STUDY, DESIGN, DEVELOPMENT TESTINGS & MEASUREMENT OF 'CD' SYSTEM -USING LASERS & MICROCONTROLLER

PROJECT REPORT 1994-95

P-1305

Submitted by

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In partial fulfilment of the requirements for the Degree of

BACHELOR OF ENGINEERING IN ELECTRONICS & COMMUNICATION ENGINEERING

of the Bharathiar University, Coimbatore

DEPARTMENT OF ELECTRONICS & COMMUNICATION ENGINEERING

Kumaraguru College of Technology

(Affiliated to the Bharathiar University)

(Approved by A.I.C.T.E., New Delhi)

COIMBATORE-641 006

Certificate

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

KUMARAGURU COLLEGE OF TECHNOLOGY

COIMBATORE - 641 006.

CERTIFICATE

This is to certify that the Report entitled

STUDY, DESIGN, DEVELOPMENT

TESTINGS &

MEASUREMENT OF 'CD' SYSTEM

USING LASERS & MICROCONTROLLER

P - 1305

has been submitted by

Mr. ____________________

in partial fulfilment for the award of the Bachlor of Engineering in the Electronics of communication Engineering Branch of the Bharathiar University, Coimbatore 641 046.

during the academic year 1994 _ 95

Guide

Head of the Department

Certified that the candidate was examined by us in the Project work viva voce Examination held on 2/4/95 and the university Register Number was ____________

Internal Examinar

External Examiner

Future Riders of 'CD'

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C E R T I F I C A T E

This is to certify that Mr. Anu [illegible]akasam, [illegible] along with Mr. Ve[illegible]velu. A., [illegible] Magesh. [illegible] and Mr. Saravanan, [illegible], Final Year students of BE (L&C) of the Kumaraguru College of Technology, Coimbatore, have done Project work [illegible] Study, Design, Development, T[illegible] and [illegible] of [illegible] Using [illegible] in our Organisation [illegible] the period from 13th August 199[illegible] to 20th [illegible] 1995.

We have received a copy of the [illegible] work from the above team.

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This is to certify that Mr. V[illegible] A. [illegible] with Mr. Saravanan S.D., Mr. [illegible]thaprak[illegible] K. and Mr. Magesh [illegible], Final Year [illegible] (L&C) of the Kumaraguru College of Technol[illegible] Coimbatore, have done Project [illegible] Design, Development, Testing and [illegible] CD System - Using [illegible] our Organisation during the [illegible] from August 1994 to 20th March [illegible]

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CERTIFICATE

This is to certify that Mr. [illegible] Mr. Vetrivelu A., M[illegible] [illegible]prakasam [illegible] [illegible] S.P. [illegible] BE (ECE) of the Kumaraguru College of Technology, Coimbatore, have done [illegible] Study, Design, Development [illegible] ment of [illegible] System [illegible] [illegible] from 12th August 1994 to 20th [illegible] 1995.

We have received a copy [illegible] the above team.

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Dedicated to my
Beloved Parents

Contents

Acknowledgement

ACKNOWLEDGEMENT

This valued project has flowered out to our exclamatory senses. It's been the result of our hardship, persistance, intellect co_operation and most fortunate combination of our skills. Our foremost gratitude to the Almighty for this sucessful completion of this project in a significance of its own and to our peak of contentment.

We are in a sincere responsibility to pay our gratitude to Dr.S.Subramanian B.E., M.Sc., (Engg), Ph.D principal Kumaraguru College of Technology.

Our intensity of thankfulness always remain to Prof.Muthuraman Ramasamy M.E. for his gem of suggestions those which have thrown light for the visibility of perfection in this project. This ideal reputation and prudence also belongs to our humble guide Mr.S.Booshnan B.Sc., B.Tech. Senior Manager R&D, who has brought out of our minds the illustrations, ideas and talents put up together practically. He is our guide and right soul to the body of our project. And to our random pace of completion of this project was Mr.Sirdardan B.Tech., Deputy Manager, R&D. We pledge to him heartfully many_a_times towards his moralities, advices and motivation on his behalf.

Preface

PREFACE

The term audio is defined as pertaining to audible sound,normally considered to extend over the range of 15 to 20,000 HZ.Engineering involves putting scientific knowledge to practical use ,thus audio engineering involves the application of electrical and acoustic technology to audio equipment and systems.

Covering a span of over a country,audio engineering dates back to 1876,when Graham Bell first demonstrated the electrical transmission and reproduction of voice signals.Although sound had been converted to electrical signals 15 years earlier by Philip Reiss,the revolutionary feature of Bell's invention was the use of an electronic transducer for reproducing the voice signals. Thus,audio Engineering,as demonstrated by the telephone, is an old and established scientific discipline.

One the other hand the use of audio technology for applications beyond voice communication over very limited distances was not to be realized until 50 years later with the invention of the dynamic loudspeaker ,by Kellogy and Rice in 1925 and the availability of the vacum_ tube amplifier. Consequently,audio engineering can be considered a relatively new discipline ,since it involves a variety of electronic devices and the wide application to many entertainment ,educational and industrial fields,most of which have been developed in the last few decades.In recent years audio technology has witnessed the advent of revolutionary noise reduction ,digital processing and laser recording systems that have required audio engineers to update and broaden their base of expertise.

There are about a billion reasons that prompted us to do the project on Digital Audio Systems. Specifically,about a billion CD's are sold every year, and the number is still climbing. Exceeding the exceptions of even its most advent supporters,the compact disc has become one of the most successful consumer electronics products ever introduced.The eager world wide marketplace has encouraged rapid development of CD technology and spawned entirely new applications for the dimpled disc.

The entertainment world is going digital sophisticated digital technology holds promise for virtually complete fidelity in sound reproduction and super picture quality in television sets.A number of products based on digital technology such as the compact disc (CD),digital audio tape (DAT) and digital TV have been developed.

Among the most popular forms of entertainment in digital technology is the compact disc. Compact disc is the most radical change in recording technology since Thomas Edison demonstrated his tin-foil cylinder receiver over a century ago.

The new dimension in audio entertainment is digital sound reproduction. The latest trend world wide is inclined towards the compact disc system which has changed audio preferences of music lovers.

Exactly how important is it to bone up on a new technology such as the compact disc ? How significantly will it influence our everyday lives.and for how long ? There are a number of factors that determine the life span of a technology.

Introduction

INTRODUCTION

In the 1980,the compact disc system has prospered beyond the wildest dreams of its inventors.The original audio CD was one of the most successful new electronic products ever introduced,everyone was surprised by its rapid acceptance by music lovers.Over the first few years the biggest news concerned shortages,particularly of discs themselves.Then the disc caught the attention of data lovers as well.CD-ROM quickly stared its own acceleration into the market of mass storage.As if that wasn't enough.CD-I's future looks similarly promising as an interactive audio-video medium.Of course,while reading data is fun, writing data is even more fun. Thus a standard for write-onceCD's was introduced, and erasable CD technology has been developed. Further more, CD's with graphics and MIDI information and mini-CD's were also introduced into the market place.

Exactly how important is it to be bone on a new technology such as the Compact disc ? how significintly will it influence our every day lives, and for how long ? there are number of factors that determine the life span of a technology Such as Manufacturing cost,product performance ,market penetration,user boredom,and innovative competition all move technological evolution forward and heighten expectations.As for as the CD family goes it is safe to say that the future looks bright indeed.In market share continues to expand ,and new and consequential applications are undoubtedly waiting to be discovered by insightly entrepreneurs.Of course,no technology, except the very primitive kind last forever, and some day (Perhaps rela-tively soon) the CD will be only a curiosity. Fittingly , the discs themselves should long outlast their usefulness.

Meanwhile ,this project book covers all the fundamentals of CD player and should prove useful to anyone delving into this technology, especially for the first time. Hopefully,it will answer all questions about a audio player.

INTRODUCTION : To Digital Audios

In spite Of the fact that professional audio has been handled predominantly in analog form for more than a century, it would appear that a total transition to digital audio technology is under way after less than 15 years of experience .The general reason is,as the analog elements in the audio chain such as recording, processing, transmission, distribution and reproduction are replaced with digital elements,a number of advantages appear.These advantages relate to audio quality. operational efficiency, new processing possibilities or economy.

TABLE : Some Important Events In The History Of Audio

1857 -	Leon Scott demonstrated the phonautograph (a talking machine)
1877 -	Edison filed his patent "Improvenent in Phonograph or speaking machine.
1925 -	Electrical recording and reproduction introduced by Bell Laboratories.
1927 -	First motion picture with sound released (The Jazz Singer).
1937 -	Binary-Coded PCM invented by A.H.Reeves.
1948 -	Columbia introduced LP(Long-Playing) Record.
1961 -	Digital reverberation simulated in computers.
1967 -	NHK(Japan) demonstrated Digital Audio Tape Recorder.
1971 -	Digital delay line Introduced.
1972 -	Records digitally mastered by Nippon Columbia,Japan.
1972 -	PCM used in united kingdom for high quality sound distribution for radio and TV.
1975 -	Real-Time digital reverberation system available.
1976 -	Digitally restored caruso record released.
1976 -	BBC Demonstrated 10 - Channel digital records.

1977 - Several professional digital audio records released.

1980 - Idea of COMPACT - DISK System announced.

1982 - COMPACT - DISK Players released.

1984 - First Professional Digital Audio mixing Console released

COMPARISION OF ANALOG AND DIGITAL AUDIO SYSTEM SPECIFICATIONS

Specifications or Feature	LP Record	Compact Disc System
Frequency Response	30 Hz To 20 Khz	20 Hz To 20 KHz
Dynamic Range	70 dB	> 90 dB
S / N Ratio	60 dB	> 90 dB
Harmonic Distortion	1 - 2%	0.004%
Channel Separation	25 - 30 dB	> 90 dB
Stylus Life	500 - 600 Hrs	5000 Hrs

A brief description of the inherent drawbacks of conventional LP's and audic tapes will help to appreciate the advanced feature and high technology used in compact discs

Phonograph (LP) records store information in a continuous groove cut into the surface of the disk. The storage technique is analog, i.e the amplitude or depth of the groove is a direct representation of the audic signals via a mechanical pick up (needle or stylus) that detects the shape of the groove and is in direct contact with the surface of the disk.

In case of audic tape, the audic signals are recorded in the form of magnetic patterns on the tape. The length of the magnets depends on both the speed of the tape and the frequency of the signal applied, while their field strength is commensurate with the strength (Loudness) of the signal in replay mode as the tape passes the head these small magnets produce changing magnetic fields in the pole pieces of the replay head. These are replica of the currents which were applied to the winding during the recording process. Some of the main drawbacks of LP's and tapes as follows

1. The recording and replay of the audio signals in case of LP's and Tapes are all in the analog form,distortion can easily occur any where from the pick up level to the final output stage.

2.There is physical contact between the record and the stylus,In the case of tape between the record/replay head and the tape. This results in wear over a period of time,requiring cleaning and renewal of the stylus and the record/replay head periodically to obtain best performance.

3.The speed at which the record moves and the tape passes the head should be critically controlled any variation in these speeds has a direct effect on the reproduction.Wow and flutter are caused due to variations in this speed.

4.Damage to the Record/Tape can occur which cannot be rectified,resulting in loss of that particular section of information.

Because of the above mentioned draw backs we are going for Digital Audio Systems, particularly CD's.

Introduction to 'Compact Disc' (CD)

INTRODUCTION TO THE COMPACT DISC :

COMPACT DISC :

Compact disc is an optical disc storage system developed and licensed by PHILIPS and SONY corporation. This system stores 74 minutes, 33 seconds of information (both digital audio and subcode), or other non audio data on a 12 cm Dia optical disc. The Disc is made of plastic (Polycorbonate),with a top metalised layer, and is read by reflected laser light.

Audio signals stored on a Compact Disk are in high density digital format,which means that the signals recorded are an extremely accurate representation of the original audio signal .

On replay,the digital information is read by a laser beam and converted into analog electrical form.Since the disk is scanned by laser beam,there is no physical contact with the disc surface,eliminating the wear found in LP's and audio tapes.The 1000 th play of a compact disk will sound every bit as good as the first,a feature which LP's and tapes cannot claim.

The disk rotates at 500 RPM when the laser beam starts at the center and slows to 200 RPM as it follows the spiral pits to the outer side of the disk,maintaining constant speed of read out.As the digital information on the disk is protected by a layer of clear plastic,the disks are more robust then conventional LP's.

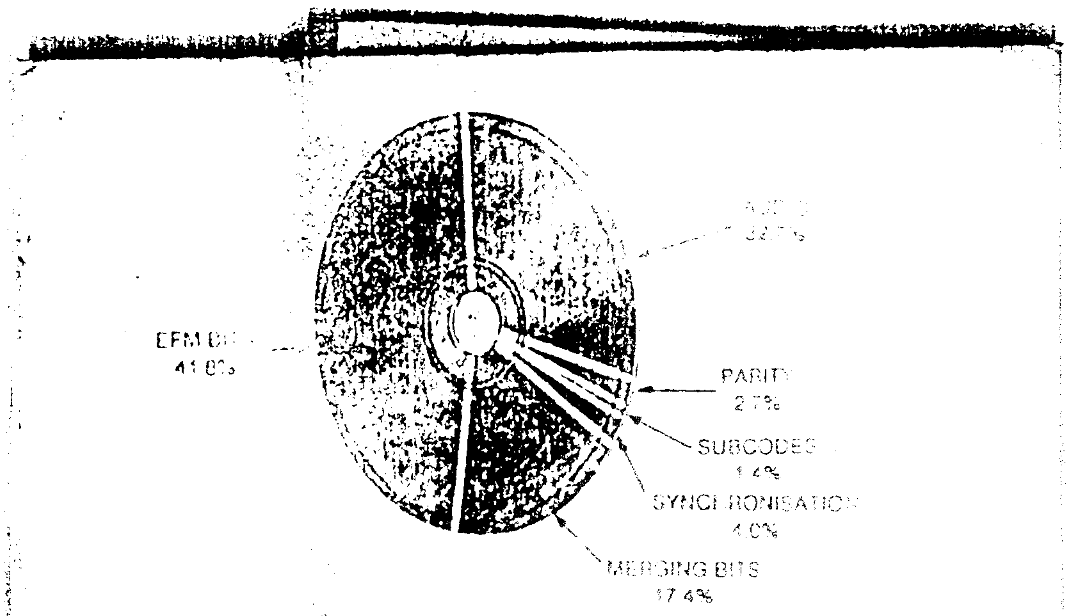
If a certain number of bits are lost, sophisticated error detecting codes are used to restore the information.CD's offer better quality sound with less distortion and no audible background noise.Digitally recorded music is "clean".

The stereo image is rock steady because of a [illegible] dB channel separation and negligible phase shift over the whole audible sound range.Conventional pick ups provide slightly over 30 dB channel separation,at best,and this subjected to phase changes over the frequency range giving an unstable stereo image. Thus all the inherent draw backs of conventional LP's and audio tape recorders are eliminated in CD's.

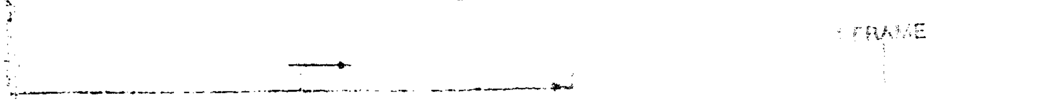
Besides these advantages ,CD's have a special feature called programming.Any particular section on the disk can be selected for play and others skipped (even replay of the same section is possible!). They can be programmed for random or shuffle play. In Pratice the CD system is an extremely complex piece of equipment. For more than just audio information is encoded on the disk. In fact,only one-third of the CD's capacity is used to store digitised sound; the rest is used for error correction,subcodes,indexing,parity checks,synchronization etc., as well index details which give the number of tracks and location of tracks on the disks.The more sophisticated players display the title of the disk.

There are 2,861,800 bits of non audio information is processed for every second of music - 10,302,500,000 bits for each hour of music.In all,a CD can contain a total of just under 20 Billion Bits - to be precise 19,919,878,200 bits !

All this digital information is encoded on the disc in a very complex format inorder to reduce the possibility of errors when being read.



Information encoded on a compact disc.



Configuration and manufacturing process of compact disc: The basic specifications, the signal characteristics produced by the put, and the mastering and replication sequence to produce discs.

DISC SPECIFICATION AND PIT GEOMETRY.

Disc Specification

The specifications and dimensions of the compact disc are shown in table and fig. The diameter of the disc is 120mm, and the center hole is 15mm. The signal is readout through the 1.2 mm transparent disc substrate. The disc rotates counter clockwise as seen from the reading side. The spiral track pitch is 1.6 micrometer and is read out from the inside to the outside. Density is about 16,000 tracks per inch. The track length is given by

[illegible]

Where P = track pitch

S = area of program zone

r_o = outside diameter of program area.

r_1 = Inside diameter of program area.

The program area starts as a 50 mm diameter and ends at a maximum of 116 mm.

The lead-in and lead out zones are used to control the player system such as track access and automatic playback.

TABLE 9.1 Basic Specifications of the CD System

Recording method	
Signal detection	Optical
Linear recording density	43 kbit/in (1.2 m/s)
Area recording density	683 Mbit/in^2
Audio specifications	
Number of channels	2-channel stereo
Playing time	Approximately 60 min
Frequency response	20 ~ 20,000 Hz
Dynamic range	> 90 dB
Total harmonic distortion	< 0.01%
Channel separation	> 90 dB
Wow and flutter	Equal to crystal oscillator
Signal format	
Sampling frequency	44.1 kHz
Quantization	16-bit linear/channel 2' complement
Preemphasis	No or [illegible] μs
Modulation	EFM
Channel-bit rate	4.3218 Mbit/s
Error correction	CIRC
Transmission rate	2.034 Mbit/s
Redundancy	≈ 30%
Disk specifications	
Diameter of disk	120 mm
Thickness of disk	1.2 mm
Diameter of center hole	15 mm
Program area	50 ~ 116 mm
Scanning velocity	1.2–1.4 m/s, CLV
Revolution speed	500 ~ 200 r/min
Track pitch	1.6 μm
Pit size	0.11 × 0.5 × 0.9 ~ 3.2 (μm)

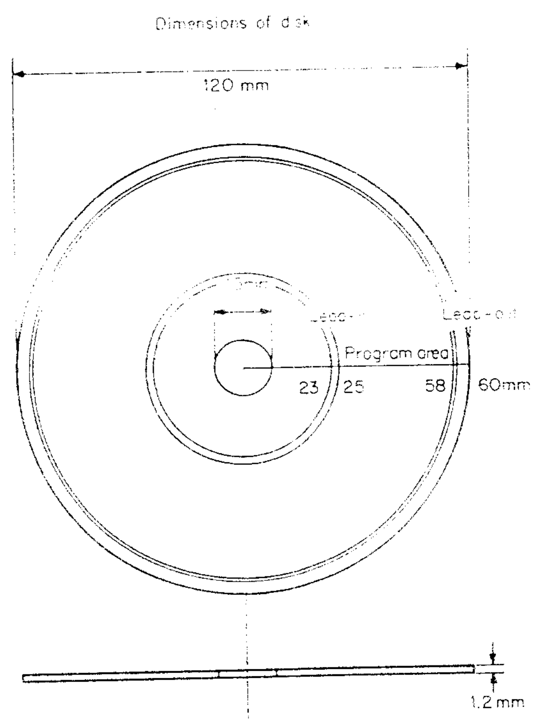


FIG. 9.40 Dimensions of the program area of the compact disk.

To maximize playing time, the CD is recorded by the constant linear velocity (CLV) method. The scanning linear velocity of the disc (V) is specified as 1.2 to 1.4 m/s. The revolution speed decreases from 500 to 200 r/min. However, the frequency response of the read out signal is the same at any disc radius.

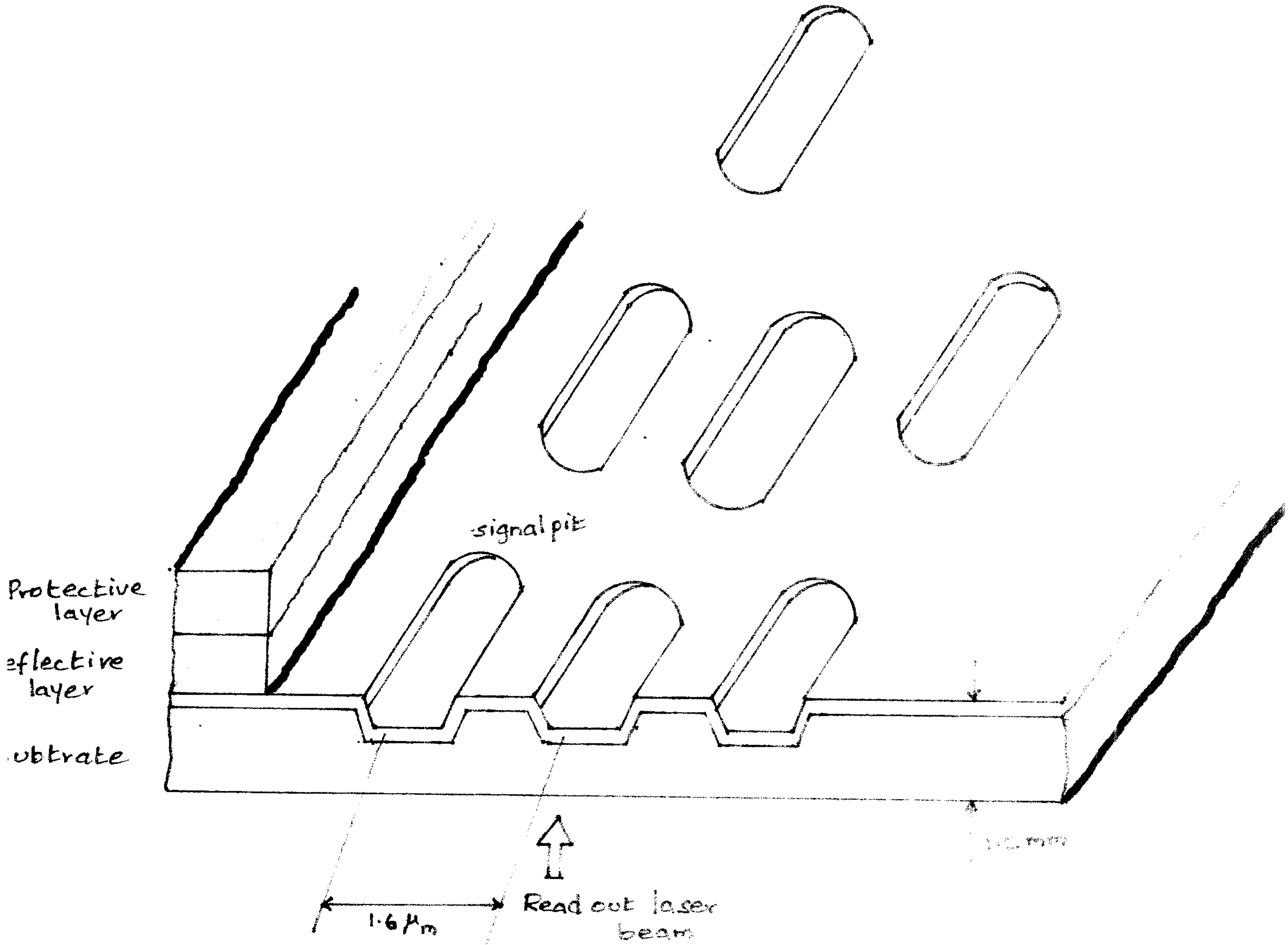
The playing time of the music program is given by

From this eqn the maximum recording time of a CD is about 74 min at 1.2 m/s.

Cross section of the Compact Disc:

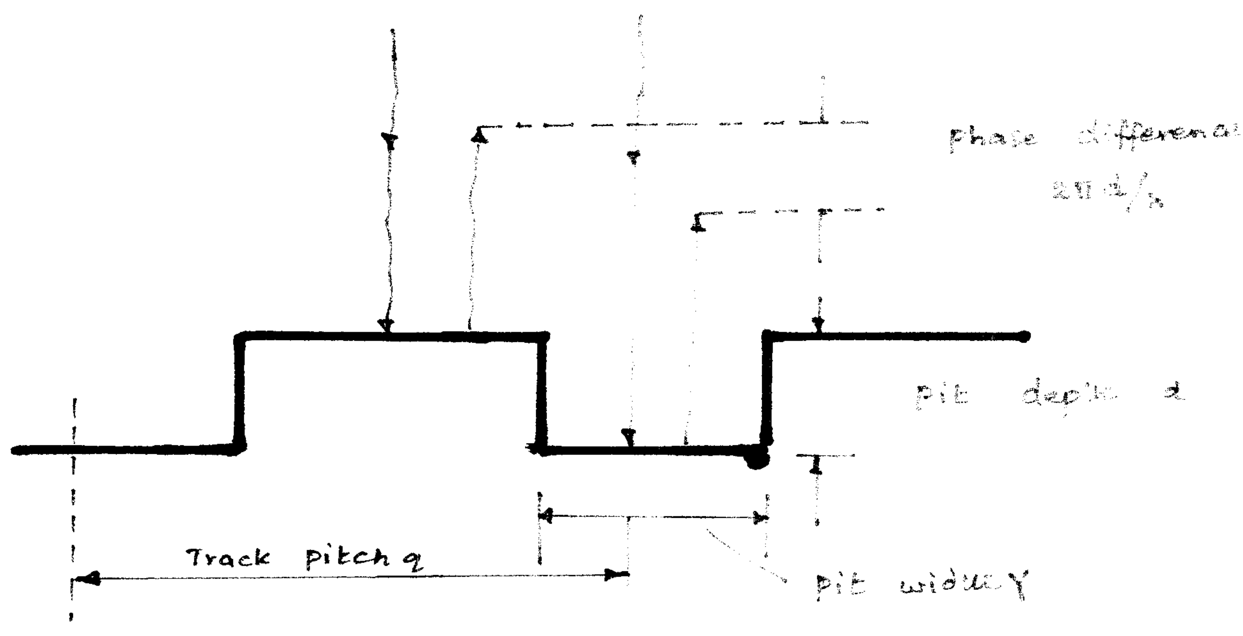
Fig shows a cross section of the Compact disc. The signal is picked up by a focussed laser beam through a transparent substrate. Its 1.2 mm thickness prevents signal disturbance by dust or finger prints. The material of the substrate must satisfy various optical and mechanical requirements such as birefringence, absence of defects and reliability. Polycarbonates, Polymethyl methacrylates and glass are suitable for disc - production requirements.

The replicated bits on the signal surface are about 0.1 micrometer deep, 0.5 the wide and several micrometers long. The signal surface is covered with an aluminium layer to reflect a laser beam. Thus reflective layer is coated with ultraviolet - light - cured resin to project it from scratches, moisture and other harmful effects. The label is printed on the protective layer by a silk-screen method.



cross section of a compact disk.

Phase Difference of a reflected beam



Pit-profile and signal characteristic:

The principle of CD signal detection is based on the diffraction phenomenon of a laser spot caused by the phase pit. A reading laser beam and pit geometry determine signal performance for an optical pickup.

