the decoders are capable of detecting 4 errors and of correcting 2 errors. The length occupied by a frame on the disc is, however, so small (0.4 mm) that a scratch or the presence of dust or dirt may easily cause the loss of more than 2 symbols per frame.

Scrambling and interleaving are the methods resorted to for minimizing the effect.

Interleaving has for result that symbols from different frames are placed next to one another on the disc, whereas scrambling changes the order of the symbols.

The risk of having more than one error in the same frame can thus be greatly reduced.

The final processing step is to feed the symbols leaving the error-correction encoders to a modulator where they are converted into a 14-bit return-to-zero code. This creates the possibility of generating a sync signal in the player.

The DAC converter converts a digital signal at its input into an analogue current proportional to the value of that digital signal.

Appendix B

The DA converter

Questions

1. *What principle is used by the DAC to convert a digital word into an analogue signal?*
2. *Of which basic elements is the DAC composed?*
3. *How is the digital signal finally converted into an analogue signal?*

The DA converter

The Philips DA converter converts a digital number at its input into an analogue current proportional to the value of that digital number

The Figure opposite shows the basic circuit for the current sources.

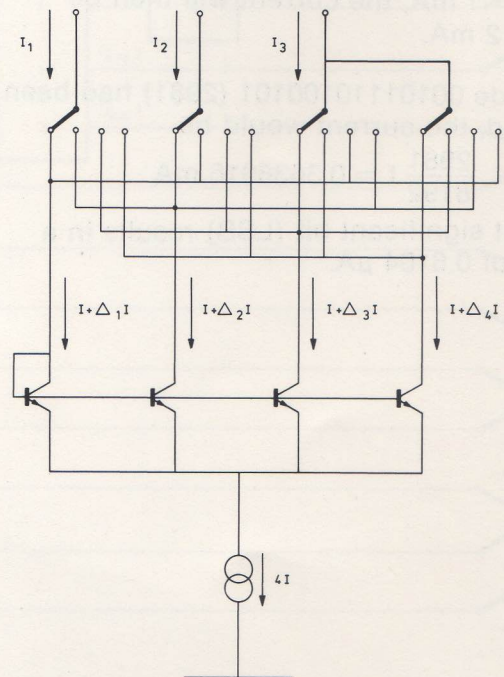
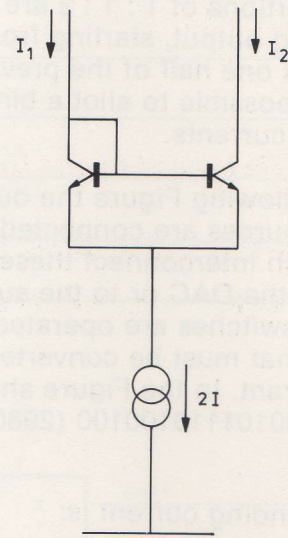
The current $2I$ is divided in I_1 and I_2 ; these currents are almost equal to each other so that only minor deviations $\Delta_1 I$ and $\Delta_2 I$ will occur.

The sum of the currents I_1 and I_2 always equals $2I$; consequently, $\Delta_1 I + \Delta_2 I = 0$.

The next Figure represents a circuit in which four transistors are connected similarly.

The collectors of these transistors are, however, connected via a matrix to four switches which, in turn, are connected to three outputs. The switches are actuated by a clock pulse. Each pulse causes the switches to make one step.

After four steps the switches have returned to the position shown in the Figure.



A consequence of this configuration is that the mean currents I_1 , I_2 and I_3 are equal to I , I and $2I$ respectively, for $\Delta 1 I + \Delta 2 I + \Delta 3 I + \Delta 4 I = 0$.

The proportions of the currents I_1 , I_2 , I_3 are thus defined very accurately as follows:

$$I_1 : I_2 : I_3 = 1 : 1 : 2 = 1 : 1 : 2.$$

The mean values of the currents are obtained by means of smoothing. The current proportions remain of course valid when the reference current is not $4I$ but $2I$, as occurs in the following Figure.

In this Figure the squares with the output current proportions of $1 : 1 : 2$ are cascaded. Each following output, starting from I , has a current that is one half of the previous one. In this way it is possible to allot a binary weight to the output currents.

In the next following Figure the outputs of the current sources are connected to switches which interconnect these outputs to the output of the DAC or to the supply voltage. The switches are operated by the binary code that must be converted into an analogue current. In the Figure shown, the binary code 00101110100100 (2980) is presented.