The relation between pit shape and signal amplitude when the phase pit is illuminated by a readout laser is reviewed in the following paragraphs :

PIT Depth

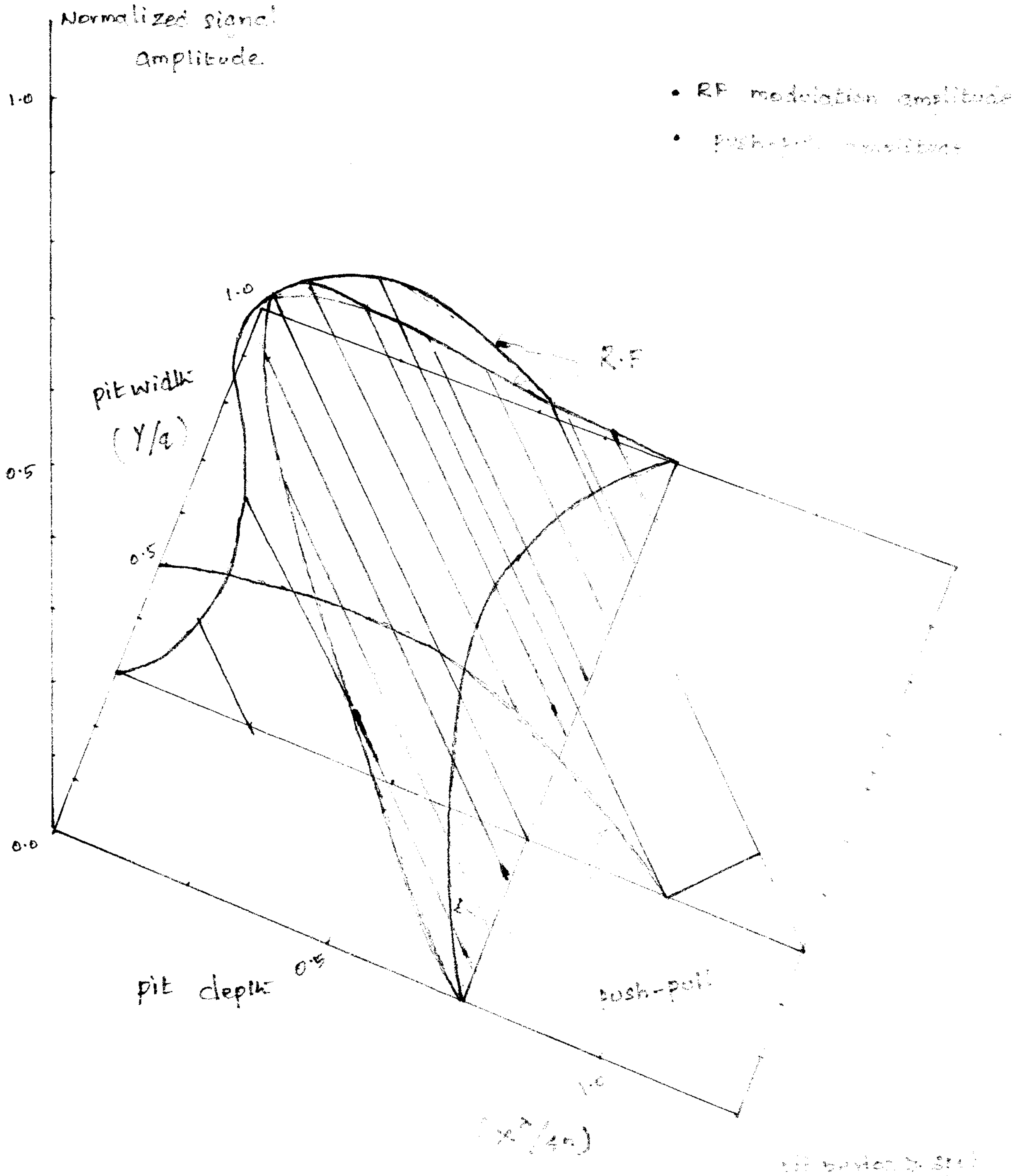
There is a phase difference between the reflected light rays from a pit and those from a land.

When the phase difference is , the modulation index of the reflected beam is at a max. Value by the resultant diffraction. Since a laser beam is reflected from a pit and the pit exists inside the transparent substrate of which the refractive index in n = 1.5, the pit depth gives the maximum high - frequency signal amplitude.

Pit width :

This parameter also affects signal quality, Viz., the amplitude, distortion, and frequency response of high - frequency and track - following signals. The pit width is equal to a recording spot size of 0.5 to 0.7 micrometer in mastering

Fig shows the relation between the signal amplitude and the signal cross - section pit profile.



Normalized signal amplitude Vs pit shape

Pit Length :

Pit length is related to the pulse width of the CD signal format. With a scanning velocity of 1.25 M/s, there are nine different pits on the signal surface : 0/87, 1.15, 1.45, 2.02, 2.31, 2.89 and 3. 18Km. Each pit length is effected by the disc - production processing operation and the read out characteristics of the optical pickup. This phenomenon is called asymmetry. Within a certain range asymmetry in not a problem because the correction circuit corrects it automatically.

Pit Edges :

The replicated pit does not have an ideal square cross section but does have a slope of pit edges. This pit shape is called the "Soccer Stadium" model.

The signal characteristics can be simulated by using a computer. An example of a simulated signal is shown in fig.

(CD) Glossary

IMPORTANT BASIC SPECIFICATION OF THE CD SYSTEM

('CD' TECH., GLOSSARY)

(i)	DYNAMIC RANGE
(ii)	SAMPLING FREQUENCY
(iii)	QUANTIZATION
(iv)	FREQUENCY RESPONSE
(v)	WOW AND FLUTTER
(vi)	INTRO SCAN
(vii)	SHUFFLE
(viii)	SIGNAL TO NOISE RATIO
(ix)	CHANNEL SEPARATION
(x)	HARMONIC DISTORTION
(xi)	TRANSMISSION RATE
(xii)	PROGRAMMABILITY
(Xiii)	PLAYING TIME

DYNAMIC RANGE

A maximum signal level which can not be exceeded without introduction unacceptable distortion and a minimum level where the system noise becomes prodiminant. The difference between these levels is usually called the "DYNAMIC RANGE".

SAMPLING FREQUENCY :-

Sampling means how fine the time is cut up. The number of samples in 1 second is called the sampling frequency.

According to Information theory, the number of samples in 1 second (sampling freq) must be at a frequency at least twice the highest frequency of the waveform one wishes to reproduce.

(i) if the upper limit of the range of frequencies to be reproduced is 20 KHz, a sampling frequency of 40 KHz (or) more must be selected.

QUANTIZATION :-

Quantization means how precisely the height of the waveform is expressed in modulo 2. (ie) number of bits.

DYNAMIC RANGE gets larger in proportion to the quantization number.

If 16 bits linear quantization dynamic range = [illegible] dB

(iv) FREQUENCY RESPONSE :

A uniform frequency response for direct current up to the low-pass filter cut-off frequency (for example 20 KHz) can be obtained. However in order to prevent aliasing signal components above the frequency can not be recorded.

(v) WOW AND FLUTTER :

These are due to the variations in the speed. These are reduced to the level of precision of the crystal oscillator that produces the clock frequency.(In practice 0)

(vi) INTRO SCAN

Using this facility certain number of tracks of a 'CD' are to be copied in a particular order on to a cassette.

(vii) SHUFFLE

This is also called "RANDOM PLAY" using this facility we can play any program which we like.

(viii) S/N ratio (in db)

The value of S/N ratio indicates the quality of sound reproduction. In is around 96bd for a practical case.

(ix) Channel separation :-

A value the same as the dynamic range can be obtained between the left and right channels.

(x) DISTORTION :

The noise and distortion form the D/A convertor, Low pass filter, line amplifier and other analog circuits. It is very negligible for practice.(0.003%)

(xi) TRANSMISSION RATE :

The value of this parameter indicates how many bit's it can transmit in signal second. 4.32 Mbits / Sec. Practical case.

(xii) PROGRAMMABILITY :

CD's have a special function called programming.

Any particular section on the disc can be selected for play and others skipped (even replay of the same section is possible).

(xiii) PLAYING TIME :-

This parameter indicates how much time can play the disc in the 'CD' player. It varies form 50 Seconds to 60 seconds.

Steps of Digitalization

INTRODUCTION

The essence of digital Audio lies in its numerical basis. A digital audio system converts the analog waveform into numbers, stores, transmits, or process those numbers, and then converts them back to an analog waveform.

The above steps may be divided into the digital audio recording of a music performance. The storage of that performance on a compact disc, and the reproduction of the performance with the CD player.

The digital audio chain, from the recording studio to our living room, is predicated on several basic phenomena. First, the analog waveform must be discrete time sample at a precise, unvarying rate. This divides the waveform into discrete intervals with sampling the signal must be filtered to prevent aliasing, a form of distortion.

Next, each sample must be converted to a number; this is the process of quantization. To minimize an inevitable error associated with quantization; dither signal must be introduced. In many digital audio recorders / reproducers these steps take place in a Linear Pulse Modulation System. This chapter examines all of these fundamental elements.

SAMPLING

Analog is continuous. Digital is discrete . To transform the original analog audio waveform into a digital audio signal, we examine the waveform at specific points in time. We must choose points on the original analog signal. This decision process is called discrete - time sampling or simply sampling.

HARRY'S THEOREM

This theorem defines the relationship between Sampling frequency and audio frequencies.

If we take samples at a frequency at least twice that of the highest audio frequency, complete reconstruction (of the original signal) can be accomplished.

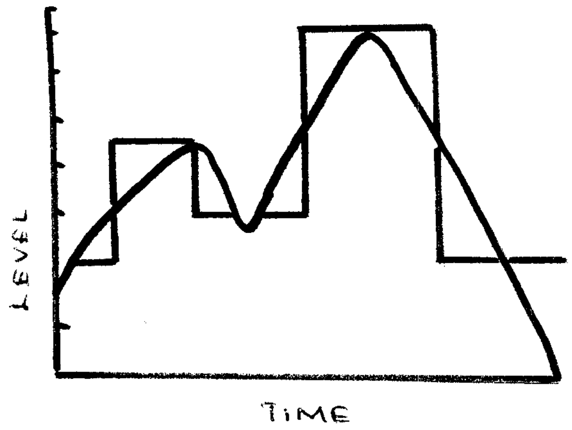
BRICKWALL FILTERS

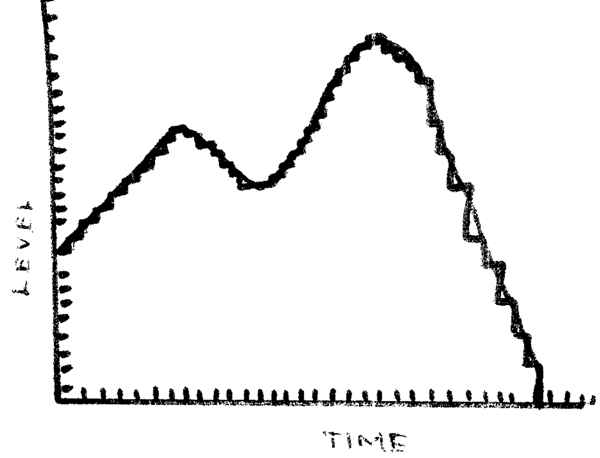
If a frequency is too high, it would not be properly encoded and would even create a kind of distortion called aliasing.

To prevent that, we limit the frequency content of the of the input signal with a low-pass filter. Its job is to ensure that all frequencies greater than one half the sampling frequency are attenuated below the amplitude resolution of the system.

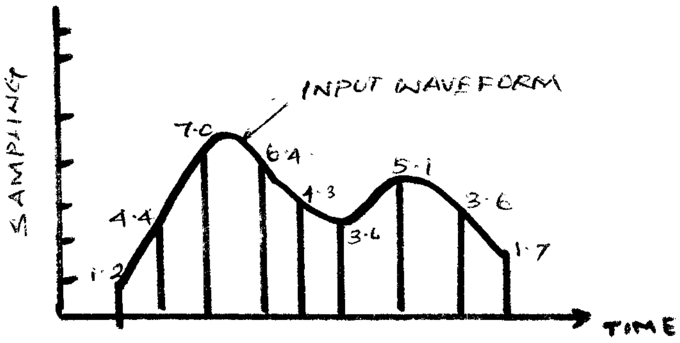
DIGITALIZATION

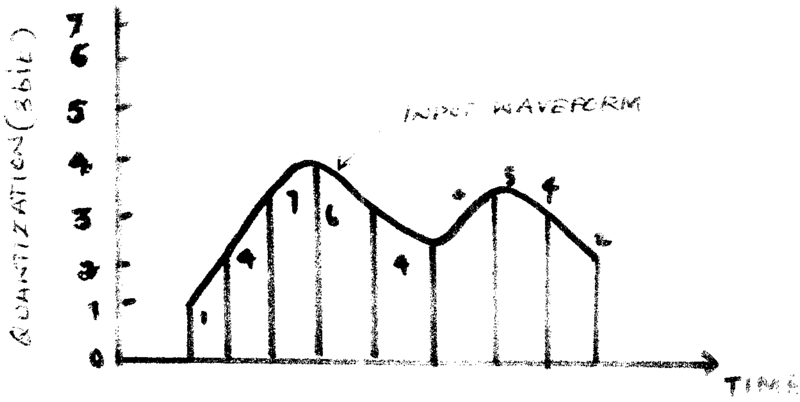
Converting the continuous wave to the presence or absence of pulse is "digitalization".





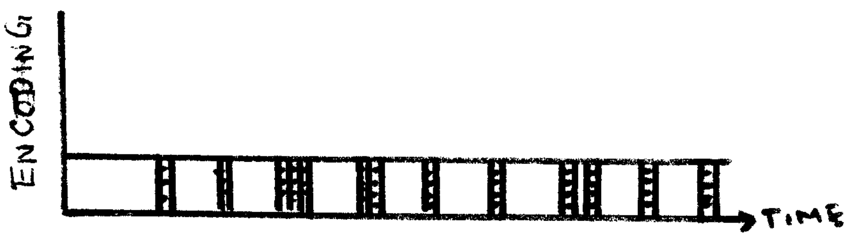
TIME & LEVEL IN REPRODUCTION OF ACOUSTIC WAVEFORM.

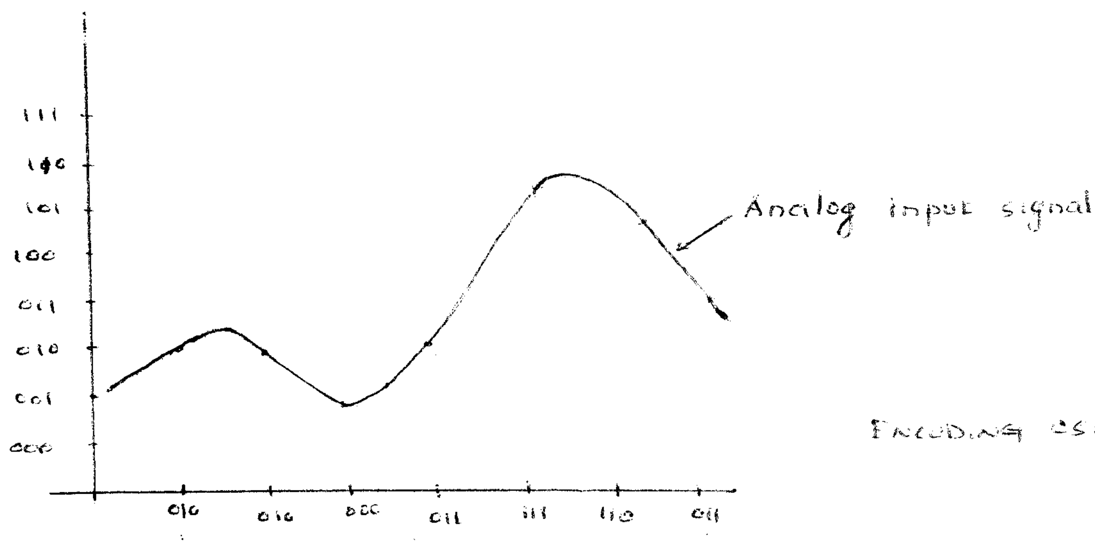




SCHEMATIC OF SAMPLING QUANTIZATIC

AND ENCODING.





ENCODING USING 3 BITS

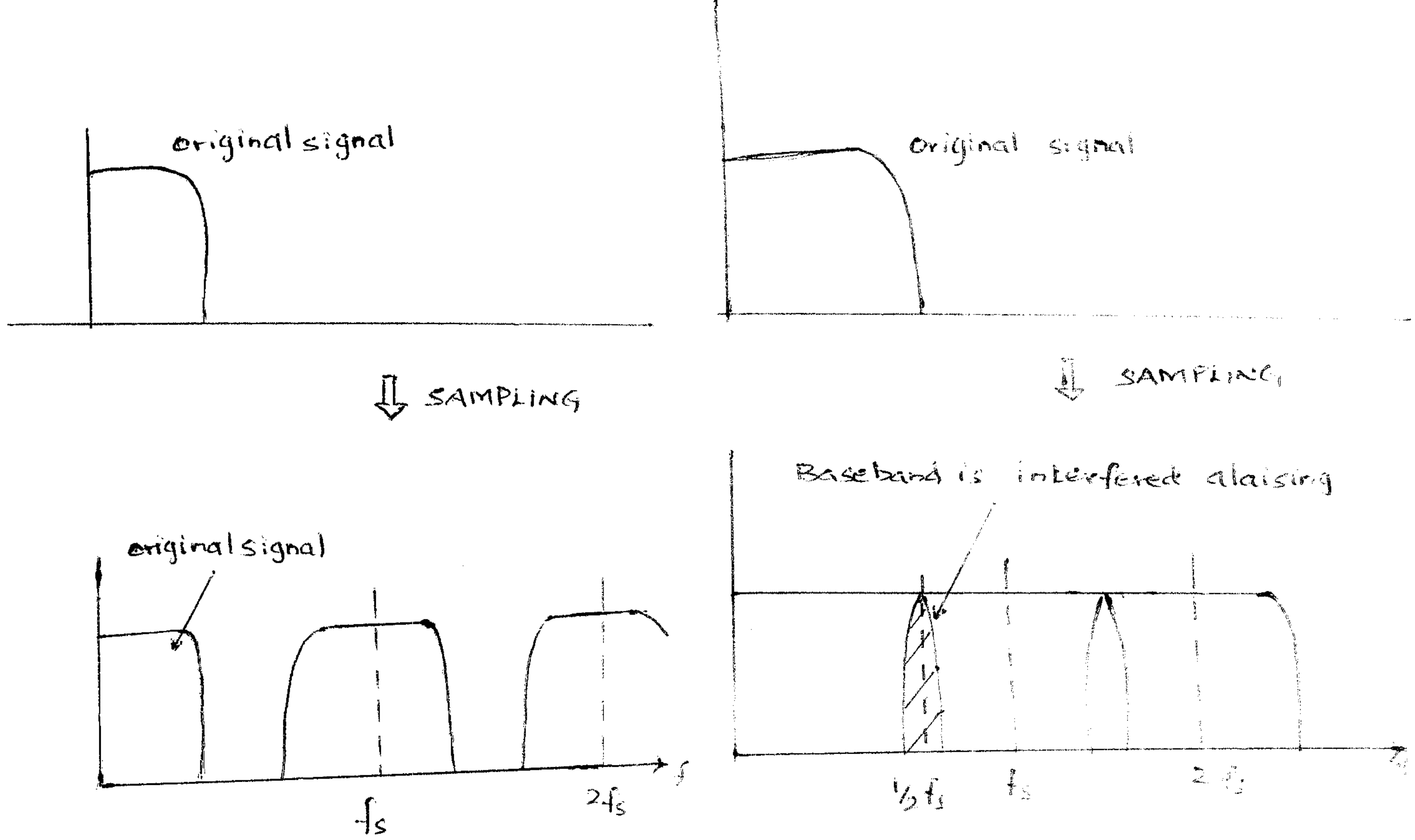
Sampling & Quantization

For the digitalization of an acoustic signal, the prevision of the regenerated wave is determined by both how fine the time is cut up (what is called sampling) and how precisely the height of the waveform is expressed in modulo 2 (what is called quantization) as in Fig. According to information theory the number of samples in 1 second (sampling frequency) must be at a frequency at least twice the highest frequency of the waveform one wishes to reproduce. That is, if the upper limit of the range of frequency to be reproduced is 20 Khz, a sampling frequency 40 Khz or more be selected.

To put it another way it is necessary to have an interval smaller than 1/40KHz = 25 msec between samples. The precision with which the height of the waveform is read is determined by the number of modulo 2 places (quantization). If the height is read with 3 places in modulo 2 (3bits), the range of strength of the signal is divided in to 8-divisions, and if it is read with 16 places in modulo 2 (16 bits), the range of the strength of the signal is divided into 2^{16} = 65,536 divisions.

That is since it is read with a scale for which the lowest value is 0 and the highest is 65, 536. The dynamic range is expressed as follows.

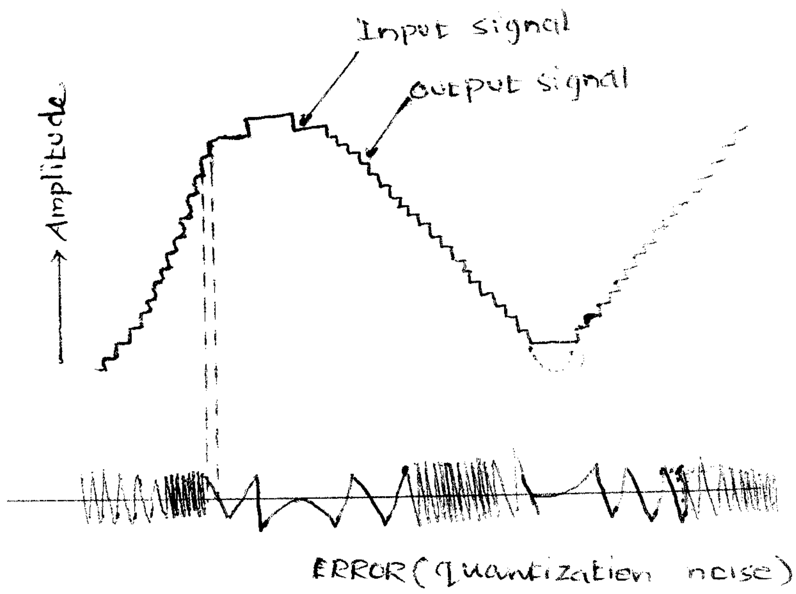
If we have a stereo acoustic signal (2 channels, Left and Right) with a sampling frequency of 40 Khz and a 16 bit quantization, the number of either on or off [illegible] a second of code is $40 \times 10^3 \times 16 \times 2 = 1.28$ bits/sec



a) original signal $< f_s/2$

b) original signal $> f_s/2$

Changes in frequency spectrum because of sampling f_s: sampling frequen



Quantization noise

The sampling frequency must be atleast 2 times that of the upper limit of the frequencies we have to record, but conversely if we fix the sampling frequency there can be no frequencies included in the signal we digitalize that exceed half of the sampling frequency. If frequencies of more than half of the frequency are included and then sampled, the modulated signal will overlap in terms of frequency when it is demodulated, the original signal cannot be brought out and distortion will be generated as shown in fig. This generation of distortion is called ALIASING.

The amount may be very small, but music and singing voices include frequencies above 20 KHz. However if the frequencies above 20KHz are cut, it is 99% inaudible, but if the cut-off is 18 KHz or 15 KHz, there is a suddenly distinguishable difference. If we work from this, a 20 KHz requirements for Hi-Fi systems can be though of as sufficient.

Next is the quantization number, and as was mentioned before, the dynamic ranges larger in proportion to the quantization number. Music and singing Voice have a dynamic range of 100 dB or more, and to cover all of this it would be necessary to have a quantization of more than 18 bits; however, it would be difficult to realize this in terms of technology and current products.

The dynamic range of current analog recorders is around [illegible] dB and thinking in terms of both that and engineering aspects, a dynamic range of 85-95 dB is a reasonable goal; therefore if we change this to quantization numbers, 14-16 bits can be thought of

as the requirement with quanization, there is always on small amount of error when the analog signal is measured only in limited modulo numbers as shown in Fig. This error is known as quantization noise.

If the quantization number is increased, the quantization noise becomes smaller, but even if it the quantization number is increased, there is a strong correlation between the signal and this noise. When the input signal's amplitude gets small. Rather then noise, it could be called a distortion near the higher harmonics of the i/p signal. This condition makes for an audible problem and the generation of higher harmonics is suppressed by adding noise to the signal on purpose. This noise added for this purpose is called dither.

The frequency band necessary for recording is 1-1.5 MHz, but the range of strength its only the width necessary for the detection of pulse or no pulse 20 dB is enough, but for safety under various practical conditions, 30dB is the criterion used. currently available record layers and tape recorders can obtain a rang of 70dB, but the frequency band is at most 20 KHz therefore they cannot be used a digital recorders without any changes. Luckily, however, the dynamic range of video tape recorders is 45 - 50 dB and the frequency band is 4-6 MHz , so if these are used digital recording is possible . We can say that the frequency band and dynamic range are more than enough. If there were no such video machines, digital audio would not have become a reality as quickly as it has.

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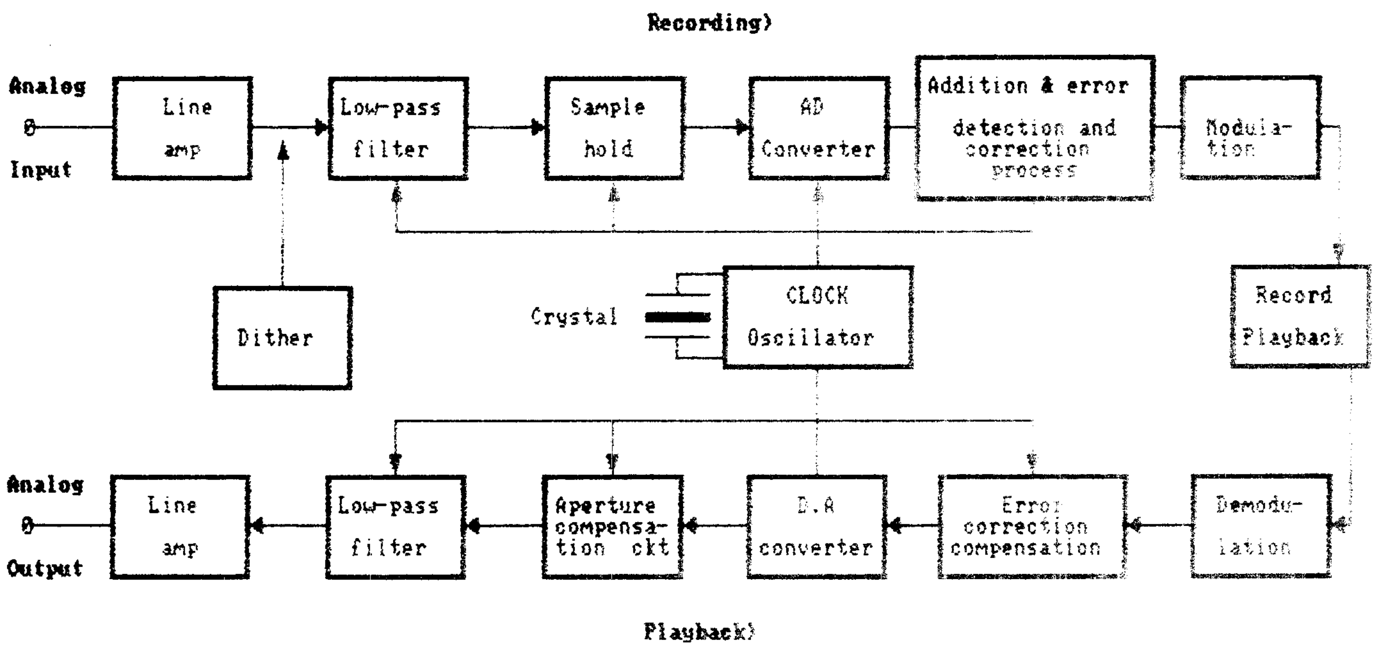
METHOD OF DIGITALIZATION :

We will look at the process of digital recording and play-back following the order in Fig.

(a) LINE AMP (RECORDING)

This is to change the level and Impedence of the input signal to facilitate the processes that follow. So it performs in the same way as a standard audio amplifier. The high frequency contents of the music and singing voices are small, so the high frequency is emphasized (pre-emphasis) at the time of recording, and the reverse operation (de-emphasis) is performed at the time of play as shows in Fig.



STRUCTURE OF DIGITAL RECORDING & PLAYBACK.

Effectively the dynamic range is enlarged. The emphasis circuit is built into the line amp.

(b) DITHER GENERATION CIRCUIT :

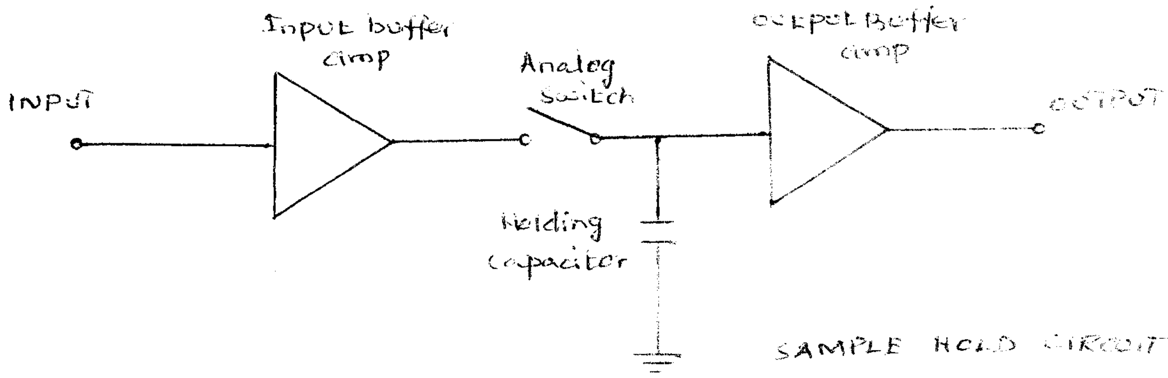
This is the circuit for generating the noise that is added in order to suppress the higher harmonic frequencies created by quantization noise when the signal is small as was mentioned before. Noise that has a frequency distribution contened near a specific frequency is added. Since even noise of the same amplitude it becomes less audible.

(c) LOW-PASS FILTER (RECORDING);

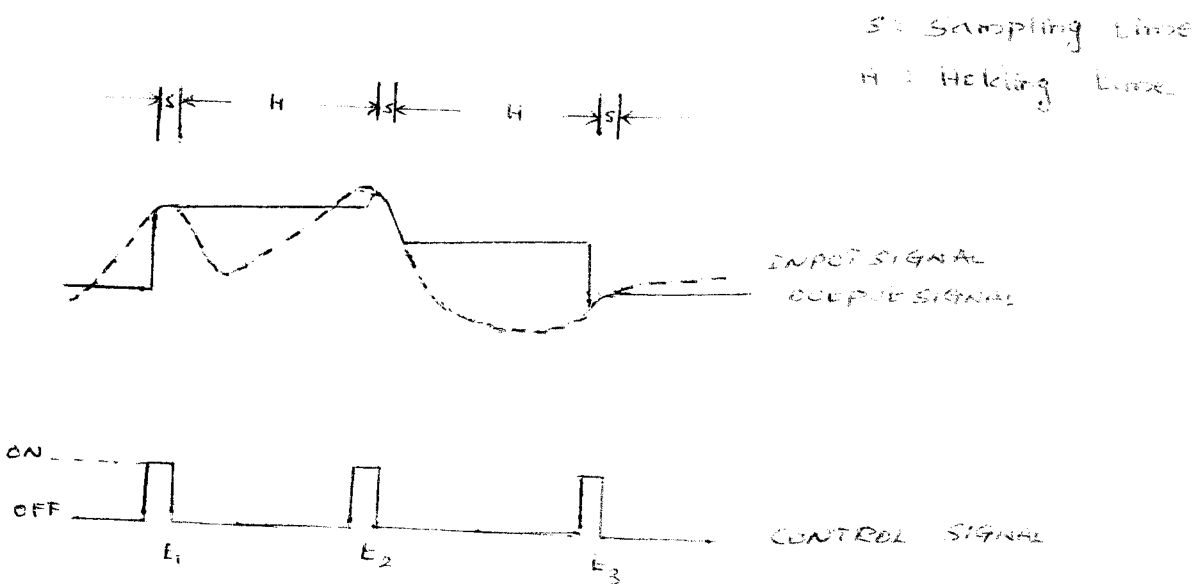
This is a filter that cuts out the extra signal above the frequency range to be reproduced in order to supress the aliasing. with a range of more than half of the sampling frequency. It is necessary to have about the same amount of attenuation as the dynamic range which is determined by the quantization member.

(d) SAMPLE HOLD CIRCUIT

It takes time, even if only a little, to sample and quantize the continuously changing acoustic signal. Because of this it is necessary to hold the value until the operation of AD converter is complete that role is played by the sample holding circuit. The actual circuit is constructed with buffer amplifier a switch and a capacitor as shown in Fig.



SAMPLE HOLD CIRCUIT



SAMPLE HOLD CIRCUIT SIGNAL WAVEFORMS

When the switch is closed it is sampling time, and the voltage across the capacitor changes according to the incoming signal so the output signal is same as the input signal.

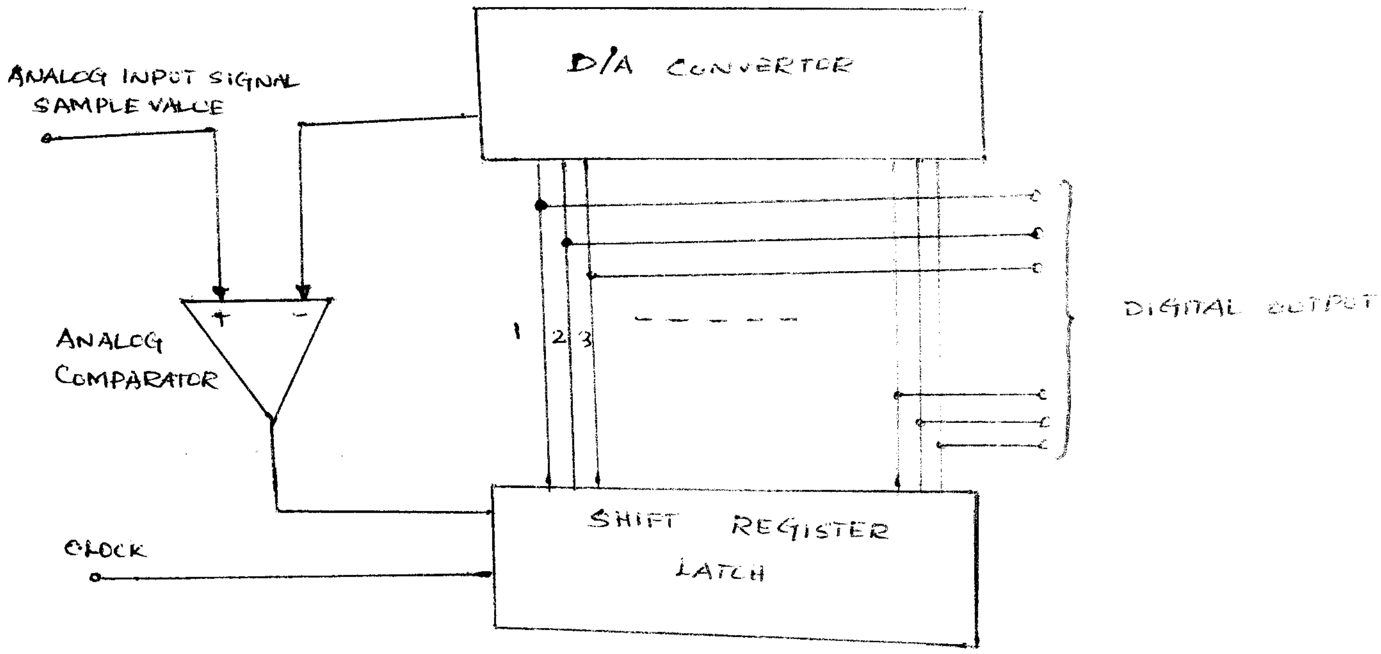
When the switch is open, the capacitor maintains a fixed voltage which is the value just before the switch was turned off.

(e) A/D Converter

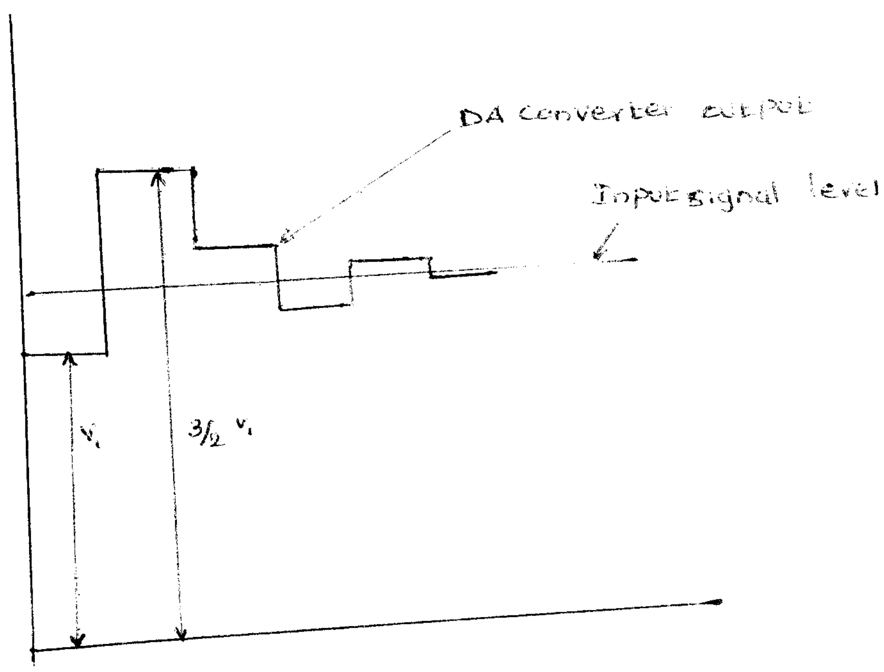
This is the part that quantizes each of the sampled values and changes an analog signal to a digital signal. For example, if it is 16bit AD converter it must have the capacity to divide the sampled value into 2^{16} = 65,536 gradations and must accomplish the conversion in a short time, [illegible] ms. There are design for various types of AD converters, and as a typical examples we will introduce the successive approximation type.

The circuit is made up of a DA converter, a shift register and latch, and an analog comparator as is shown in Fig.

When the sampled value from the sample holding is placed across the I/P leads of the analog comparator, the first bit of the DA converter is set to V1. At this time (the sampled value and the o/p of the DA converter V1 are compared, and if the former is larger, the first of the shift register is latched at 1, and if the later in larger, it is latched at 0. Next the second DA converter bit is set to 1 by the clock. At the time if the first bit from the DA converter remains set at 1 because of the latch that resulted from the previous time, the o/p voltage of the DA converter becomes (3/2) V1 and this value is compared



Successive approximation type AD converter



operation of successive approximation type AD converter to the sampled value. If the first bit of the shift register has set to 0, two o/p voltages becomes 1/2 V1, and it is compared by the comparator. After that the same procedure is repeated and the o/p voltage of the DA converter gets closer to the i/p voltages as shown in Fig.

(f) CODE ERROR DETECTION / CORRECTION CIRCUIT:

With tape recorders the tape is not always in uniform contact with the head because of in uniform contact with the head because of dirt stuck to it or scratches, and there is signal loss at times. Even with disc players, the same phenomenon is produced. This is called dropout and it is frequent a large noise in generate or the sound completely disappears. In order to prevent this, it is necessary to detect the lost section and till it is. To do this, a signal error correction code related to the main signal is added, and there is an operation in which the recording order of the pulse to code is interleaved according to a certain rule. Other factor that general code errors are Dropout and jitter (deviation of the digital signal along the time axis). Interference between codes generated when the recording or playback bandwidth is insufficient, intersymbol interfence of recorded bit patterns) and noise (hiss of the tape crosstalk between heads, etc.)

(g) MODULATION CIRCUIT

The characteristic of tape and disc system that make use of digital recording deteriorate below several , and reproduction from direct current cannot be hoped for. If a code of '1' of '0' is lined up for a long stretch "1111.......' Or "0000......" in a machine with this kind of characteristic) it comes close of a direct current and playback is difficult. Also, during playback, when bits must be accurately distinguished, mistakes in adding up the number of 0's or 1's are easily made. In order to eliminate this inconvenience, the encoded pulses are not recorded directly, and a rule is devised for rearranging the pulses so continuous 1's and 0's are eliminated. The work of the rule is called modulation itis carried out by the modulation circuit.

(H) DEMODULATION CIRCUIT :

From here we enter the playback operation first, the recorded signal is read off the disc and it is processed in the reverse order of recording. The signal that has been modulated according to a certain rule is return to its original form.

(i) ERROR DETECTION, CORRECTION / COMPENSATION CIRCUIT

Code errors caused by dropout and other factors are detected and corrected, and when they cannot be corrected, they are compensated for the interpolation from the adjacent signals before and after the error. Also, the time variations produced during recording and playback by mechanical vibrations.

Rotational variations etc. are absorbed by the stable synchronized signal obtained from the digital memory and the crystal oscillator; therefore the precision of the crystal used in this oscillator is extremely important.

(J) DA CONVERTER

The digital signal in which the error have either been corrected or compensated for is converted to an analog signal. This is the function of this circuit.

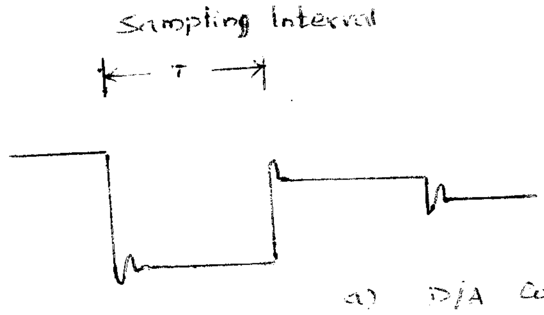
(k) APERTURE COMPENSATION CIRCUIT :

It takes time for the converted analog signal to reach a valve that corresponds to the digital input. As is shown in Fig.

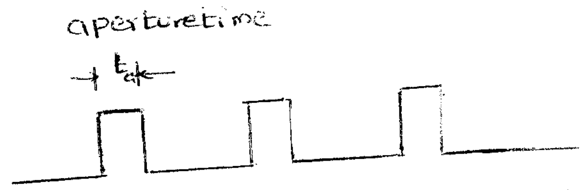
(1)LOW - PASS FILTER (PLAY BACK)

The output signal from the aperture compensation circuit includes many high frequency components that are not part of the desired signal. Since these components are outside the audible range, there would be no large effect even if they were not removed by filter.

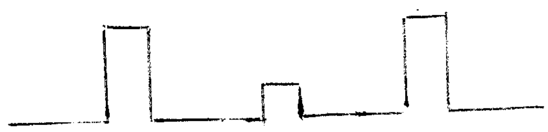
However, in the circuit that follows, even a small amount of non-linearity produce intermodulations so a filter is inserted in order to obtain a high quality acoustic signal.



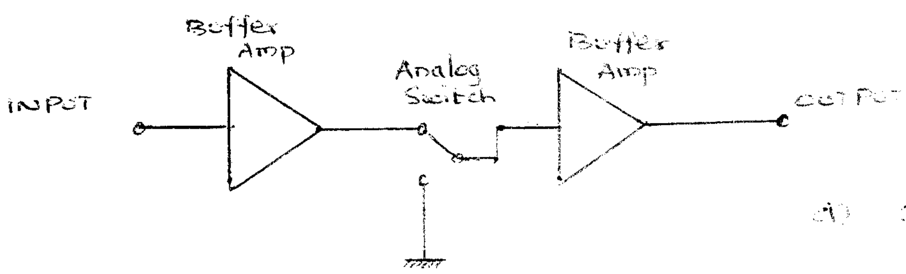
a) D/A Converter output



b) Analog Switch control Signal



c) output waveform from aperture compensition circuit



d) circuit

APerture compensition circuit.

(m) LINE AMP (PLAYBACK)

This is included to adjust the level and impedence for input to outside circuits, and it perform in the same way as a standard audio amplifier. The de-emphasis circuit is also built in to this line amplifier.

In compact Disc's signal are recorded in PCM format at The sampling frequency of 44.10 KHz.

Advantages of Pulse code modulation.(PCM)

1. Power consumption is less.

2. Easier to store the message because of the easier memory available.

3. Information can be secret (or) privacy can be maintained.

4. Because modulating and Demodulating are digital circuit, high reliability is possible and suitable for IC logic design.

5. Unnecessary information is avoided.

Disadvantages.

Circuit is complex compare to analog type.

THE CHANCE FOR DIGITALIZATION

The frequency and dynamic ranges in the music and singing voices we enjoy change with type of music, the differences of men and women voices etc. The fig. shows the average of result of the Measurement of frequency distribution for various music program of broadcasts over long periods of time where each category contains Measurement on several to twenty or more programs.

There is a large difference in the low frequency range and high frequency range. For each program, but in general the frequency distribution for the program sound gradually decreases above 2 KHz.

In terms of dynamic range, with Measurements of peak level on 36 piece orchestra, the highest was a sound pressure level of 120 dB and the average was 114 dB. In popular music since the arrangements is for each instrument and singer to have a separate microphone and each one is at an extremely short distance from the microphone. The peak level reaches 130 - 135 dB. On the other hand, lower limit of dynamic range is determined by the studio noise, the vibration and air currents picked up by the Microphone, the noise originating in induction and the noise from the preamplifier for the Microphones.

There are many low range components in frequency distribution. Therefore if we pay attention to frequency and sound pressure level. We can say that the range in which acoustic sound exists is a range like that in the summary Fig

By the way, with the current software made by collection of acoustic signals and mixing - records, tapes. FM broadcasts - it is very difficult to secure this range of sound pressure level and frequency.

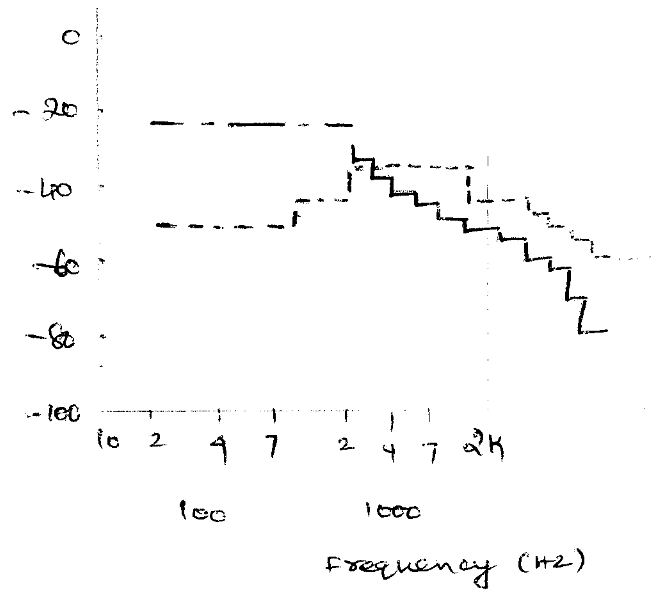
Even with the stereo tape recorders with a speed of 38 cm/s that are used commercially it is as shown in Fig.

And LP record players with software with the base sound qualities we can expect to obtain give us something like Fig.

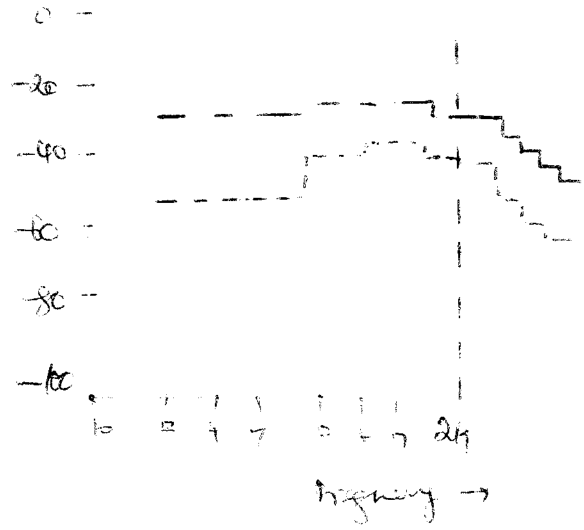
In each of these, even through there has been improved with developments in materials and production methods of tape and record, compression and expansion of the signal, use of equaliza- tion and improvements in recording and playback heads cutters and cartridges.

It is absolutely impossible to cover the rang in Fig.

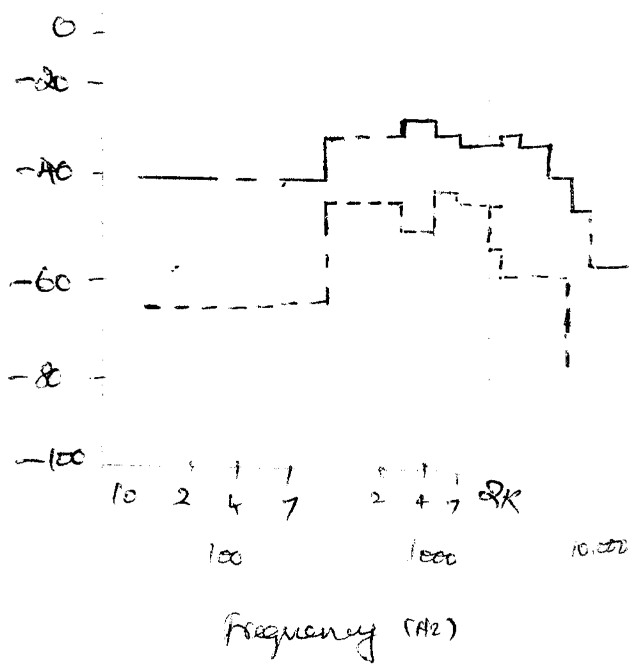
This is especially evident in the range of changes in dynam- ic range and the recent tendency toward increasing playing [illegible] has the effect of reducing dynamic range which makes it seem a step in wrong direction, if looked at from this point of view.



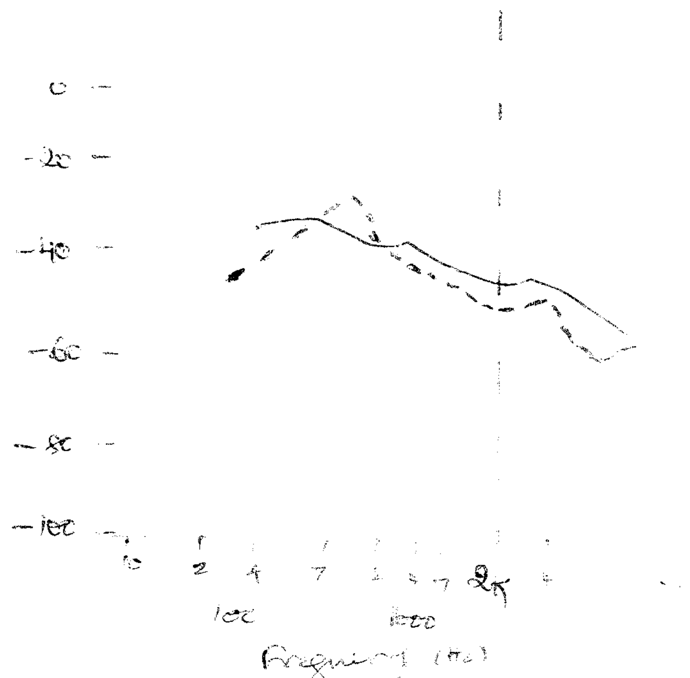
(a) Western Music



(b) Light Music

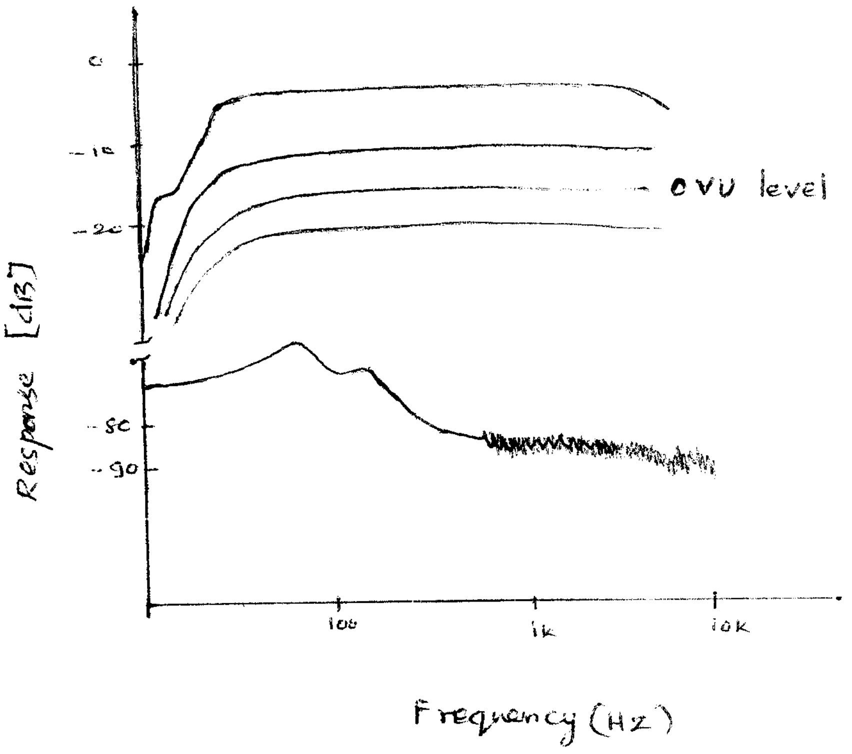


(c) JAPANESE MUSIC

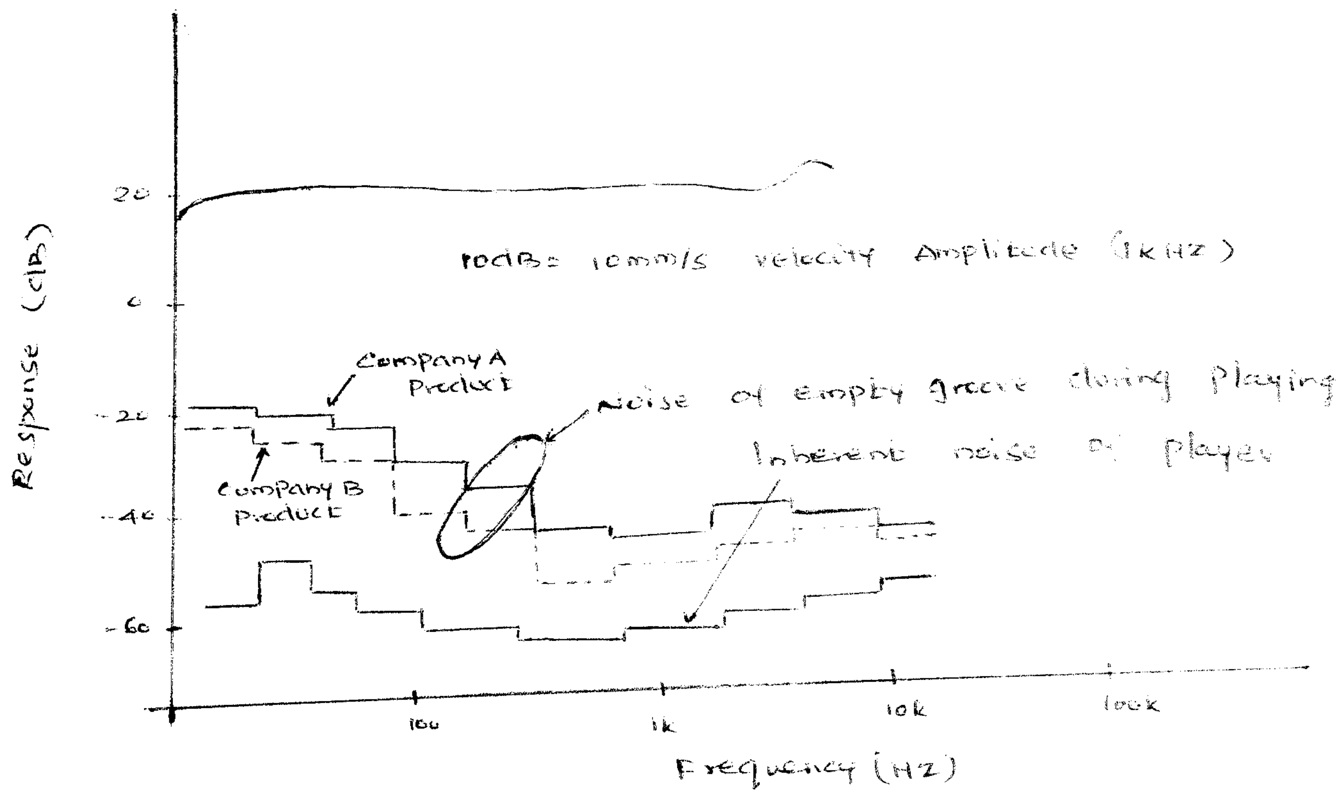


(d) SPEECH (male / female, average for 4 speakers?)