The corresponding current is:

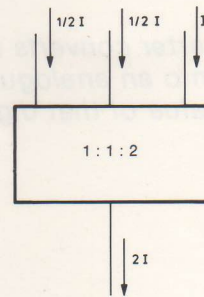
$$\left(\frac{1}{2048} + \frac{1}{256} + \frac{1}{64} + \frac{1}{32} + \frac{1}{16} + \frac{1}{4} \right) I = \frac{744}{2048} I$$

When $I = 1 \text{ mA}$, the current will then be 0.3632812 mA .

If the code 00101110100101 (2981) had been presented, the current would be

$$\frac{2981}{8192} I = 0.3638916 \text{ mA}$$

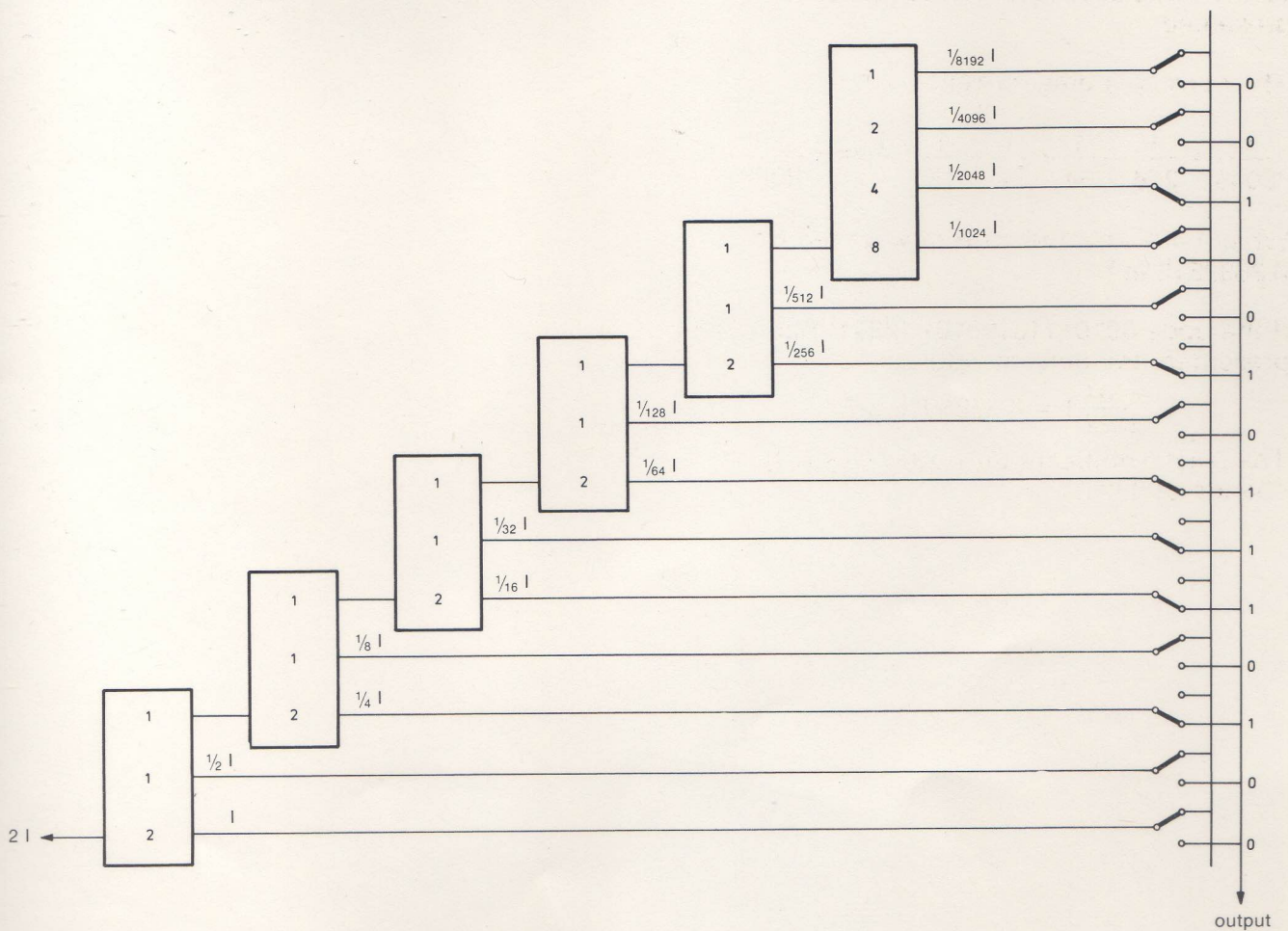
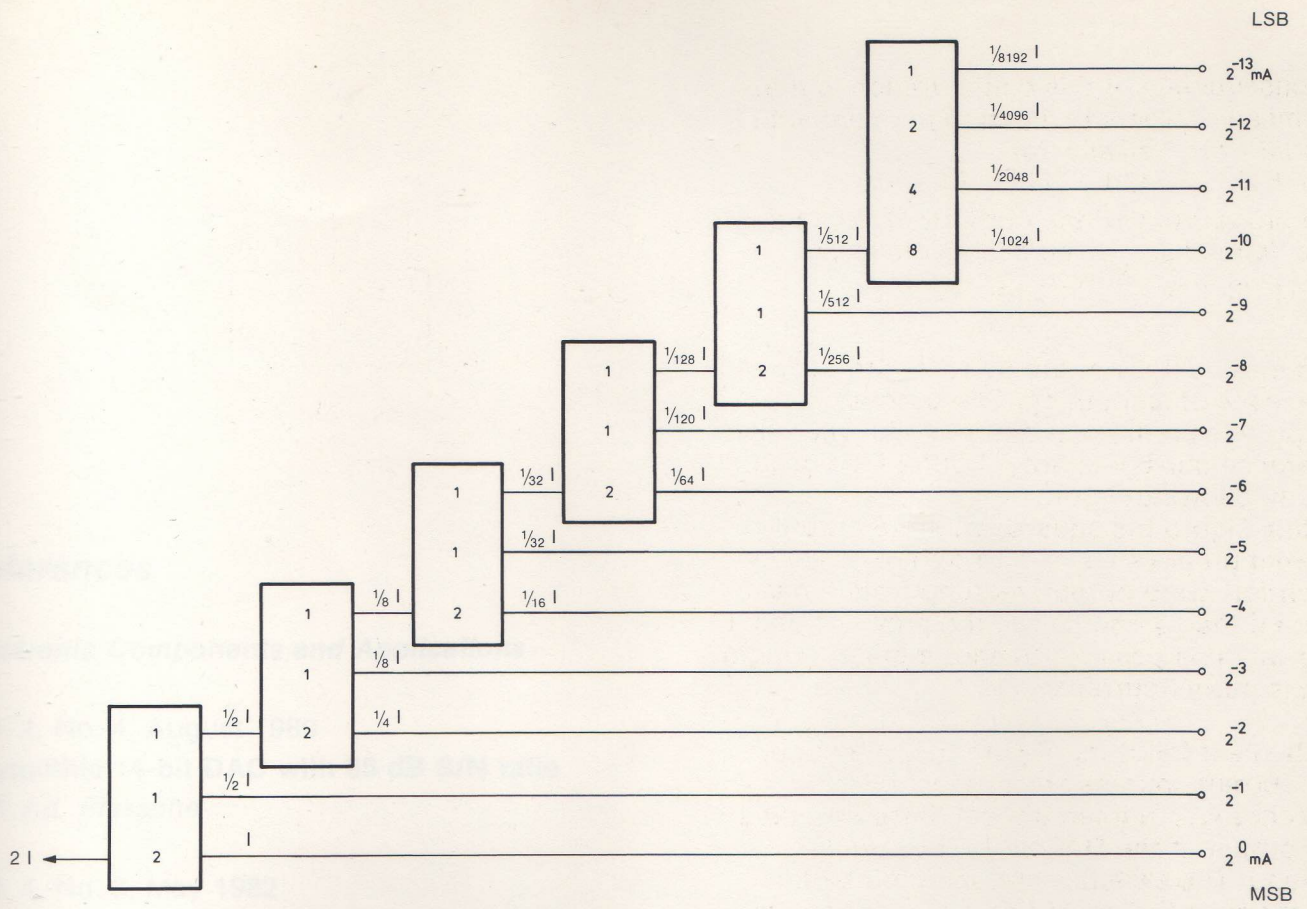
The least significant bit (LSB) results in a change of $0.6104 \mu\text{A}$.



$$I_1 = \frac{4I + \Delta 1 I + \Delta 2 I + \Delta 3 I + \Delta 4 I}{4} = I$$

$$I_2 = \frac{4I + \Delta 1 I + \Delta 2 I + \Delta 3 I + \Delta 4 I}{4} = I$$

$$I_3 = 2 \times \left(\frac{4I + \Delta 1 I + \Delta 2 I + \Delta 3 I + \Delta 4 I}{4} \right) = 2I$$



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ICs for Compact Disc decoders

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