AVERAGE SPECTRA OF VARIOUS TYPES OF BROADCAST PROGRAMS.



Response of industrial tape recorder with a tape speed of 38cm/s



Response of high quality LP record player

What is digitalization

First of all it is impossible to put an acoustic signal with time based changes in strength that go over 110 dB in to the variations of magnetic flux of a tape or the very small changes in the width of a groove in a disc. With analog methods that have been used upto now, the upper limit of signal strength is determined by the magnetic saturation in the case of tape and by the width of the groove in the case of record; on the lower side, the range is made narrower by the noise of the tape or the record. In order that there be no influence from distortion and noise in the recording medium, the continuous acoustic signal is converted in to morse code-like arrangement of long and short signal, or into the presence or absence of pulses used in computers. Even if there is noise (or) distortion between the recording and playback, the idea is that the original acoustic signal will reappear as long as presence or absence long a short can be distinguished on the reproduction side.

Also, since the space between the pulses is electronically controlled, the rotational variations generated by the motor are taken out.

To express the value of an acoustic signal, which might for example be read as 12V on a meter, in a code of pulse or no pulse, we must use the '1' and '0' of module 2 not the module 10 in which we are used to counting its.

A example of the relationship between module 10 and module 2 is shown in table.

A example of the relationship between module 10 and module 2 is shown in table.

Table : Correspondence, between 10 and module 2

Module 10	Module 2
0	0
1	1
2	10
3	11
4	100
5	101
6	110
7	111
8	1000
9	1001
10	1010

and o and 1 of module 10 can be expressed in one decimal place in module 2, while to express 8, Four places are necessary. The wave form of the analog signal is cut up of appropriate time intervals as is shown in Fig 2.14, amplitude read at these times in modulo 2, and digitalization is completed by lining up a series of correspondent pulses. In this case one place in modulo 10 is called a digit, and one place in module 2 is called a bit. Furthermore, the music signal touches both the negative and positive ranges, and a methods for showing negative and positive are various as is shown in Table.

Various binary systems

	binary	2's component	Folded binary
-7	1111	0111	0111
-6	1110	0110	0110
-5	1101	0101	0101
Similarly			
+7	0001	1001	1111
+6	0010	1010	[illegible]
+5	0011	1011	1101

but the CD systems uses what is called the "2's complement" method of expression in which the cases where all two bits are either 0's or 1's are close to zero.

Characteristics affecting sound Quality

The characteristics that affect sound quality are various and can be brought together as follows:

(a) FREQUENCY RESPONSE :

A uniform frequency response from direct current up to the low - pass filter cut - off frequency for [illegible] 11 KHz [illegible] obtained. However, inorder to prevent aliasing, signal components above that frequency cannot be recorded.

(b) DYNAMIC RANGE :

It is proportional to the quantization number N, and a dynamic range of (6N + 1.8)dB which covers the entire reproduction frequency range can be obtained conversely, the components near the upper and lower limits of the dynamic range of singing voices and music (vary with mixing levels) are cut out at the time of quantization and cannot be reproduced . One method for widening the dynamic range is called non - linear quantization in which the quantization steps are made larger when the amplitude is large.

(c) CHANNEL SEPARATION :

A value the same as the dynamic ~~sample~~ range can be obtained between the left and right channels.

(d) DISTORTION NOISE :

The principle one is quantization noise. Also the noise and distortion from the DA converter, low pass filter, line amplifier and other analog circuits are added to this, but the value is less by a factor of ten or more when compared to the analog recorders available up to now. However, when code errors are produced by drop - out, jitter, etc as long as they can be corrected, they must be compensated for, thus create distortion. If they go over the compensation limit, noise is generated.

(e) WOW AND FLUTTER :

These are reduced to the level of the precision of the crystal oscillators that produces the clock frequency (in practice 0)

Generally if the sampling frequency and the quantization number are determined the reproduction frequency and dynamic ranges determined, and within those ranges, high quality sound with extremely low wow and flutter and distortion can be obtained. However, any information in the acoustic signal outside that range is cut out at the time of digitalization and reproduction is impossible. It would be ideal to make that range wider, but the frequency necessary for the recording medium becomes necessary. According to use, the way in which the sampling frequency and the quantization number are determined takes on an extremely large significance.

Characteristics affecting operability :

a) Operation :

Since various control signals can be put between the audio signals in the digitalization process automatic starting and stopping as well as choice of songs and various other operational conveniences can be conceived of.

b) SHAPE DIMENSIONS :

Recording and reproduction are done through high density recording process, and the steps of digital processing are carried out with LSI's. So all components can be smaller and lighter than their analog counterparts.

c) DUBBING :

Even if recording and reproduction are performed over and over again, there is no deterioration of response within the limit of error generation with analog equipment, each time the recording reproduction process is repeated, the dynamic range narrows by 3dB, and wow and flutter can cause additional deterioration of the sound quality.

‘CD’ Recording

With analog equipment. It is easy to use out a section of magnetic tape by hand and interchange it with tape form a different recording, but with digital recording, this is difficult because information is lost at the cut and noise is produced. Electronic editing making use of several recorders and electronically mixing their outputs onto a new tape must be employed.

U matic tape Recorder :

U matic tape Recorder is a rotary head tape recorder used for CD mastering. Mastering is the process of making a glass masker and nickel stamper.

Pre - mastering :

It is the preparation of an edited master tape to be used in the recording process.

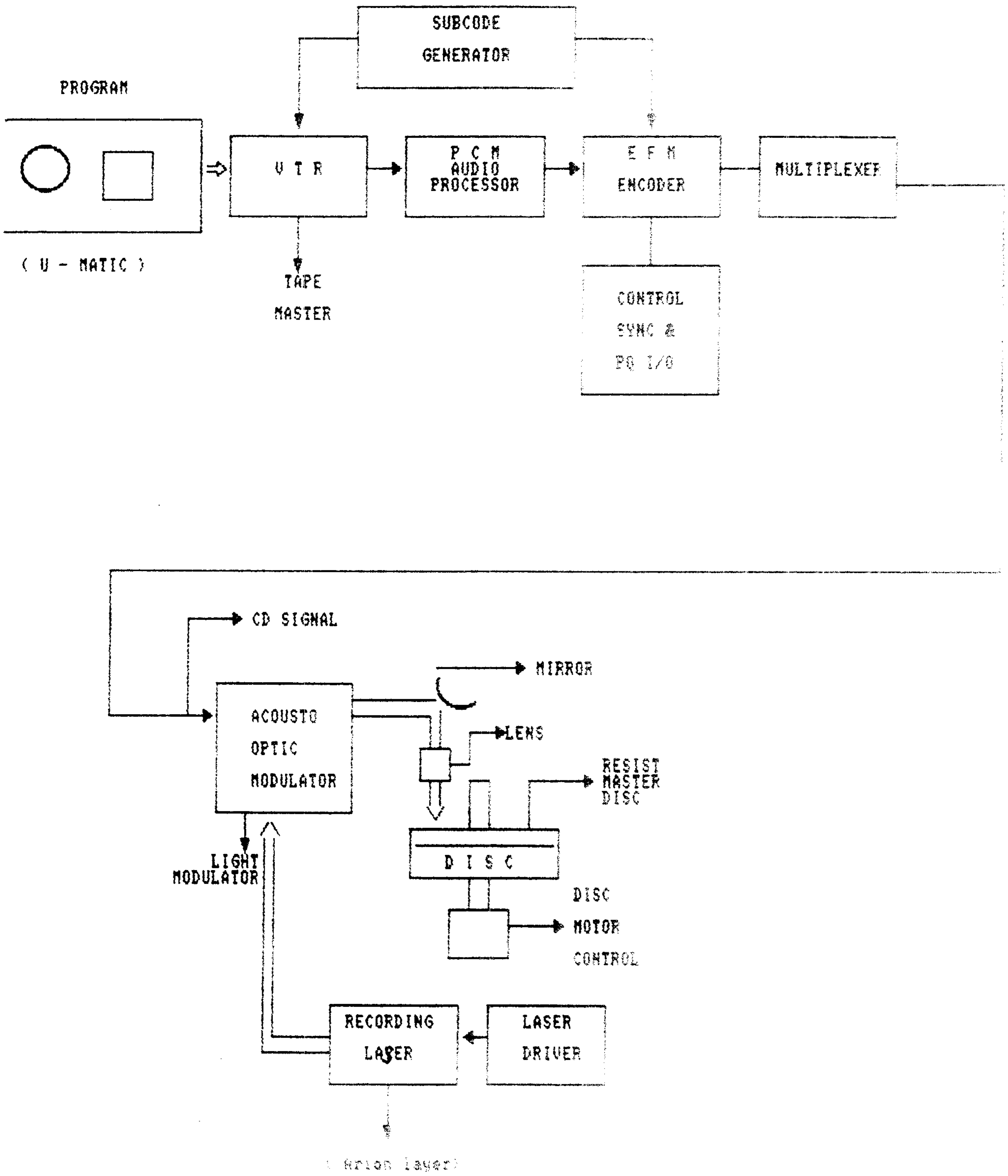
In accordance with sony Red book standards the audio program must fall into one of these categories

(i) regular stereo

(2) one mono signal, simultaneous encoded on both audic channels and

(3) one mono signal on one channel and no signal on the other channel.

Thus the use of two audio channels for different programs is not allowed.



BLOCK DIAGRAM OF LASER RECORDING SYSTEM.

Thus the use of two audio channels for different programs is not allowed.

In pre-mastering the original master tape is copied to a specially prepared tape master, which is used to produce the master disc prior to replication. It is the first step in CD replication.

The original tape of any form is referred to as the master tape. By means of a DAP (Digital audio processor) the audio program is copied by a 3/4 - inch U-matic tape.

Rules for CD master tape preparation.

1. Sampling rate must be 44.1 Khz

2. Tape format must be 3/4- inch U-matic NTSC standard.

3. Time code must be SMPTE 60 HZ non drop frame time code.

4. Time code must be synchronized to the NTSC video signal.

5. Time code should continuosly and uninterrupted code should not cross over 00 hours, 00 minutes, 00sec, 00 frames.

6. Time code should be recorded on analog track # 2 of the U-matic tape.

7. Recorded digital mute with time code for a min of 1 minute before the first track and 1 1/2 min after the last track.

8. A min of 2 seconds must be allowed for emphasis charge between tracks.

9. Prepare a list that is frame accurate to locate beginning and ending points.

10. Prepare a guideline of spurious noise information which indicates nay unusual noises or general notes regarding the quality of the recording.

VTR :

The next step in the pre-mastering process turns the CD master tape into a CD tape master (VTR) ready for mastering. Music programs are edited in the 3/4 in videc tape cassette.

Subcode editing :

Using the CD subcode processor/editor the engineer enters the preliminary required data that is requested by the system's menu:

(1) album title
(2) artist
(3) record label
(4) catalog number
(5) UPC number
(6) Analog or digital source
(7) mastering engineers name.

PCM Processor

The signal output form VTR is fed to PCM audio processor.

EFM Encoder:

The PCM processed audio data is encode by an EFM encoder. The encoded data is multiplexed by a multiplexer.

The multiplexed signal is fed to Acoustc optic modulator. (Light modulater). This modulator uses ultrasonic waves is employed to modulate laser light.

The recording laser should satisfy the following requirements.

(i) Capability of producing a small spot.

(ii) Sensitivity to photosensitive material.

(iii) Continuous - wave emission and light source stability.

Ar ion laser (457.9 Nm) and He cd laser (441.6Nm) are suitable for recording. The modulated CD signal (optical signal) is transferred to the glass master where it will be repressed as pits.

DISC MANUFACTURING :

The injection moulding is widely used in CD production because of its high productivity.

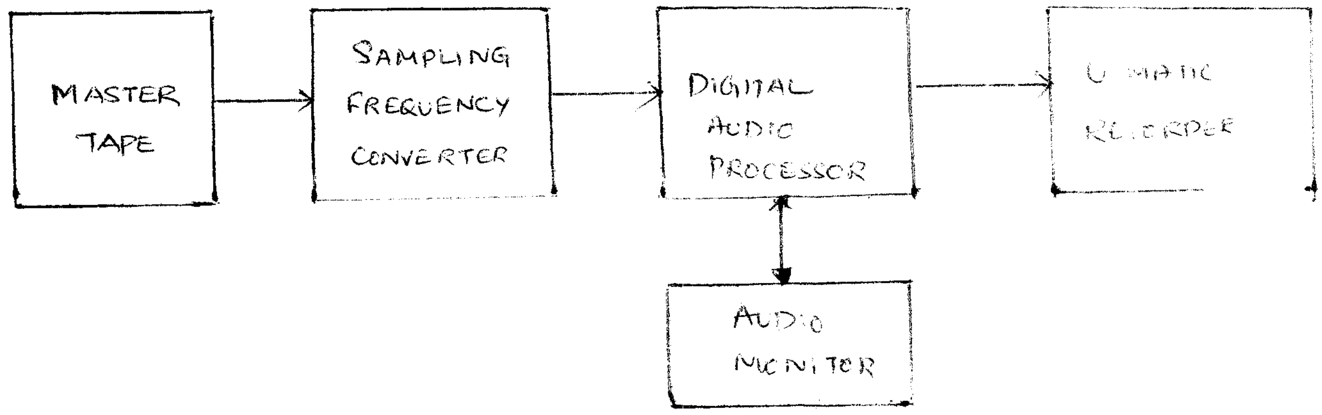
Since the moulding process needs a high temperature and high pressure a glass master cannot be used. The information pits must be transferred to a nickel stamper.

The sequence of operations are diagrammatically shown below. Replication is the process where by many CD replicas are produced from the stamper the flow sequence is shown below.

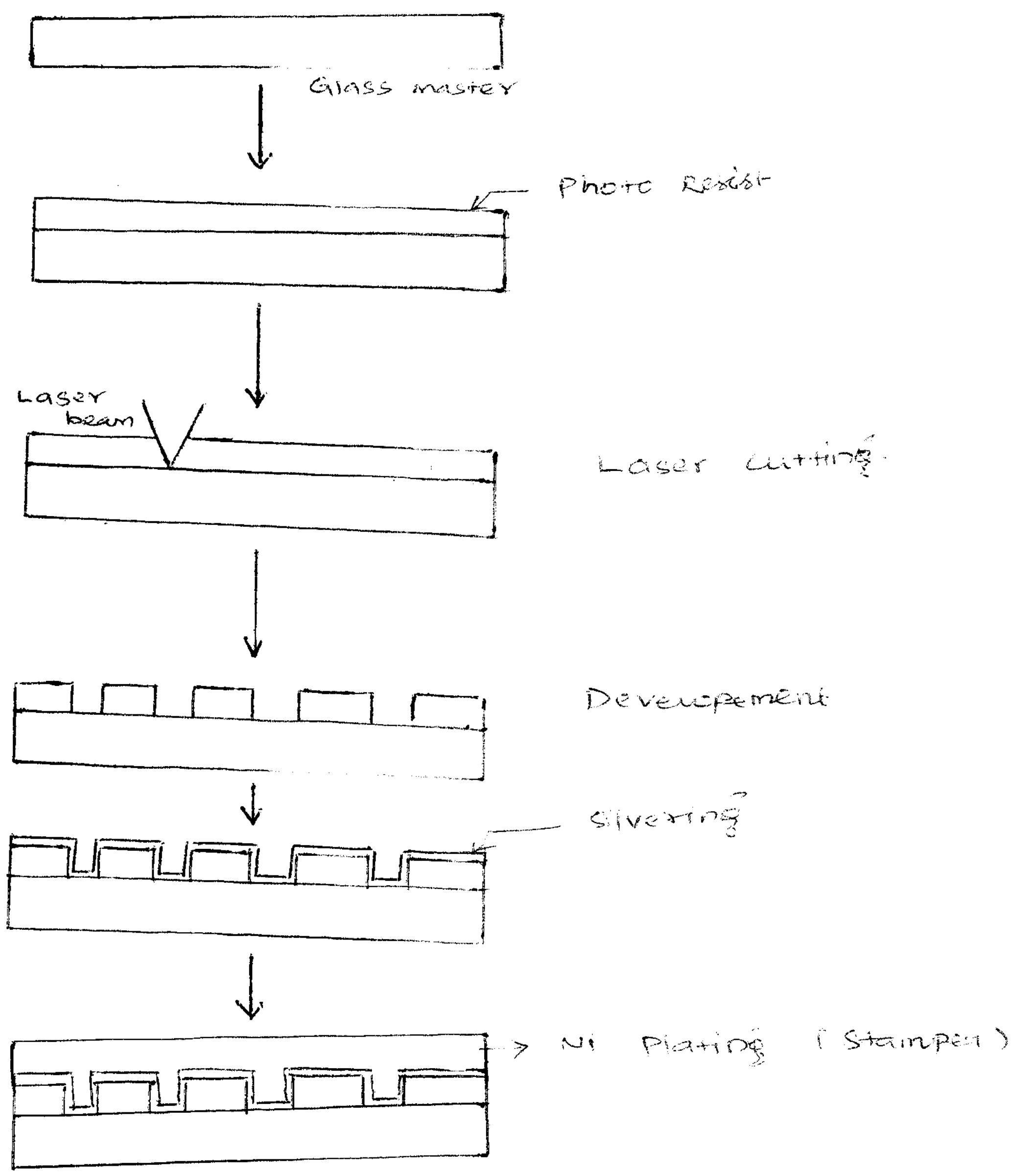
There are about 5 Giga bits of information on the signal surface of a CD stamper. The moulding process is designed to make many good replicas with high productivity.

After moulding a layer of metal aluminum is evaporated into the pit surface. Subsequently a protective layer is spun on to the reflective layer. This protective layer consists of ultraviolet light curing resin about ten micrometer thick.

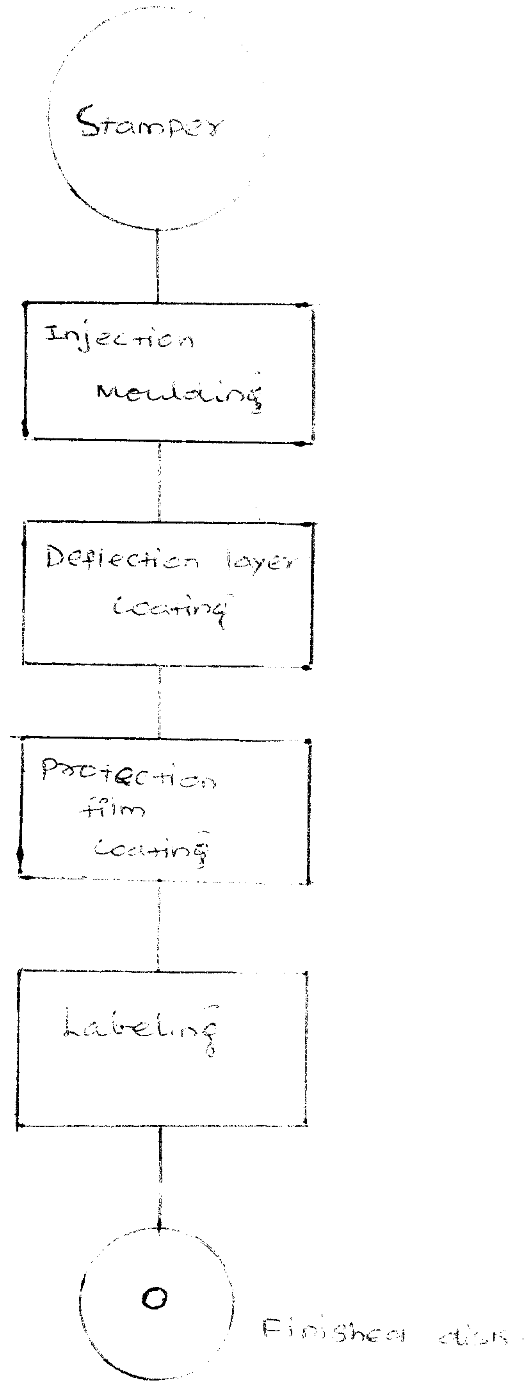
The label is printed on the protective layer. The compact disk is now finished and ready for shipment after final inspection.



BLOCK DIAGRAM OF U-MATIC RECORDING SYSTEM.



DISC MANUFACTURING



REPLICATION FLOW SEQUENCE.

Error Correction & Control

Unlike an error in computer data, an error in digital audio data (if the error can be detected) can be concealed. Indeed, simple linear interpolation is sufficient in most cases. Th error correction was used in a CD system must satisfy the following criteria.

1. Powerful error correction capability for randon and burst errors.

2. Reliable error detection in case of an uncorrectable errors.

3. Low error detection in case of an uncorrectable error.

CIRC satisfies these criteria and can control errors on the disk properly.

BASIS ERROR CORRECTION CODE :

The basic error correction procedure is shown in Fig.

A group of data is transulated into a code word by adding check data and transmitted through the recording channel. At the receiver side, received data are compared with all the code words, and the nearest are selected.

If a group of K symbols (the data) is encoded to a longer word of n symbols (the code word) and the code words satisfy special check equations, then this code is called are (n,k) linear block code. The encoding process is in other words, a process of assigning as much nonparity check data to the original data.

Error correction and control technique :

This section describes the error control technique and the CIRC error correction code : its construction and performance.

Necessity of Error Control :

The CD system employs the optical noncontact readout method. Because the signal surface, is protected by a plastic layer and the laser beam is focussed on the signal surface, the disk surface itself is kept free from defects such as scratches. As a result, most of the errors which occurs at and in the vicinity of the signal surface through the mastering and manufacturing process are random errors of serial bits. Even through the CD system is resistant to finger prints and scratches, defects exceeding the limit will natenally cause large burst errors.

A typical bit error rate of a CD system is 10^{-5} which means that a data error occurs 2×10^{6} bit/sec $\times 10^{-5} = 20$ timer per second such data errors, even though they may be 1-bit errors, cause unpleasant pulsive noise; so an error correction technique must be employed.

For ex, Suppose x = (X1, X2,............XN and

y = (Y1, Y2,............YN, are

code words an in fig: than the Hamming distance betweer the two code word is defined as the number of different pairs of symbols.

It t symbol errors induced in the channel are not to lead to confusion at the receiver side as to whether x or y was transmitted, z and y should differ from each other (as in fig) by atleast (2t + 1) symbols.

Therefore a fig. of merit of the code called minimum distance d is defined as the min distance among all pairs of different code words z and y. A code is t- error correcting if and only if $d \geq (2t + 1)$; and if the location of the errors (erasure location) are known (d-1) erasure correction is possible. If the number of errors exceeds these bounds, error correction and defection capability are no longer guaranteed and the decoder may make an erroneous decoding.

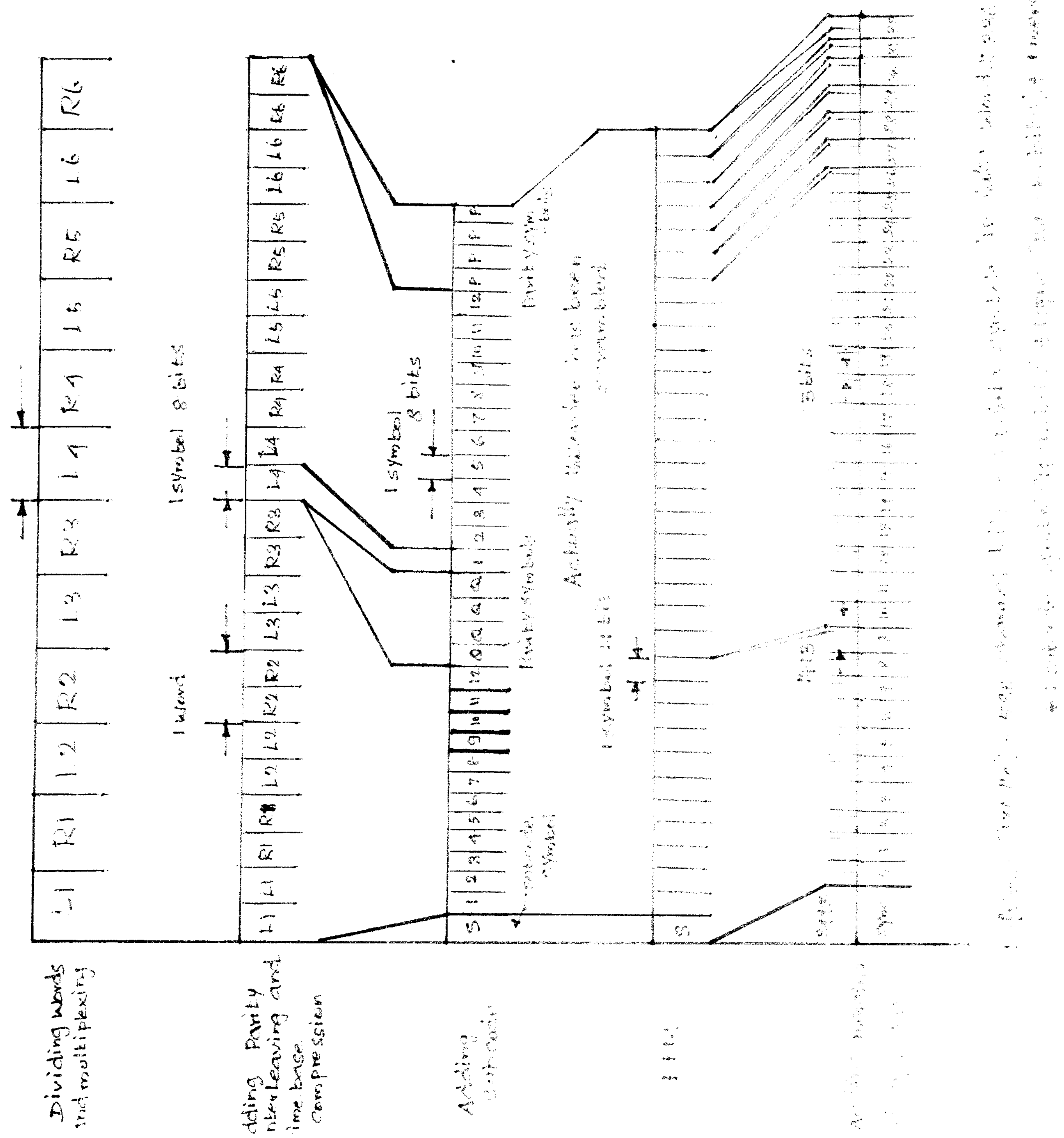
DATA ENCODING

Compact disc encoding is the process of placing audio data in a format suitable for storage on the disc. A frame structure provides such a format. The frame furnishes a means to distinguish the data types : audio data and its parity, synchronization word. and subcode. The information contained in a CD frame (prior to modulation) contains a 27-bit synchronization word, 8-bit subcode, 192 data bits, and bits. CD frames are assembled when the master disc is encoded. Assembly of the frame involves several processing steps as well as modulation and the addition of merging bits. The complete encoding sequence is shown in figure.

Encoding begins with the audio data. Six 32-bit PCM. audio sampling periods (alternating 16-bit left and right channels) are grouped in a frame the left channel proceeding the right. Each 32-bit sampling period is divided to yield for 8 bit audio symbols. (The original 16-bit number is called a word, and it is split into two 8-bit symbols.) The audio program may consist of stereo or dual monaural left and right channels. Although some recordings have use independent left and right monaural channels, the CD standard does not support this.

After grouping audio data into symbols, error correction encoding takes place. The CD system, of course employs the cross interleave Reed Solomon code (CIRC). As we've seen, it use a combination of interleaving and parity to make the data more robust against errors encountered during storage.

Complete CD Encoding Process



SUBCODE

Following CIRC encoding an 8-bit subcode symbol is added to each frame. The eight subcode bits are designated as P,Q,R,S,T,U,V, and W, some times referred to as the PQ code. The CD player collects subcode symbols from ninety eight consecutive frames to form a subcode block with eight 98-bit words. Only the P and Q bits used in audio CDs. Included is information specifying the total number of selections on the disc, their beginning and ending points and timings, the index points within a selection, the program lead-in and lead-out point and updated information on the pickup's position as the disc is played. The other 6 bits R S T [illegible] available for encoding other information on audio CDs subcode is discussed in detail later in this chapter.

EFM Modulation

After the audio, parity and subcode data are assembled, the bit stream in modulated using eigth - to - fourteen modulation (EFM).

Modulation method

This section describes how CIRC encoded data are modulated and recorded on disk.

Basic requirements for a modulation method.

CIRC encoded data cannot be directly recorded on the disk because they don't satisfy the following requirements for optical recording; thus, EFM modulation is employed. The basic requirements are;

1. Correct readout of high density recording:

The frequency characteristics of the CD system are basically the same as those of a LPF, and the cutoff frequency is $\frac{2NA}{\lambda}$.

Therefore to avoid ISI (Inter Symbol Interference caused by a limited bandwith, 7min (Minimum run length of the channel data) must be as larger as possible on the other hand, if the detection window Tw is too small, the deterioration of the signal caused by such factors as lens aberration, defocus, and skew may cause random errors. As a result, a large 7min and a proper T_w min are preferable.

2. Clock Content:

A bit clock which must be regenerated from the readout signal edge is used for demodulating the data and for CLV control. To maintain clock stability the modulated signal must have a sufficient number of transitions, and 7max (the maximum run length) must be small.

3. Low Frequency Components:

If the modulated signal contains no low-frequency components, low frequency disturbances caused by dirt and scratches on the disk surface or asymmetry of the readout signal can be removed easily. Also low-frequency components leak to the servo loop, making the servo unstable.

Error propagation

The error correction system employed in the CD system treats 8-bit symbols as a unit; thus, 8 bit unit modulation without error propagation is preferable.

EFM (Eight - to Fourteen Modulation).

EFM is a run-length-limited (RLL) code developed for the CD system. First, 8-bit data sre mapped on a 14-channel-bit pattern.

EFM is a process which converts 8 bits pattern into 14 bits pattern to prevents Rapid transitions.

For 14 bits pattern, there is 16384 combinations and only 256 combinations are chooses to satisfy the condition which not fever than 2 zeros and no more than 10 zeros Between 1s otherwise rapid transition occurs.

Here where 1 means 'transition' and 0 mean' no transition'.

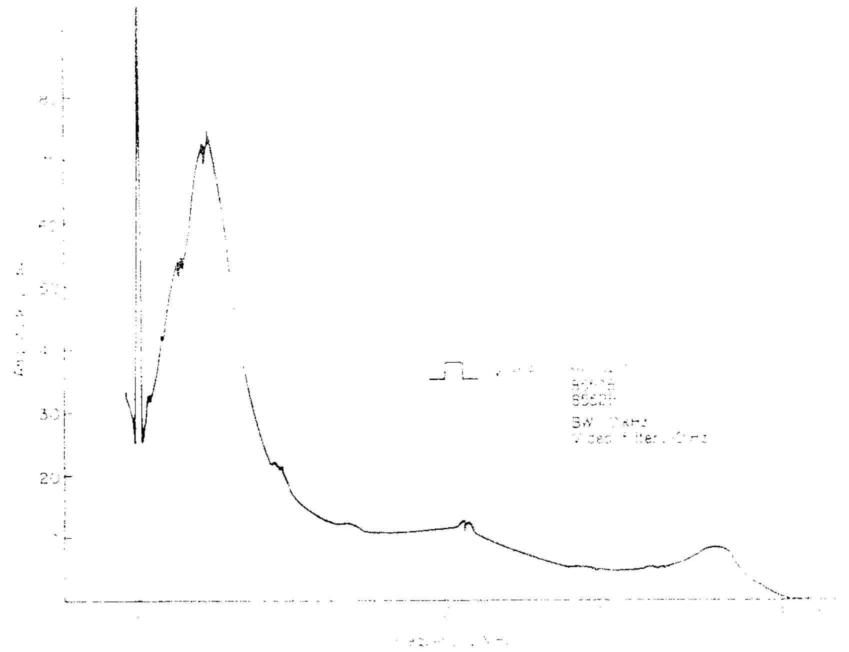


FIG. 9.14 Power spectrum of EFM.

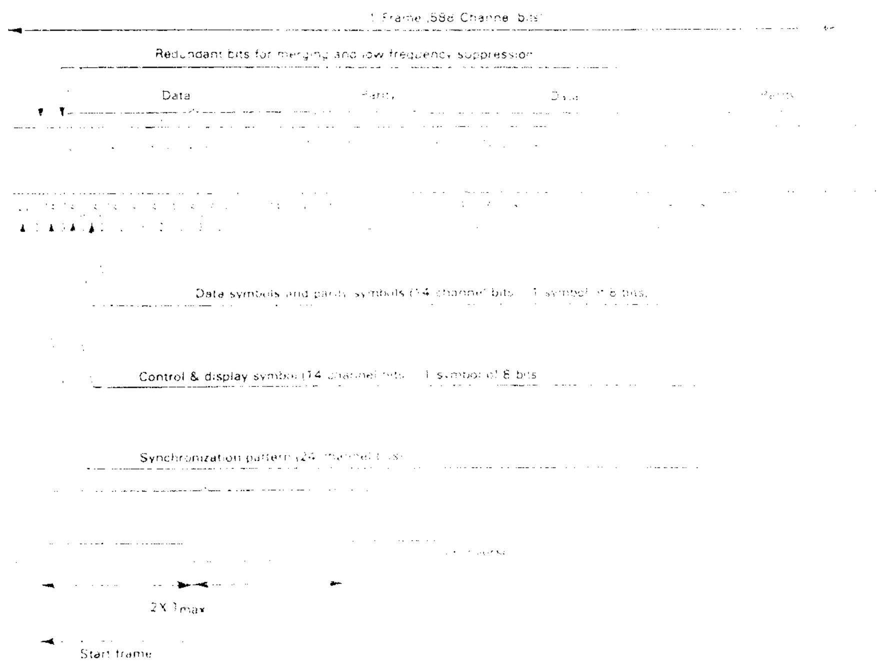
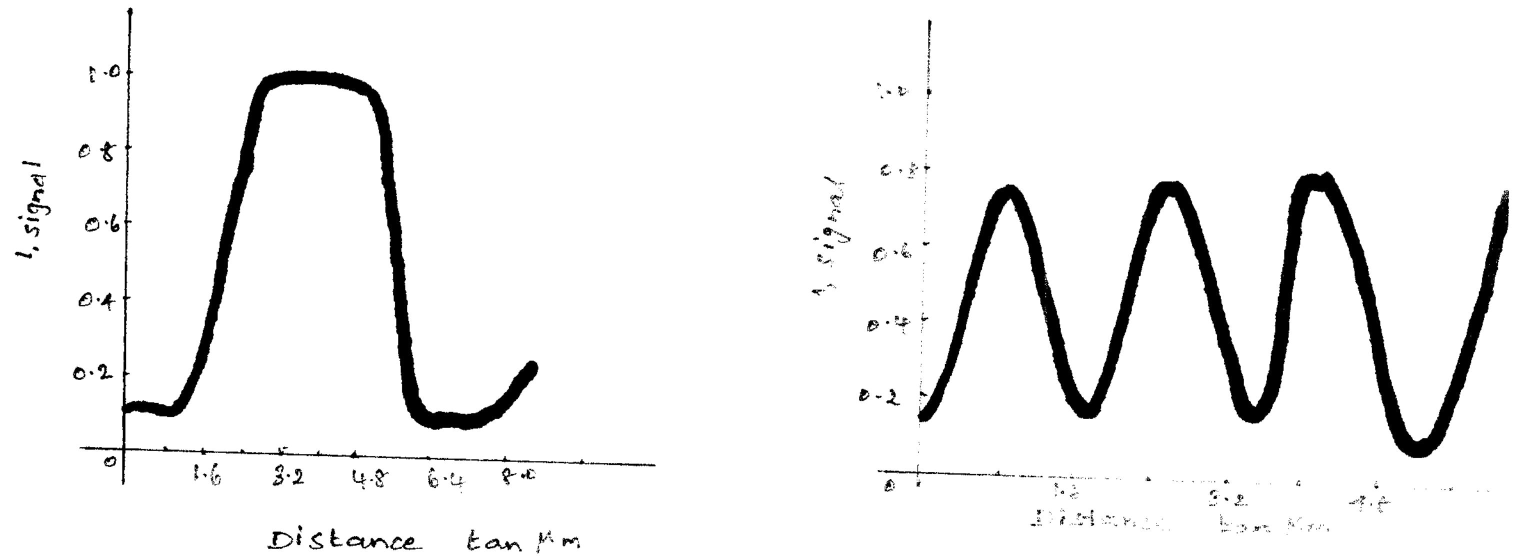
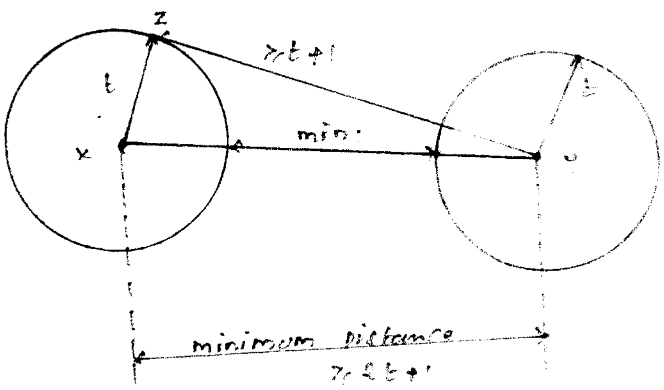


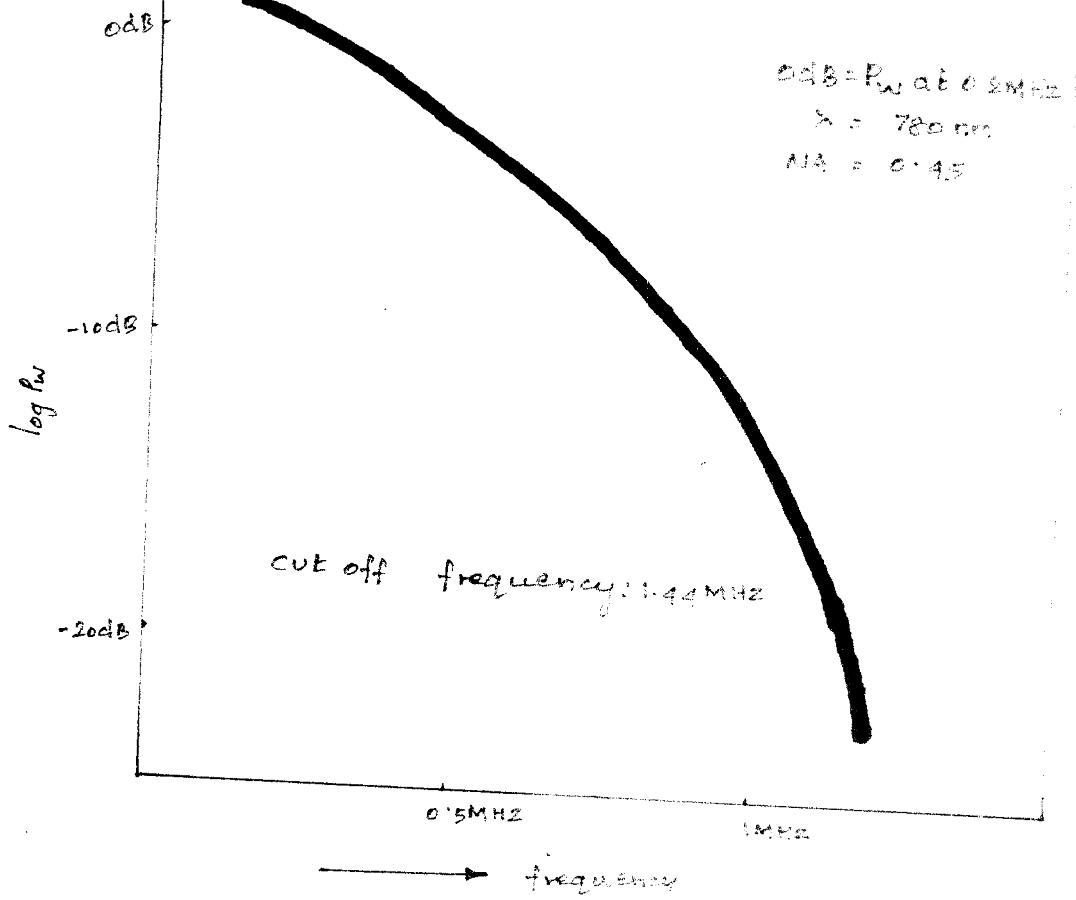
FIG. 9.15 Frame format.



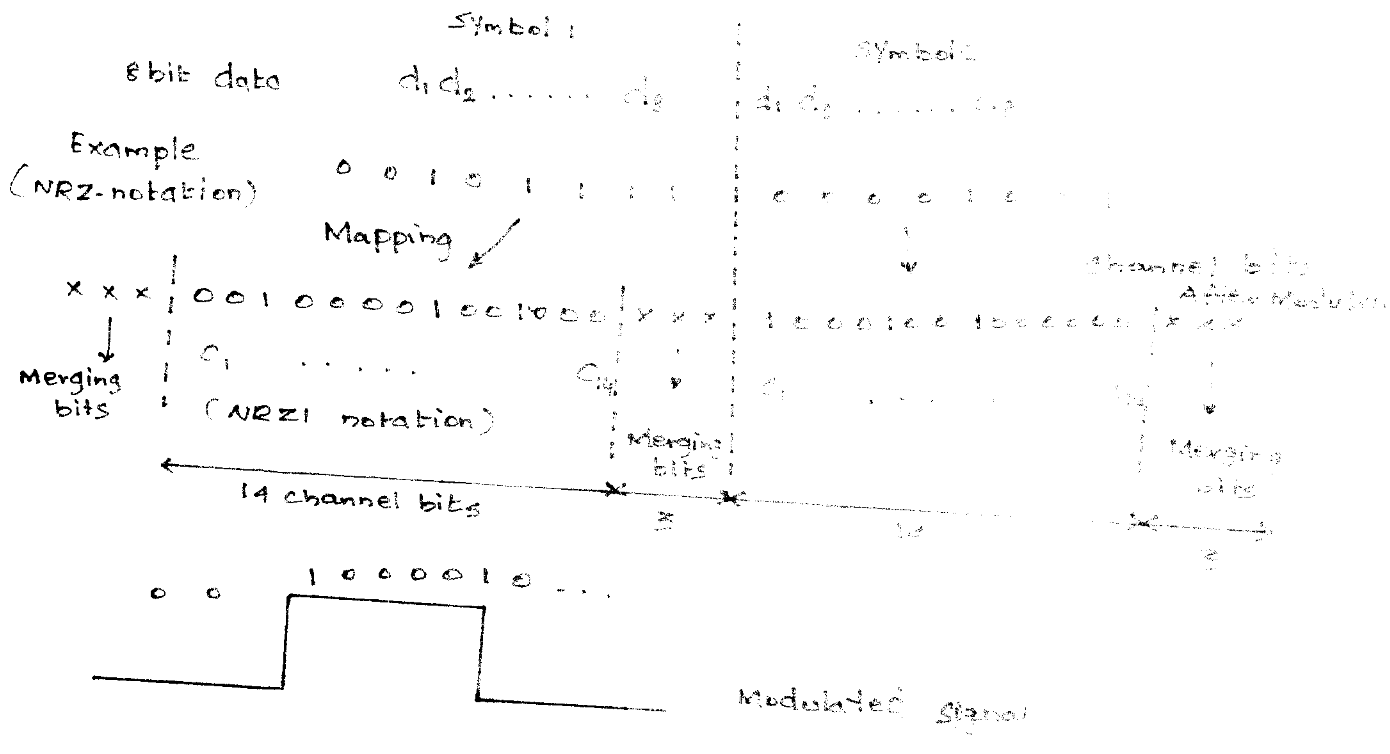
Simulated Waveform of HF signal



Minimum distance for t error correctio

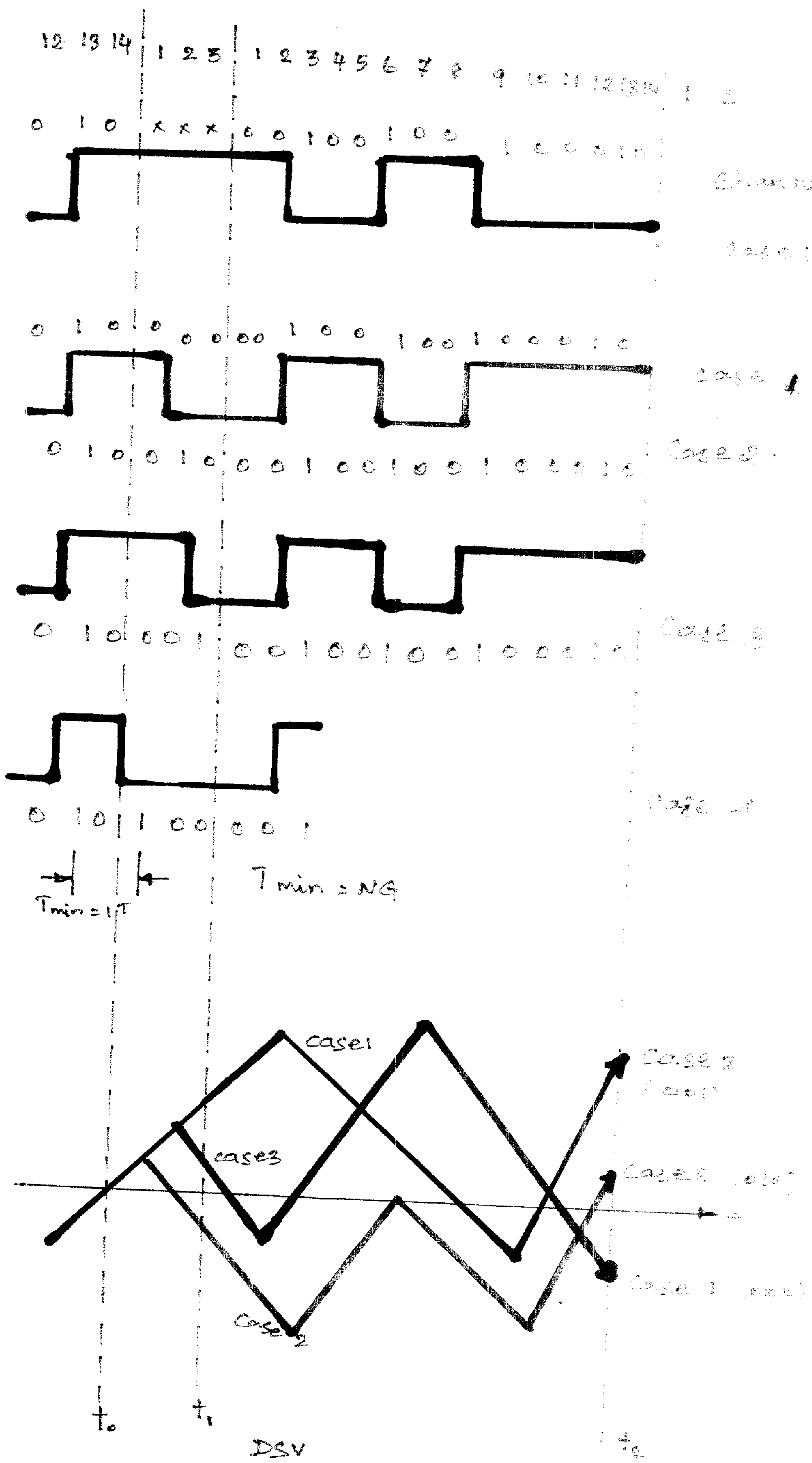


Frequency response of the compact disc system at a scanning velocity of 1.25 m/s 0dB = P_w at 0.2MHz; Wave length = 780 nm; NA = 0.45.



EFM MODULATION.

Example of EFM



100	01100100	01000100100010
101	01100101	00000000100010
102	01100110	01000000100100
103	01100111	00100100100010
104	01101000	01001001000010
105	01101001	10000001000010
106	01101010	10010001000010
107	01101011	10001001000010
108	01101100	01000001000010
109	01101101	00000001000010
110	01101110	00010001000010
111	01101111	00100001000010
112	01110000	10000000100010
113	01110001	10000010000010
114	01110010	10010010000010
115	01110011	00100000100010
116	01110100	01000010000010
117	01110101	00000010000010
118	01110110	[illegible]
119	01110111	00100010000010
120	01111000	01001000000010
121	01111001	00001001001000
122	01111010	10010000000010
123	01111011	10001000000010
124	01111100	01000000000010
125	01111101	00001000000010
126	01111110	00010000000010
127	01111111	00100000000010

FIG. 9.12 Part of the EFM conversion table.

In EFM, 3 channel bits are inserted so as to control run-length and low frequency componentents of the modulaed signal.

An ex. of this control is given in fig.

In this case, possible channel bit - patterns which satisfy the Tmin and Tmax constraint are 000,010 and 001. To control low frequency components, the digital sum variations (DSV) is calculated for these three case, and the best pattern(000) is selected. In such a way, the 8-data bits are converted to a (14+3 = 17) channel bit pattern.

Demodulation in fig is effected by detecting the 14 - channel bit pattern. Thus error propagation is limited to 8 bits, which suits CIRC.

Subcodes:

Many features of the CD player are based on various sub-code recorded on the disk.

Basic Concept :

One of the advantages of digital recording is that the digital format can handle not only the main audio signal but also subcoded nonaudio digital data. In the CD system 3 percent of total recordable data is reserved for sub codes whose main purpose is control and display.

Subcode Format :

Eight bits per frame are used for subcoding these 8 bits are used as eight different subcoding channels (P,Q,R,S,T,U,V,W) this giving each channel a bit rate of 7.35 kbits/s. One subcode block consists of 98 subcode symbols resulting in a block rate of 7.35Khz/98 = 75 Hz. For block synchronization. Two special 14-channel bit synchronization patterns are used thus 96 bits are available for each channel. Channel P is a simple music trade separator flag intended to be used in low - cost search system channel Q is available for more sophisticated controls. The general data format of channel Q [illegible]

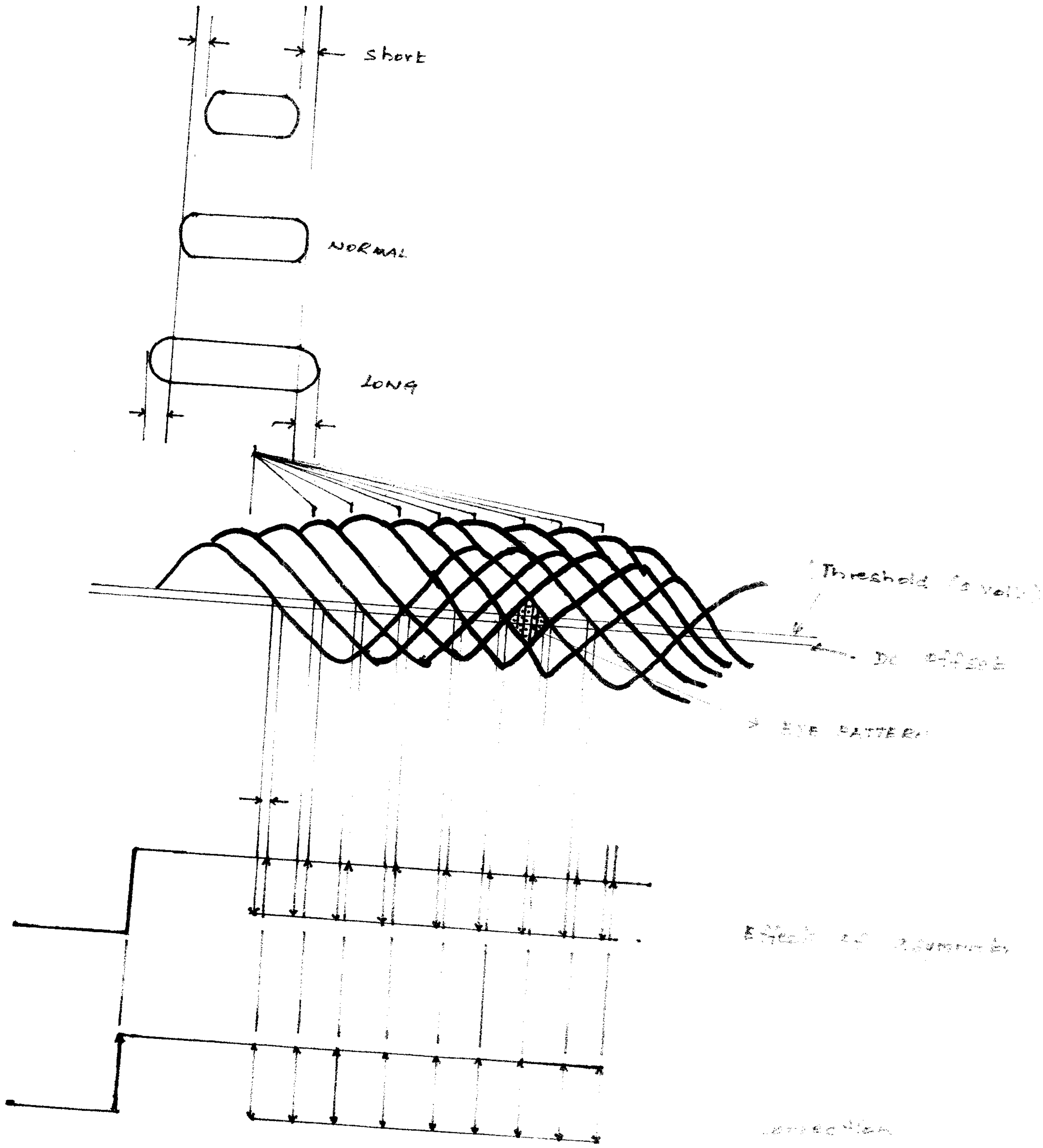
First 4 controls bits indicated essential information such As the number of channels whether or not there is preemphasis and so on. The 4 bits address (ADR) and the succeeding 72-bit DATA-Q section have several data modes. The basic mode indicates the trade number, the elapsed time of correct selection and the total elapsed time. In the lead - in trade, which is located inside the music trades, channel Q indicates the table of contents. Information which is very help for high speed accessing the other mode can be used for such information as the disk manufacturer wishes to make available. The last 16 bits on the cyclic redundancy check (CRC) code for detection.

(Fig) General data format of channel Q channel [illegible] through can be used for display purpose. The maximum available data rate is 6x96x75 = 4.32 Kbit/s.

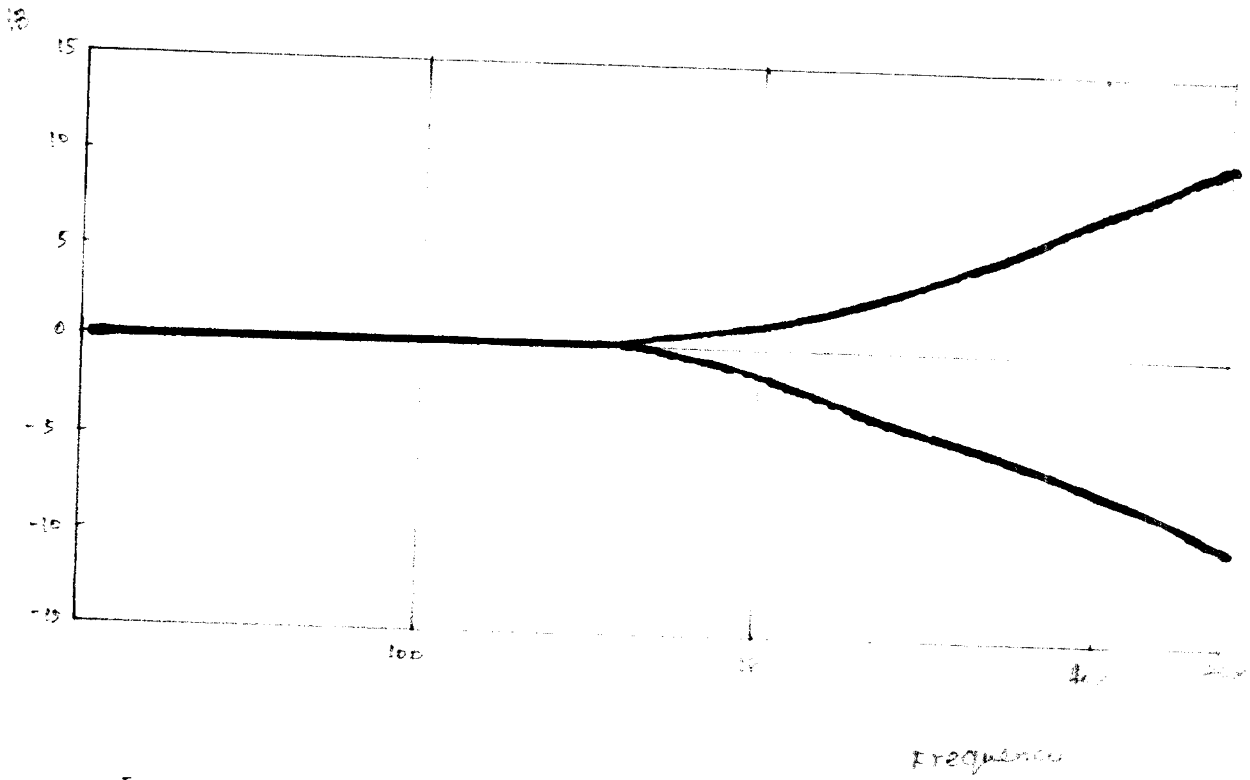
EFM Demodulation

When the laser is reflected at the disc surface during playback, the varying intensity of the returned beam is detected by a photodiode sensor. It is the voltage from the sensor which is ultimately transformed into the analog audio signal output from the player. However, the encoded data from the pickup must first be decoded. The signal encoded on the disc utilized eight - to - fourteen modulation (EFM), which specified that the signal be composed of not less than two 0s or more than ten 0s between logical transitions (pit edges). This results in pit lengths expressed in a variety of combinational patterns from a pit length 3 units long to a pit length 11 units long ; this sets the frequency limits of the EFM signal read from the disc. As we've seen, this range is sometimes referred to as a 3T-11T signal, with T referring to the period of one bit The audio output is ultimately derived from the lengths(read as timing intervals) of the data pits and lands.

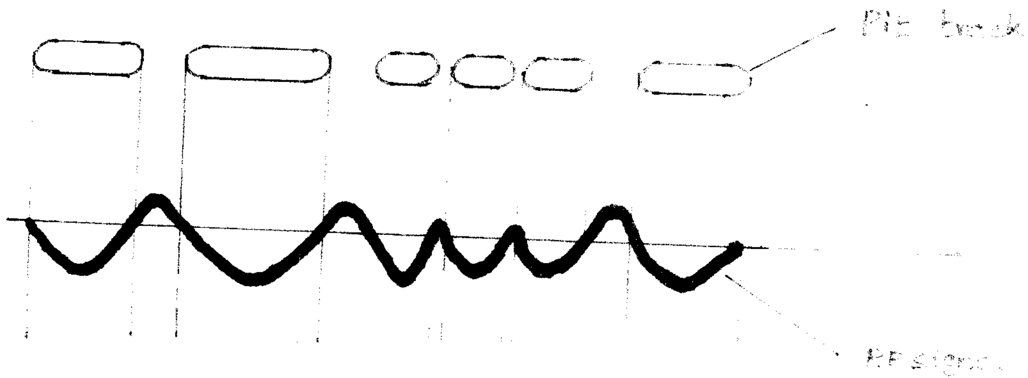
The photodiode and its processing circuits produce a signal resembling a high - frequency sin wave called EFM signal. Because of its high frequency, it is something referred to as an RF (radio frequency) signal. Figure shows the RF signal from a series of pits. A collection of data periods is often called the eye pattern. (The diamond-shaped area in middle of the pattern, along the axis, looks like an eye, if you've got a lively imagination.) An eye pattern is shown in figure



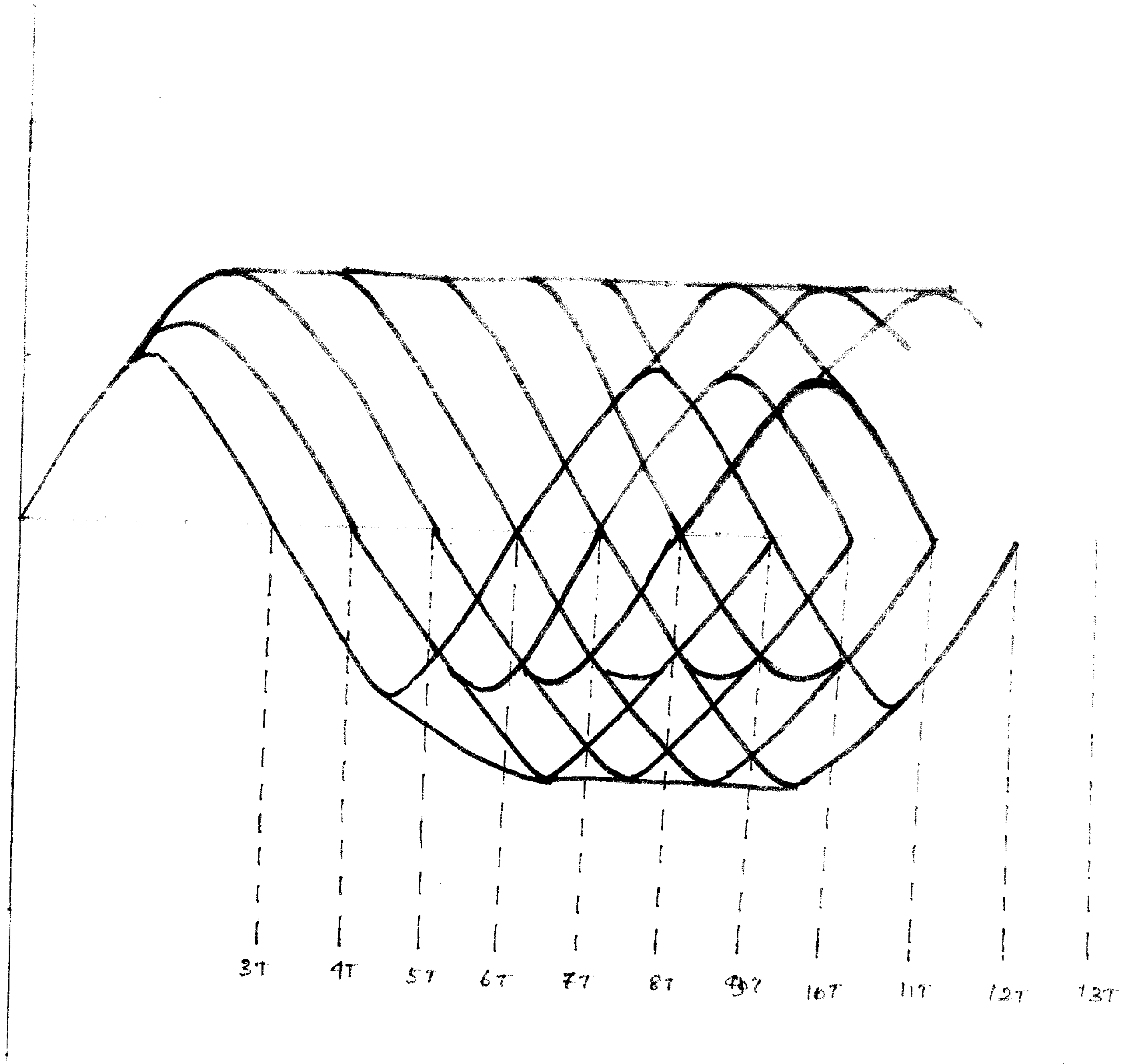
A threshold control circuit eliminates DC offset, due to asymmetry in pit length, in the RF signal



The CD standard pre-emphasis [illegible] curve and [illegible] curve characteristics.



Data pits are used to produce a representative RF signal.



A collection of RF signals forms the eye pattern.

The varying periods of these sine waves (3T,4T, etc. reflect the periods of time required to read the disc pits. the eye pattern is thus an electrical equivalent of the disc's data. The digital data can always be recovered from the eye pattern if it can be determined when the signal crosses the zero axis relative to the 3T - 11T time patterns. Whenever a player is tracking data, the eye pattern is present, and the quality of the signal may be distored eye pattern.

The zero-crossings of the eye pattern contain the sole information content of the EFM signal. Because pit walls are not vertical, the resulting signal is not square. In addition, pits can be longer or shorter than the actual pulse width of the recorded EFM signal, depending on cutting conditions. This causes to be signal to be asymmetrical to the DC level. Figure shows a threshold control circuit which adjusts the threshold level, eliminating DC offset inorder to achieve correct pulse width. The resulting binary waveform now has a more normal digital appearance; the varying width of the waveforms still contain the EFM information.

Because the eye pattern at the photodiodes has a very small amplitude, it must be boosted by a preamplifier. An equalization network or transversal filter must be used to ensure that the smallest signal duration (high frequency) 3T is equal in amplitude with the longest duration (low frequency) 11T. The filter must compensate for the fact the shorter period signal(37,47, etc.) are lower in amplitude than the others. The overall signal is processed through an automatic gain control circuit; no matter what the reflectivity of the CD (within certain tolerances), this circuit optimally adjusts the level of the eye pattern.

Although the EFM signal is composed of varying sinusoidal waveform. It is truly digital; it undergoes processing to convert it into a series of square wave more easily accepted by the digital circuits to follow. This does not effect the encoded data. Since it is the width of the EFM periods which holds the information of interest. Following the conversion of the eye pattern to square waves, the NRZI (non - return to zero inverted) signal is returned to NRZ (non-return to zero), the representation in which a high level is a 1 and a low level is a 0.

The transitions are used to phase - lock a separator clock with period T. This clock identifies the number of T periods between transitions and thus distinguishes between the 17T symbols (each of the thirty - three symbols of a data frame is 17T long). Thanks to our paints in the encoding process. The data of the disc is thus self-clocking. The pulse generator has a slope sensing window of one period (1T) for correctly reading a signal with any jitter or time base error.

The first piece of data to be extracted from the NRZ signal is the synchronization word, three transitions separated by 11T periods. The start of each data frame can thus be identified. Thus synchronization information is used to synchronize the thirty -three symbols of channel information in each frame, and a synchronization pulse is generated to aid in locating the zero crossing of the EFM pattern and to generate a transition at these points to produce a binary signal. The 37 merging bits have served their purpose of preventing EFM code violations and suppressing DC. They are separated from the 177 symbols, leaving 14T symbol.

Because the circuit which controls the disc's precise rotational speed has a limited range. disc speed must be controlled approximately. This is often accomplished by examining the durations of transitions in the EFM signal. The 37-11T range means that transitions cannot occur faster than approximately 0.66 microseconds or slower than 2.44 microsecords. When a CD player first starts to play a disc, or after it jumps to a different track radius (with a different track speed) a correction signal is general to bring the disc rotation to within the capture range of the time base why C1 and C2 parity bytes are inverted in the CIRC encoder; Is are placed in the bit stream when the data in muted with all 0s. Thus minimum and maximum pit length are

recorded during the disc lead-in where the player must quickly regulate disc speed.

The EFM signal now demodulated so that every 14-bit EFM word again becomes 8 bits. Depending on player design, demodulation is effected either by logic circuitry or a lookup table, which is a list stored in memory that use the recorded to refer to the original patterns of 8 bits.

The data stream now contains audio sample, redundancy, and sub-code. The EFM demodulator sends subcode data to the display circuit to convey timing and other information to the player front panel. Other control signals contained in the sub-code are used, among other things, to alert the player's control microcomputer to the exact location of the pickup.

Out Put processing

Following demodulation, interleaved audio and parity data are routed to the CIRC error correction circuit of de-interleaving. error detection, and error correction. Interpolation and muting circuits, previously discussed follow the CIRC decoder. Using error flag from the decoder the player's signal processing circuit determine whether to output data directly, to interpolate it, or to mute the sound, Interpolation and muting performance may also differ from player to player.

Following error correction, the left and right audio channels must be demultiplexed and their respective samples joined together in the same sequence and at the same rate in which they were recorded.

ERROR CORRECTION

Error correction is one of the great opportunities and obligations of digital audio storage media such as the compact disc. It is a great opportunity because it offers something never before possible with analog media the chance to correct mistakes. When you scratch a long playing record, the grooves are irrevocable damaged, along with the information contained in them. Forever after, there will be a click or pop when the damaged part of the groove passes underneath the phonograph needle.

But when you scratch a CD, the nature of the data on the disc and the player's design offer you a second chance. Specifically, the data on the disc has been specially encoded with an error correction code, and your player uses the code to correct for damage data. Thus it delivers the original undamaged variety instead, performing its error correction every time the disc is played.

To illustrate a fundamental aspect of error correction consider the two message below.

Assume the first message represents the information in an LP groove and the second is the data in a CD pit track. Now, place one of your fingers vertically across this page, representating a scratch. You'll observe that part of the first message is irrevocably gone, whereas the second message is intact, because it can be reconstructed. Specifically, we have used redundant data to protect the massage. By comparing the redundant messages, we can overcome the effect of the error. Note that we haven't prevented the error: we have simply insured against its effect. Cleaver, eh?

Of course, even the most clever of ideas has its limitations. As you place more fingers over the page, it becomes more difficult to reconstruct the digital message successfully. Thus the severity of the error plays an important role; our ability to correct errors as limitations. Furthermore if you place your finger horizontally across the page you might completely obscure the entire line, destroying both the message and its redundancy. The nature of the error thus plays a part too. Finally we should note that error correction exacts a cost in this case, the message coded with error correction requires twice the storage space.

On the other hand error correction is an obligation. Digital audio on a compact disc might require storage of fifteen billion bits. With such great data density even the smallest speck of dust would wipe out a considerable number of bits as shown in figure. In reality the type of error, correction utilized would have to be a great deal more sophisticated than a sample redundancy scheme. Even one bad bit could wreak havoc. For example, if the digital word 0000000000000000 (representing silence) was misread as 1000000000000000 (representing a pretty loud level), a click some 90dB above the silence would result.

DETECTION

Before error correction takes place, the errors must be detected. While this might sound obvious, the problem can be difficult. If presented with a data word, could you tell whether or not an error had occurred? For example does 1100101000011110 contain an error? Unless you are psychic. There is no way to tell. The message could be sent twice 1100101000011110 1100101100011110 and close examination would reveal a difference between them. Obviously, both cannot be correct but which one is the right one? The message could be sent three times : 1100101000011110 1100101100011110 1100101000011110, and you might have good suspicion of which one was correct, but would you be sure? What if the message were sent three times, and they were all different?.

While simple repetition is an inefficient way of going about error detection, a more enlightened variation, data redundancy is the essence of error correction. Data redundancy is extra information derived from the original information : hence it is redundant. In general the greater the redundancy the better the error correction. As we have seen redundancy adds to storage overhead and must be optimized in a practical system: this leads to the development of elaborate error correction codes which make very efficient use redundancy by coding the information in certain ways.

CORRECTION

In practice, th redundancy contained in correction codes often takes the form of a parity bit added to every data word. The party bit is chosen so the number of 1s or 0s in the group (data word plus parity bit) is even or odd. For example, a parity bit may be formed with this rule : If the number of 1s in the data word is even (or zero), the parity bit is made 0; if the number of 1s is odd, the parity but is made1. The total number of 1s will always be an even number (or zero); this is called even parity. By the same token, odd parity could be used.

This scheme allows for error detection of received data: Any word with an odd number of 1s must be invalid. Odd parity can detect 1-bit errors in data (actually, it detects all errors involving an odd number of bits). However, if two or any even number of bits are bad, the scheme fails to detect the error and a bad parity bit would cause good data to be flagged as errone-ous. Moreover, the scheme cannot ascertain which bit is bad. Thus it cannot correct errors. Cleary a more sophisticated scheme is needed.

For correction in essence more parity bits are required to solve the problem and the algorithm used to select the parity bits determines the performance of the error correction code. For example, in some codes the data is divided into blocks and parity values are added to each block. Consider the example in figure. In addition to the sixteen data values in figure, nine extra partiy values are created and appended to the original data block. They are placed at the end of each row and column and form the sum of that row and column as shown in fig. Note that a parity value is also included for the parity row and column. If an error occurs in any data (or parity) value, the error can be easily located and the correct value can be easily calculated. Using the other data present.

For example, suppose that the data block in Fig. was received. As we recalculate each parity value at the receiving end and check it against each transmitted parity value. We observe a disagreement. In fact, there is a disagreement in both a row and a column parity value. The intersection of the row and column points to the error. Furthermore, we can now substitute the correct value, derived from the transmitted parity values. The data in the block is thus substantially more reliable. Of course, instead of twelve values, twenty are now required for the error code. In practice a code may take the form shown in figure Several parity bits (a) are generated from combinations of input data bits. Transmitted data is parity_tested, and odd parity indicates an error. Using the results of the parity check, a logic table may be used to determine which bit is incorrect. If data 0011 is received as 0111 (c), parity indicates and error in bit 2.

Numerous error correction codes have been devised. For example, Hamming codes derive multiple parity bits from combinations of the data bits. Simple Hamming codes can be constructed which will detect two errors and correct one error. Similarly, Reed_solomon codes can detect and correct large numbers of.

rors. Using codes such as Reed-Solomono, a digital audio system can detect and supply information that is as food as new. As we will see, the compact disc system uses a Reed-Solomon error correction code as part its defense against errors.

However, as we have seen, the performance of the error correction system depends on the nature of the error. For example, what if a large error obliterated both the data and its parity? There would be nothing left to reconstruct the message. It is thus important to understand te nature of the error and devise protection to fit their nature, Errors can occur in large groups, called burst errors, or isolated instances \- called random errors. The compact disc must guard against both kinds; for example a badly formed pit could cause a random error were as a dust particle could cause a burst error. Clearly, brst error are the most troubling. A good error correction code uses parity in addition to other processing. Such as interleaving.

INTERLEAVING

Interleaving is employed to guard against the very likely occurrence of burst errors. Interleaving might be thought of as shuffling a deck of cards: data symbols are redistributed in the bit stream prior to recording so that consecutive words are never adjacent on the medium. An error occurring in the medium (for example, caused by a dust particle on the disc) might prevent the successful reading of a number of consecutive values. However, upon de-interlaving the shuffled words are placed back in their original and rightful position in the stream, and the errors are scattered in time. Thus isolated, they are much easier to correct. Interleaving is particularly useful for long burst errors; consecutive error are scattered by-interleaving, becoming more like random errors, which are more easily corrected. An interleaving example is shown in figure. Interleaving appears complicated, but it can be accomplished by simply delaying the data words by differing amounts prior to recording. Delaying the data words.again (in a complementary manner) upon playback completes the technique.

Cross interleaving carries the idea one step further. Data is interleaved numerous times, over both short and long time intervals. This provides correctability for larger errors. As we shall see, the CD system uses the Cross Interleave Reed-Solomon code(CIRC) for error protection. It employs parity checking to correct random errors and cross interleaving to permit parity to correct burst errors.

CONCEALMENT

Although correction of massive errors is possible, it would be impractical to implement. In real-life digital audio systems, some errors overwhelm the error correction scheme. These errors are flagged by the correction circuits and passed on to error concealment circuits.

Without concealment, erroneous data that escaped error correction could result in an audible click. Error concealment systems employ interpolation and muting circuits following the CIRC decoder. Although an important part of error protection, they are unrelated to the CIRC. Specifically, they attempt to handle errors too massive for CIRC correction. Using error flags from the CIRC dcoder. The player's concealment circuits determine whether to output the data directly, to interpolate it, or to mute the output.

Interpolation is the technique of using valid data surrounding an error as a basis of forming a replacement for the erroneous data. These values are used to calculate a new value to substitute for the error. Because of the high correlation between music samples, an uncorrected error can be made virtually inaudible by synthesizing new data from surrounding data. Although clicks are avoided, a momentary increase in distortion is produced.

Numerous interpolation schemes are used, with different performance levels. In its simplest form, zero-order interpolation holds the previous value and repeats it to cover the missing or incorrect word. In first-order interpolation, the erroneous word is replaced with a word derived from the mean value of the previous and subsequent words.

In worst case scenarios where the error is so massive that even interpolation would fail. We choose to mute the audio signal. The brief silence is preferable to the burst of digital noise, usually heard as a click. By attenuating the signal before and after the mute, even these catastrophic errors are often made inaudible to most listeners.

However, the digital audio signal cannot be muted by switching the bit stream to zero; this could result in an audible click. Muting methods vary from player to player. For example the signal may be faded down by multiplying the samples by descending coefficients, usually take from a half cycle of a consine waveform. The fade-out must begin prior to the bad data; this is accommodated by feeding the signal through a delay before the muting circuit. The mute signal thus arrives before the bad data. Following the bad data, a fade-in is similarly accomplished. Smooth mutes are suprisingly inaudible.

CROSS INTERLEAVE REED-SOLOMON CODE (CIRC)

As we have seen, error correction is essential to the success of digital audio. Without it, any digital recording on tape or disc would sound like a badly scratched LP at best or, at worst, be simply unplayable. In fact, without error correction, digital audio would be an impossibility.

The raw error rate from a CD is around 10 to 1 or about one error for every 0.1 to 1 million bits. This is impressive storage capability, but considering that a disc will output over 4 million bits per second, the need for error correction is obvious. With error correction, perhaps 200 error per second will be completely corrected to achieve such results, the CD employs the cornerstones of error correction; interleaving to distribute error and parity to correct them. The particular algorithm used for error correction in the compact disc system is, as mentioned earlier, the Cross Interleave Reed-Solomon Code (CIRC). The CIRC circuit uses two correction codes for additional correcting capability and three interleaving stages to encode data before it is placed on a disc. Similarly, CIRC performs error correction while decoding the data during blayback.

REED-SOLOMON CODES

The Reed-Solomon code used in CIRC us a highly efficient error correcting code. It is particularly well suited for the compact disc system because its decoding requirements are relatively simple. To illustrate the operation of Reed-Solomon code, suppose that A,B,C and D are four data words, and P and Q are two party words. The Symbol denotes modulo 2 addition (EX OR) and the alphas are weighting factors. The data words are placed into two simultaneous equations:

A B C D P Q = 0

a a a a a aP aQ = 0

Solving for P and Q, we have

$P = \alpha A \oplus \alpha^2 B \oplus \alpha^{?} C \oplus$ [illegible]

$Q = \alpha^3 A \oplus \alpha^6 B \oplus \alpha^{?} C \oplus \alpha D$

Recorded data

Information words

A	001	
B	101	
C	011	
D	100	

Parity words

P = [illegible]

Q = [illegible]

Table. SAMPLE VALUES AND CALCULATED PARITY WORDS

DECODING EQUATIONS

$S_1 = A \oplus$ [illegible]

$S_2 = \alpha^6 A' \oplus \alpha^5 B' \oplus \alpha^4 C' \oplus \alpha^3 D' \oplus \alpha^2 P' \oplus \alpha Q'$

$= \alpha^6 E_A \oplus \alpha^5 E_B \oplus \alpha^4 E_C \oplus \alpha^3 E_D \oplus \alpha^2 E_P \oplus \alpha E_Q$

POLYNOMIALS

$P = \alpha^6 A \oplus \alpha B \oplus \alpha^2 C \oplus \alpha^5 D \oplus \alpha^3 E$

$Q = \alpha^2 A \oplus \alpha^3 B \oplus \alpha^6 C \oplus \alpha^4 D \oplus \alpha E$

SYNDROMES

$S_1 = A' \oplus B' \oplus$ [illegible]

$S_2 = \alpha^7 A' \oplus \alpha^6 B' \oplus \alpha^5 C' \oplus \alpha^4 D' \oplus \alpha^3 E' \oplus$ [illegible]

Where we assume that

a = 010

a = 100

a = 011

a = 110

a = 111

a = 101

1 = a = 001

0 = 000

Table shows same values for A,B,C,D,P. and Q. To detect errors in the received data, two syndromes, or error patterns, are calculated by these decoding equations :

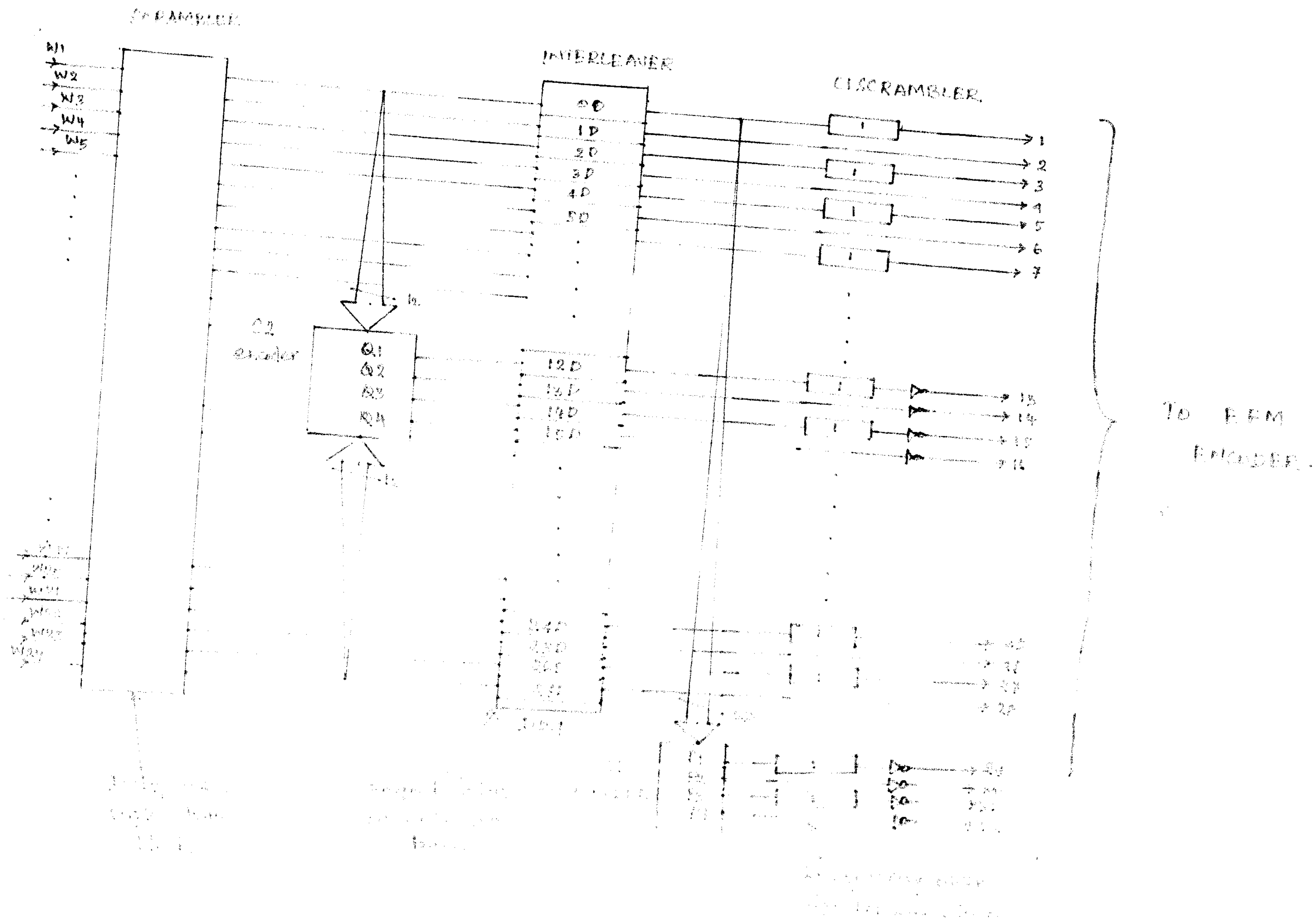
Where A' through Q' are the reproduced words, E through E are the error of each word, and the alphas (along with 1 and 0) are the weighting constants. If there is no error then S = S = 0. If the word A is erroneous, S1 = E and S = S. in other words, an error result in non zero sundromes, and the value of the erroneous words can be determined by the difference of the weighting between S and S . In parctice, the polynomials used in the CIRC correction code are :

and the resulting syndromes are :

As noted previously, CIRC employs cross interleaving, the separation of two error correction codes by an interleaving stage. Thus one code can check the accuracy of the other. Another important aspect of cross interleaving is that error correction is enhanced, but the amount of redundancy is not increased. An example of cross interleaving and the way it facilitates parity checking is shown in figure. S represents data values (called symbols), P and Q are parity values, and C1 and C2 are correction decoders capable of correcting one and two symbols respectively. If S13 is erroneous, C1Q16 corrects it. If S13 and S14 are erroneous, C1Q16 and C1Q20 correct them. If S13 and S16 are erroneous, C2P12 and C2P16 y correct them. If S13, S14, S16 and S17 are erroneous, C2P12 and C2P16 correct them by using C1's error detection pointer. If there are more errors in the P12 and P16 string, correction becomes impossible. CIRC uses cross interleaving similarly; however, the C1 and C2 decoders are more powerful.

CIRC ENCODING

The complete CIRC encoding scheme is shown in figure. With this encoding algorithm, bits from the audio signal are cross-interleaved, and two encoding stages generate parity symbols, or data values. Error correction encoding being with the first stage of interleaving, which is designed to assist interpolation. Twenty-four 8-bit symbols are applied to the CIRC encoder. A delay of two symbols is played between even an odd samples: That



CIRC encoding

is, even samples are delayed by two blocks. Interpolation can be accomplished in the case where two uncorrectable blocks occur Next, the symbol;s are scrambled so as to separate even and odd numbered data words. This process facilitates concealment.

The C2 encoder accepts the 24-byte parallel word and produces 4 bytes of Q parity. Q parity is designed to correct one erroneous symbol or up to four erasures in one word. An erasure is a word that has been erased by the decoders because detection has ascertained its value is unreliable. The parity symbols are placed in the center of the block to increase the odd/even distance. This enhances interpolation in the case of burst errors.

Cross interleaving follows the C2 encoder. The 28 bytes are delayed by differing periods, which are integer multiples of 4 blocks. As a result of this convolutional interleave, each C2 words is stored in 28 different blocks, distributed over 109 blocks. In mathematical terms, we have corded a data array in two direction .

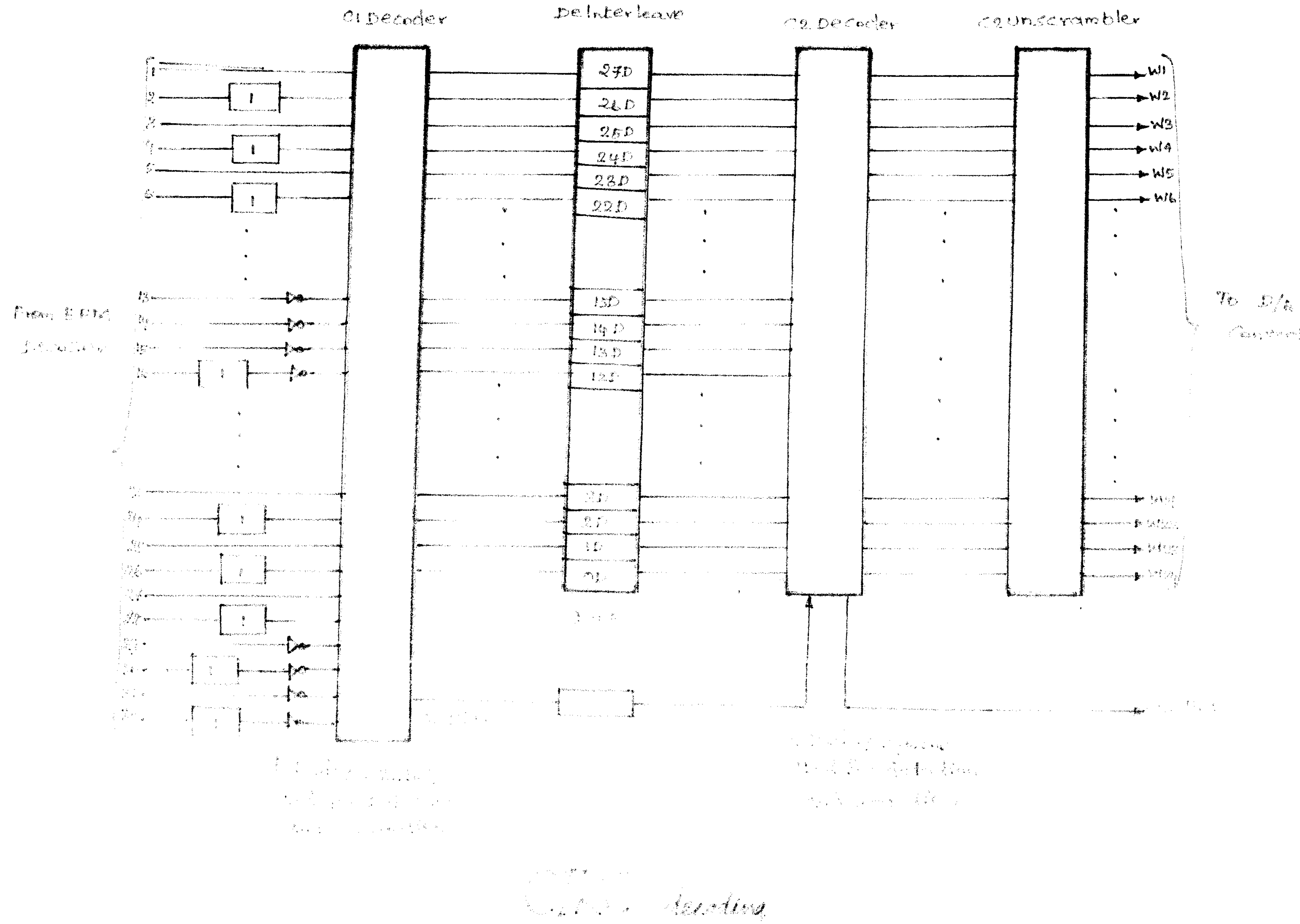
The C1 encoder accepts a 28-byte word (from 28 different C2 words) and produces 4 more bytes of P parity. The C1 encoder is used to corect single symbol errors and to detect and flag double ad triple errors for Q correction.

A last interleave stage introduces a fixed delay of one symbol to alternate symbols. This odd/even delay spreads the output words over two data blocks. This prevents random errors from disrupting more than one symbol in one word, even if two adjacent symbols in one block are erroneous. Finally, the P and Q parity symbols are inverted to provide nonzero P and Q symbols with zero data. This assists data readout during areas with muted audio with muted audio program. Thirty-two 8-bit symbols leave the CIRC encoder.

CIRC DECODING

Upon playback, following demodulation, data is sent to a CIRC decoder for de-interleaving, error detection, and correction. The CIRC decoding process reverses many of the processing steps accomplished during encoding; the complete CIRC decoding process is shown in figure. It utilizes parity from two Reed-Solomon decoders and de-interleaving. The first decoder, C1, is designed to correct random errors and to detect burst errors. it puts a flag on all burst errors to alert the second decoder, C2. Given this prior knowledge, and help from de-interleaving, C2 is able to correct burst errors as well as random errors that C1 was unable to correct.

During reproduction, the CIRC decoder accepts one frame of thirty two 8-bit symbols: twenty-four are audio symbols and eight are parity symbols. Odd numbered symbols are delayed, and parity symbol are inverted. The delay lines have a delay equal; to the duration of one symbol. Thus information of even-numbered symbols of a frame is de-cross interleaved with that of the odd-cross-interleaved with that of the odd-numbered symbols on the next frame. This de-interleaving places the even-and odd-



C1 Decoder
De Interleave
C2 Decoder
C2 Unscrambler
27D
26D
25D
24D
23D
22D
W1
W2
W3
W4
W5
W6

numbered audio symbols back into their original order by rearranging their ordr as read from the disc. Any sequence of errors on the disc is thus distributed among valid data

In the C1 decoder, error are detected and corrected by the four P parity symbols; short duration random errors are corrected and longer burst errors are passed along. Specifically the C1 decoder can correct a symbol error in every word of thirty-two symbols. If there is more than one erroneous symbol, then all twenty-eight data symbols are marked with an erasure flag and passed on. Valid symbols those adhering to the encoding rules of C1, are passed along unprocessed.

The delays between the decoders are of unequal length and longer than those at the input to the C1 decoder: this convolutional interleaving enables the C2 decoder to correct longer burst errors. Because the word arriving at C2 contains symbols from C1 decoded at different times, symbols marked with an erasure flag are now distributed among valid symbols. This helps the C2 decoder correct burst errors. Symbols without a flag are assumed error free and passed though unprocessed. In the C2 decoder errors are corrected by the four Q parity symbols, if flagged (in some implementations. only two symbols can be corrected). Burst errors and errors which might have occured in the encoded process itself (rather than in the medium are corrected. In addition, C2 can correct symbols miscorrected by C1 decoding. When C2 cannot accomplish correction because more than four symbols are flagged, twenty-four data symbols are flagged as uncorrected and passed on for interpolation. Final descrambling and delay is performed to assist interpolation.

The use of two correction decoders and cross interleaving helps tackle a particularly difficult error scenario. Innterleaving distributed burst errors perhaps caused by surface contamination, over different words for easier correction. Nevertheless, correction is difficult when a burst error coincides with a random error introduced perhaps by a manufacturing defect. In the CD standars, random errors are defined to be single symbol errors. or no more tha 17T(time periods) in length. Any longer errors are burst errors. EFM coding guarantees that a random error will never corrupt more than two symbols, and the even/odd interleave grantees that a two-symbol random error will always appear as a single error in two different C1 words after de-interleaving. Random errors are thus always correctable, and the C2 decoder retains its brust error correction capability.

CIRC PERFORMANCE

It should be noted that CD players are not created equal in terms of error correction, Any CD players error correction ability is limited to the success of the strategy chosen to decode the CIRC code on the disc. For example, for more efficient correction, decoding could be repeated alternatively, in the form C1,C2, C1,C2.

A measure of the overall correction performance is the number of interpolated sampled values for a given bit error rate (BER) in a given time. The lower the interpolation rate for a given BER value is, better the system's random error correction quality wil be. Althernative, BLER (block error rate) measures blocks containing error. An assessment of system performance must also account for uncorrected errors that pass through the error correction circuit. These errors may result in an audible click on the output.

In theory, CIRC detects and completely corrects burst errors of up to 4000 bad bits, or a period of 1.9 milliseconds or a physical defect 2.47 millimeters long, Interpolation can conceal error of upto 13,700 bad bits, corresponding to a physical defect of 8.5 millimeter. Looked at in an other way, the raw error rate before correction) from a compact disc may range from 10 to 10 , or one error in every 10 to 10 bits. After CIRC in theory the error rate is between 10 and 10 , or about one uncorrected error in every 10 or 100 billion bits.

However, the quality of the error correcting performance varies from player to player. Depending on the CIRC decoding strategy used, the theoretical probabilty of an error occurring can vary from quite often to almost never. Three different CIRC decoding strategies are shown in figure, and their random error correction performance is shown in figure. Although the simplest strategy, number1, is satisfactory for most applications strategy 3 is clearly must better. In the example, other strategies were developed by interchanging the C1 and C2 decoders of strategies 2 and 3. Strategy (2,3) uses the C1 decoder from stately 2 and the C2 decoder from strategy 3. This illustrates the effect of different error conditions on the success of the correction strategy. For example, strategy 3 is superior to strategy 2 primarily because of its ability bits are translated into blocks of 14 bits, known as channel bits, using a ROM dictionary which assigns an arbitrary and unambiguous word of 14 bits to each 8-bit word; a part of the lookup table is shown in table.

In EFM, each 1-bit words is translated to a 1--bit word selected its specific bit pattern.

By choosing the right 14-bit words, the bit patterns with a low number (and known rate) of transitions between 0 and 1, greater data density can be achieved. It would be inefficient to store the 8-bit symbols directly on the disc; the larger number of 0/1 transition would demand many pits. The 8 data bits required 2 and 256 different code patterns. However, the 14-bit channel word can offer 16.384 combinations. In addition, 8-bitsymbols have many similar patterns for example if one were wrong, 0000101 would be confused 0001011. With 14-bit words, more unique patterns can be selected. EFM thus provides a kind error correction.

To achieve pits of controlled length, only those combinations are selected in which more than two but less than ten 0s appear continuously. Only 267 combinations satisfy these these criteria. With these patterns, the minimum inversion width of the signal is limited by the recording density, and the maximum inversion width is limited by the clock bit extraction. The conversion is thus a compromise between several conditions. Because only 256 patterns are needed, 11 of the 257 pattern are discarded (two of them used for subcode sunchronization words).

3. FRAME FORMAT

Because the system must be self clocking a synchronization is necessary. Therefore the data stream is split up into frames.

Each frame contains:

-a synchronization pattern of 24 bits.

-24 data symbols of 8 bits each

-8 error correction parity symbols of 8 bits each.

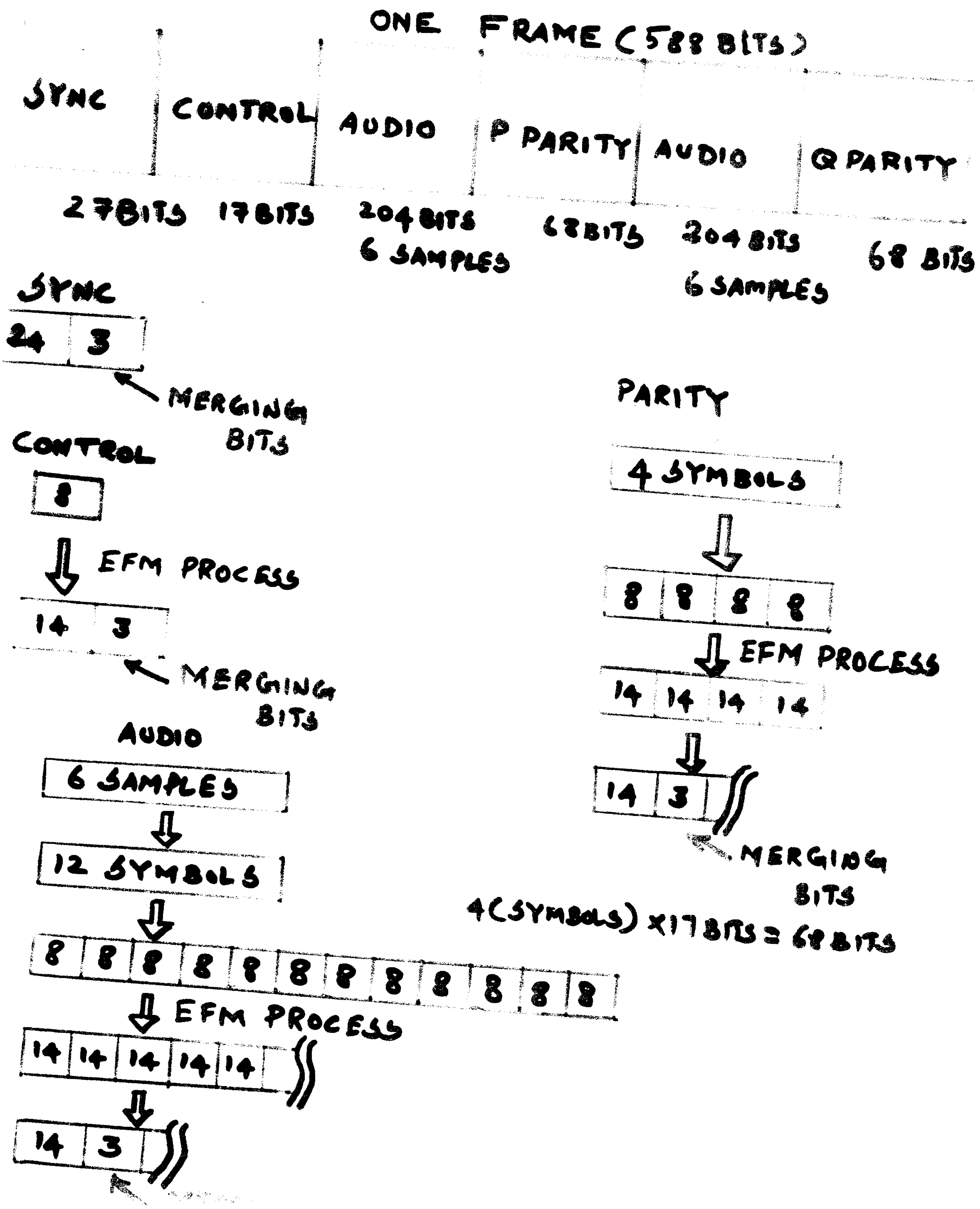
-a control & display symbol of 8 bits.

The 16-bit data words are each split up into two 8-bit symbols which are fed into the modulation circuit.

After modulation each symbol is converted into 14 channel bits. The synchronization pattern consists of 24 channel bits plus 3 bits for merging and low frequency suppression.

The total number of channel bits per frame is :

-sync pattern	:24	channel bits
-control & display	: 1 x 14	channel bits
-data	: 24 x 14	channel bits
-error correction	: 8 x 14	channel bits
-merging and LF suppression	: 34 x 3	channel bits
Total	588	channel bits


ONE FRAME (588 BITS)
SYNC
CONTROL
AUDIO
P PARITY
AUDIO
Q PARITY
27 BITS
17 BITS
204 BITS
6 SAMPLES
68 BITS
204 BITS
6 SAMPLES
68 BITS
SYNC
24
3
MERGING BITS
CONTROL
8
EFM PROCESS
14
3
MERGING BITS
AUDIO
6 SAMPLES
12 SYMBOLS
8 8 8 8 8 8 8 8 8 8 8 8
EFM PROCESS
14 14 14 14 14
14 3
PARITY
4 SYMBOLS
8 8 8 8
EFM PROCESS
14 14 14 14
14 3
MERGING BITS
4 (SYMBOLS) x 17 BITS = 68 BITS

'CD' Player Design

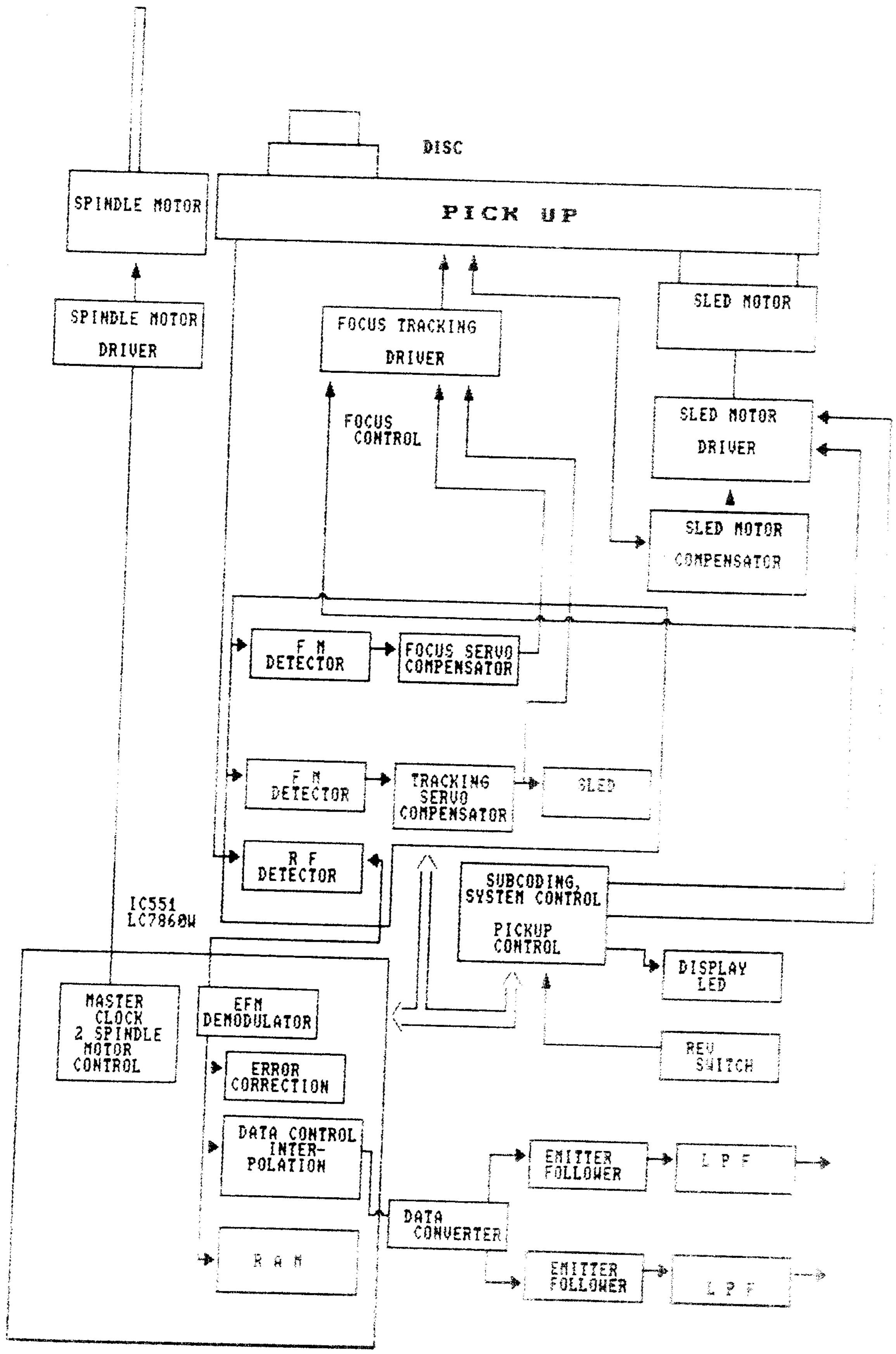
CD PLAYER DESIGN

INTRODUCTION :

A Compact Disk Player is perhaps the most sophisticated and electronically complex piece of home entertainment equipment to ever reach the consumer.This unit provides the design aspects of CD Player.

THE BLOCK DIAGRAM FUNCTIONAL COMPONENTS :

A Block Diagram of CD Player is shown in the figure.The concentrated spot on to the information layer reads the signal,which has been recorded on the disc is in Digitally Encoded Form.



'CD' BLOCK DIAGRAM.

PLAYER OVER VIEW :

The CD Player contains two primary systems.

1) The audio data processing system.

2) Servo and control system.

The data path directs the signal from the pickup (carrying information derived from the modulated light received from the disk) through a series of processing circuits,ultimately yielding a stereo analog signal.The data path of a contemporary CD Player consists of elements such as

SIGNAL PROCESSING PART :-

1) Analog signal processor (ASP)

2) DIgital signal processor (DSP)

POWER DRIVER :-

* Tracking and focus servo driver.

CONTROLLER :-

* Micro Processor (µP)

DATA CONVERSION PART :-

* Digital to analog convertor (D/A)

The servo Control and display system must direct the mechanical operation of the player including.

* Spindle Drive
* Auto Tracking
* Auto Focus

and the user interface to the player's controls and displays.

The CD Player employs a read out system using a semiconductor laser.A non contact means is required to read data,maintain the laser beam's focus and track pit spiral.The result is a highly sophisticated pick up,utilizing complex optical devices within servo loops.The servo use electrical signals from the pick up to control motors to mechanically adjust the pick up's position horizontally and vertically relative to the disk surface.

A Spindle motor is used to rotate the disk with constant linear velocity.Thus the disk must vary its speed depending on where the pick up is located underneath the surface.In another , servo loop information from the data itself is used to determine the correct rotating speed and maintain the proper data output rate.

The micro processor monitors user controls and their interface to the player's circuitry.Various modes of player operation are controlled by the s/w.Sub code data plays an important role in directing the pick up to the proper disk location.For example,a time code is used to locate the start of any track.

Once the data is recovered from the disk,the player must decode the audio information to reconstruct the audio signal.The Eight To Fourteen Modulated (E F M) Data is demodulated ,errors are detected and corrected by decoding the error correction algorithm,and the audibility of gross errors is minimized through interpolation and muting.In Many players ,this corrected data is output through a digital serial transmission interface before also being processed into an analog wave form.

Digital Filters are used to over sample the signal so that image components may be easily removed with gentle analog output filters.The digital data must be converted to a stereo analog signal ,this requires one or two D/A convertors and analog low pass filters.

A final output circuit is nothing more than an interface to outside world.It ensures that the player's line level output is appropriate to drive external amplifiers with a minimum amount of analog distortion.

NOTE :-

The electrical design of CD Players has under gone three evolutionary steps since the format's introduction.

1) FIRST GENERATION PLAYER.

2) SECOND GENERATION PLAYER.

3) THIRD GENERATION PLAYER.

FIRST GENERATION PLAYER

First Generation player uses multi-bit D/A convertors followed by analog brick wall filters to perform conversion and filtering of the signal.

SECOND GENERATION PLAYER.

This uses multi-bit D/A convertors and digital over sampling filter which is placed prior to the convertors.

THIRD GENERATION PLAYER

Our Project is based on this latest type.Here we have used low bit D/A convertor and digital over sampling filter - were possible by the advent of powerful DSP technology.

PICK UP DESIGN :-

The CD is certainly one of the most advanced storage media available.One reason is its optical pick up.A disk may contain 3 billion bits precisely arranged on a spiral track.

The optical pick up must focus on,track and read the data spiral.The entire lens assembly,a combination of the laser source and the reader,must be small enough to glide laterally beneath the disk,moving in response to tracking information and user random access programming.

The pick up must maintain focussing and tracking even under adverse playing conditions such as the playing of a dirty disc or impact and vibration.

OPTICAL REQUIREMENTS :-

The principle and some optical requirements for the readout of a digitally encoded signal from a disk by means of scanning spot are described.

BASIC OPTICS FOR READING :-

The basic optics for reading are shown in figure.This lens to concentrate a spot on to the information layer of a disc,a beam splitter and a pin diode as a photo detector which converts to electrical current.

The optical principle of non-contact read out is based on diffraction theory.Though this phenomenon by means of a narrow slot is well known,an analogous situation that occur if a light beam impinges on a reflective signal surface with pit like depressions.In the case of a flat surface (Between Pits).nearly all the light is reflected ,where as if a pit is present,the major part of the light is scattered and substantially less light is detected by the photo detector.

OPTICAL PICK UP :-

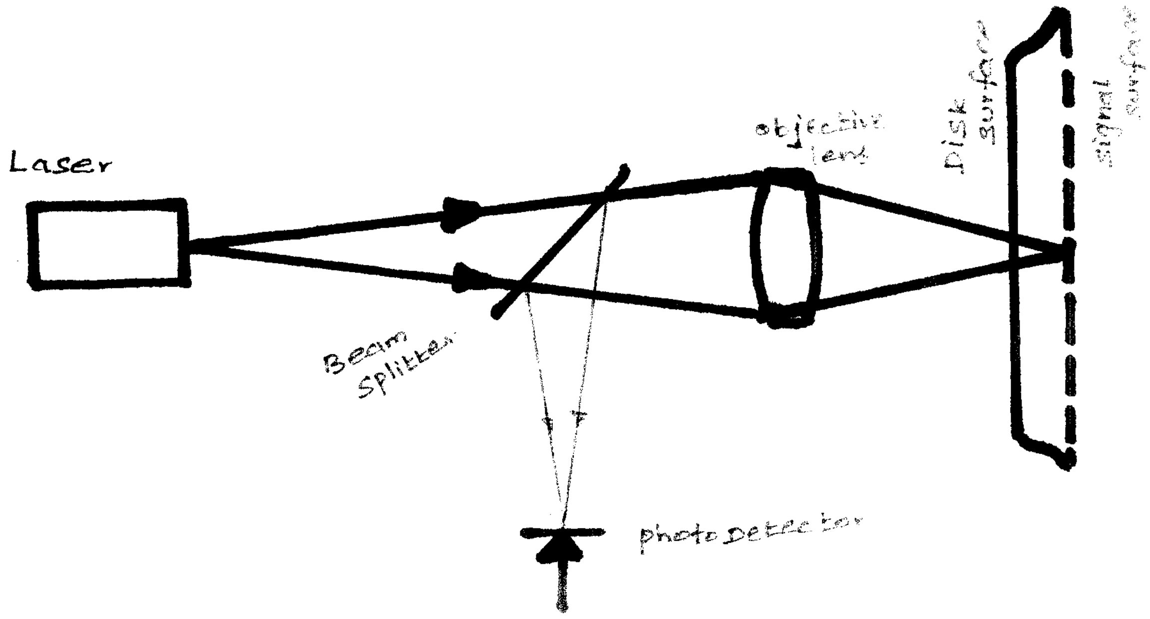
In the CD system the optical pick up detects the optical signal so it can be called the heart of the CD system.

The very important requirements for the pick up as

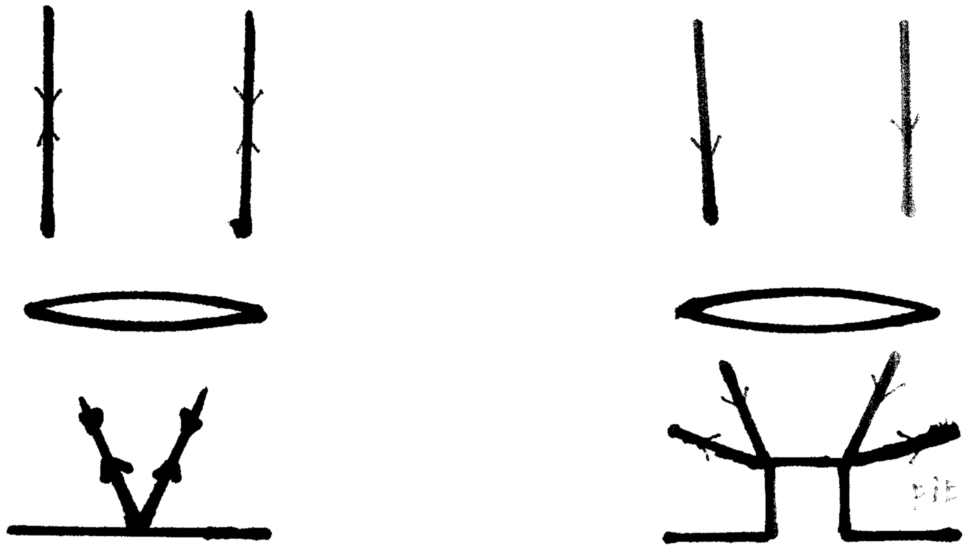
1) It Must Be small

2) It must be able to correctly read the signal under any usage conditions.

3) It Must be inexpensive i.e It must be easy to produce.



Basic optics for reading



scattering of light by phase object (pit)

Principle of noncontact reading

LASER DIODE :- The light source used in the CD system is a semi conductor Aluminum Galium Arsenide (Al Ga As) Laser Diode,it must satisfy the following conditions.

1) It must be small enough to be built into optical pick up.

2)It uses coherent light in order to focus on an exceedingly small spot.

3)Enough light intensity for read out must be provided.

Our Al Ga As semiconductor laser diode satisfy the above requirements.The specifications of such a laser diode are the following

Wave Length = 0.78 to 0.83 um

Light Power = about 3 mw

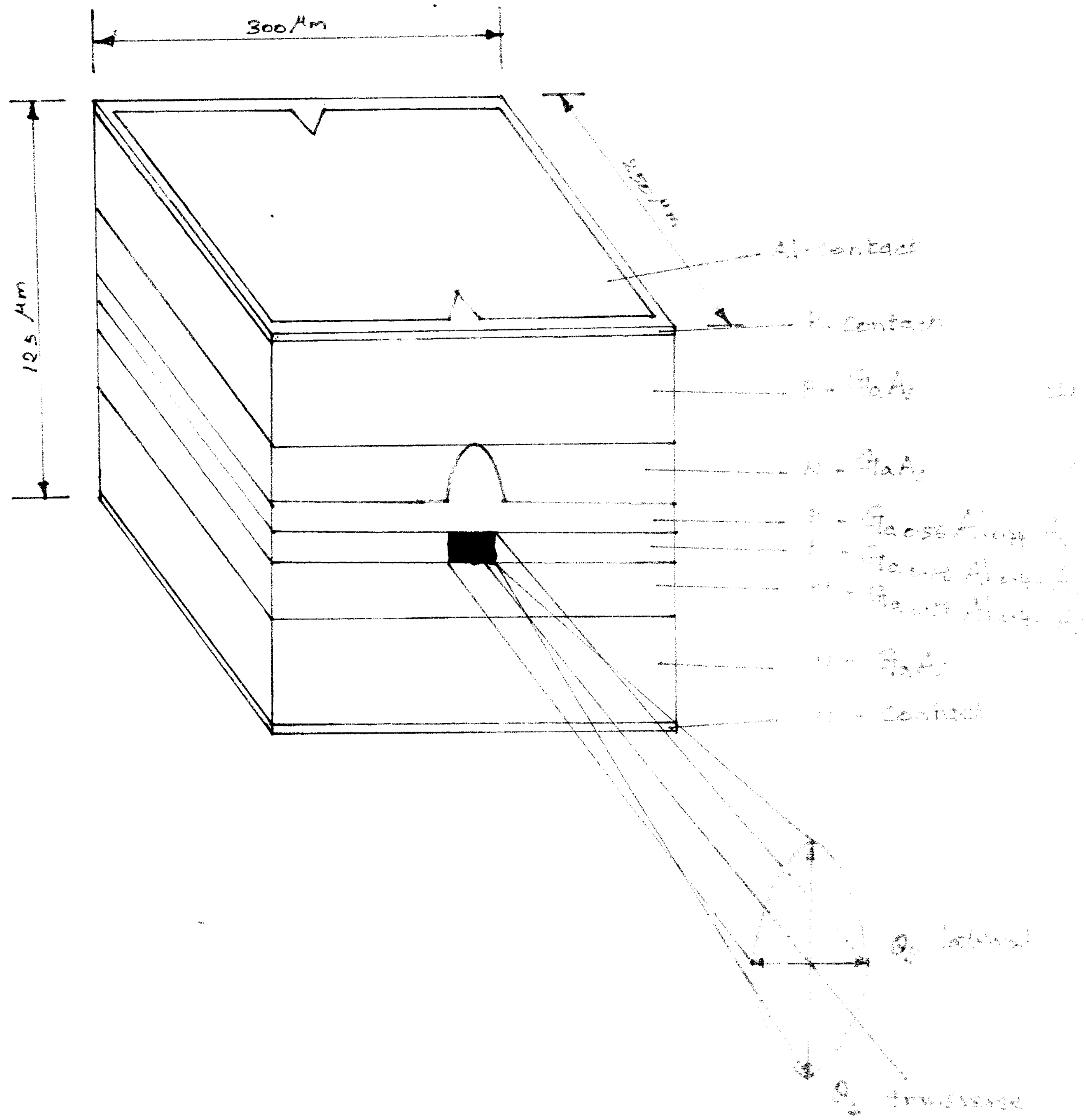
Lateral Mode = fundamental

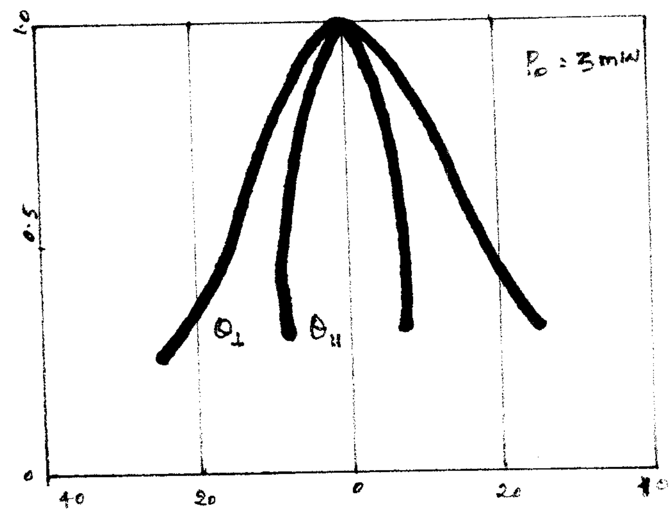
Transverse Mode = fundamental

Longitudinal Mode = multiple

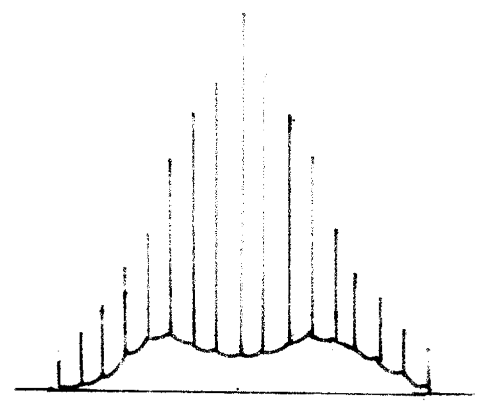
When the light from the laser diode is returned from the reflective surface of the disc, it has an effect on the light generating characteristics of the laser diode and generates large optical noise fluctuations. Thus a multiple longitudinal mode is necessary to prevent the phenomenon. A typical structure and optical and electrical characteristics are shown in fig.

Structure of the laser diode.



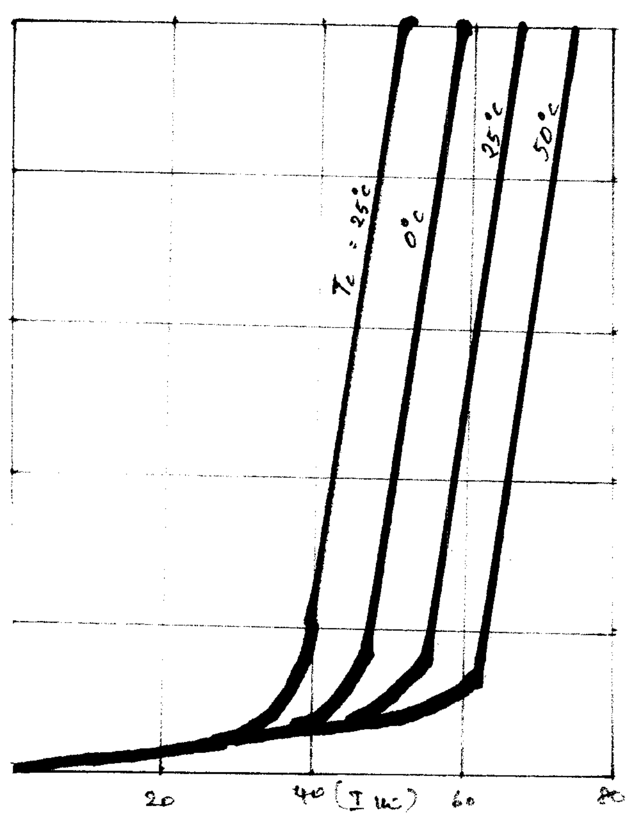


Degrees (a)



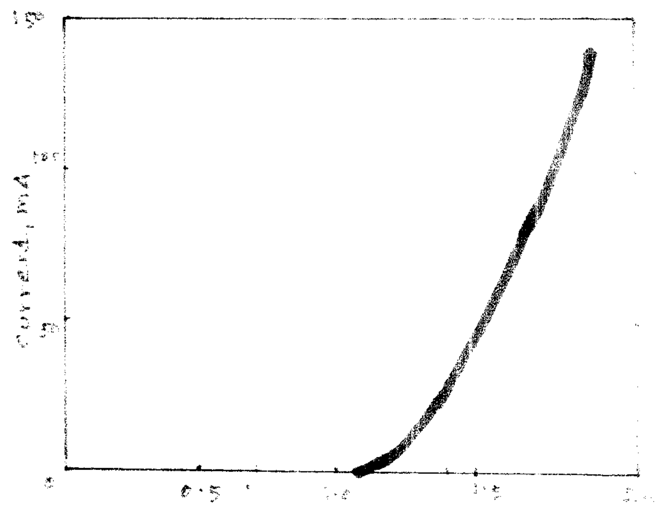
Wave length, 780 nm

(b)



Current, I_F, mA

(c)



Voltage, V

(d)

specification of laser diode (a) far-field pattern.
(b) Longitudional multimode spectrum. (c) L-I characteristics.
(d) V-I characteristics.

Advantages of Al Ga As laser diode :-

1. Power handled is high
2. Spectral width 0.2 nm
3. Coupling is good. (coherent)
4. The signal loss is less
5. It operates on both logitudinal mode , lateral mode and transverse mode

Disadvantages

1. Threshold current is not constant with temperature
2. Short working hours.
3. Complex.

LENS ASSEMBLY:-

Originally a combination lens(2 groups, 3 pieces) like that was used in the CD system. This was very expensive. The lens requirement can be best described by means of Numerical Aperture (NA).

Owing to diffraction at the lens aperture, the light beam has a limited value. It is well known that when a beam with uniform distribution of flux is incident to a lens, the beam projects a pattern known as Array disk. The diameter of the first ring, in which about 84% of the energy is concentrated, is given roughly by,

1.22 * Wave length/ Numerical aperture

If the strength is defined as 1 / e (e is the base of natural log), then the effective beam diameter is given by,

0.82 * Wave length / Numerical aperture

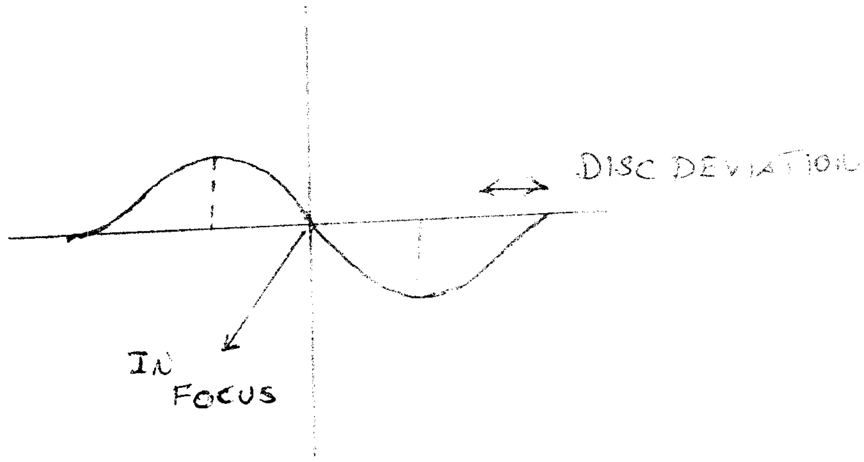
From these equations, it can be calculated that to focus on a small spot it is better to have a smaller and a larger NA. But NA also defines the following important factors,

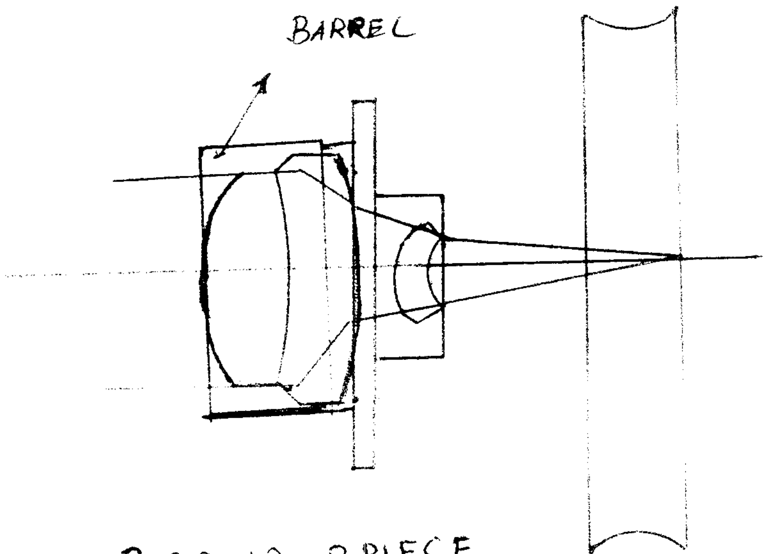
Depth of focus is proportional to Wave length / $(NA)^2$

Allowance for skew(tilt)is proportional to Wave length / $(NA)^3$

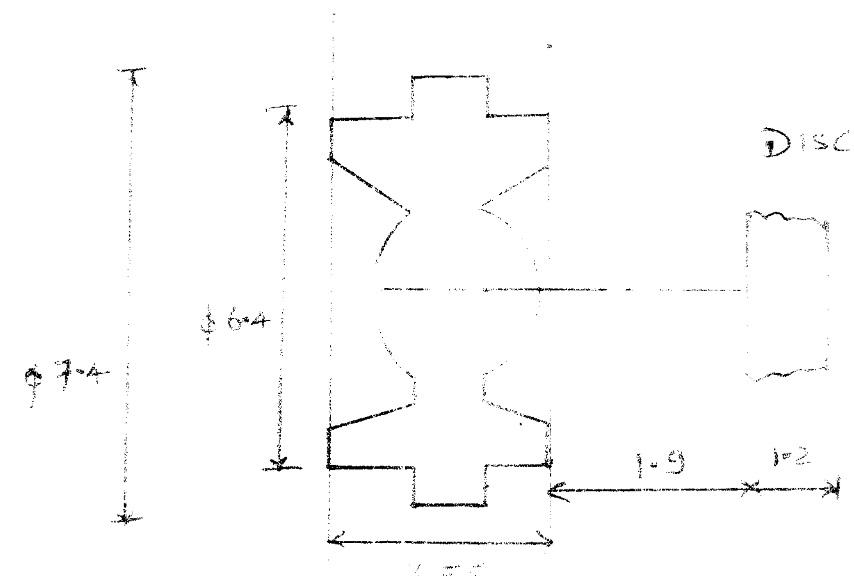
Allowance for variations in disk thickness is proportional to Wave length / $(NA)^4$

(d)





2 GROUP, 3PIECE
OBJECTIVE LENSE



For These Reasons,and NA which satisfies the following equation is recommended,

Wave length / NA <= 1.75

Accordingly, NA must be within the range of 0.45 to 0.50 in combination with the wave length of the Laser Diode.

MODULATION TRANSFER FUNCTION (MTF)

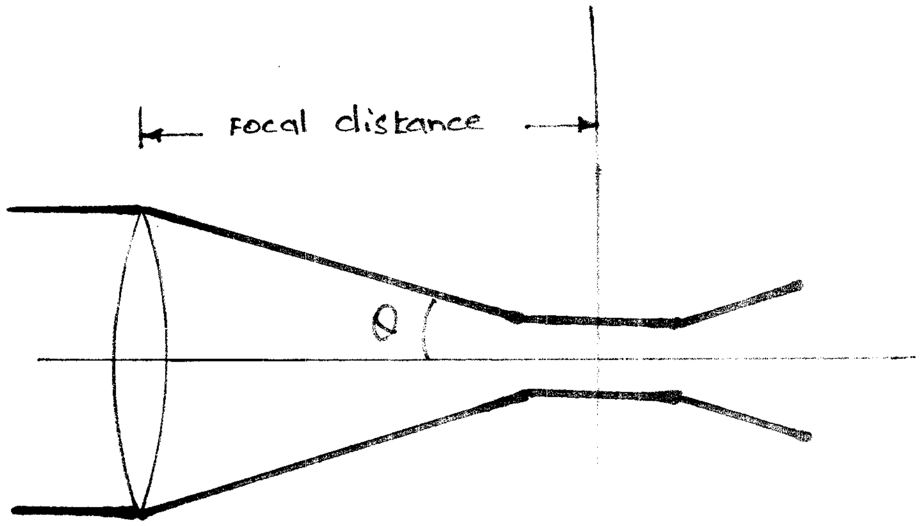
MTF is the frequency characteristics of the optical channel.In other words it is the parameter which determines the smallest size of pits that can be detected.To make this determination,the optical transfer function (OTF) is defined and expressed by a complex number.MTF is the absolute expression of OTF.The phase term of OTF is called the Phase transfer function(PTF).Generally,OTF is expressed by the cross-correlation function for the input and output apertures.In the case of a CD,a form of reflective optical disc,this becomes the auto correlation function in the following equation.

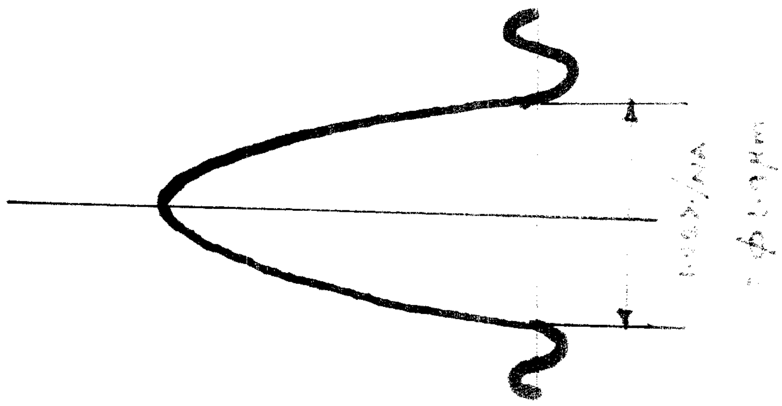
$$F(x) = \frac{2}{\pi} \cos^{-1} \frac{x}{x_o} - \frac{x}{x_o} \sqrt{1 - \left(\frac{x}{x_o}\right)^2}$$

Where x x_o

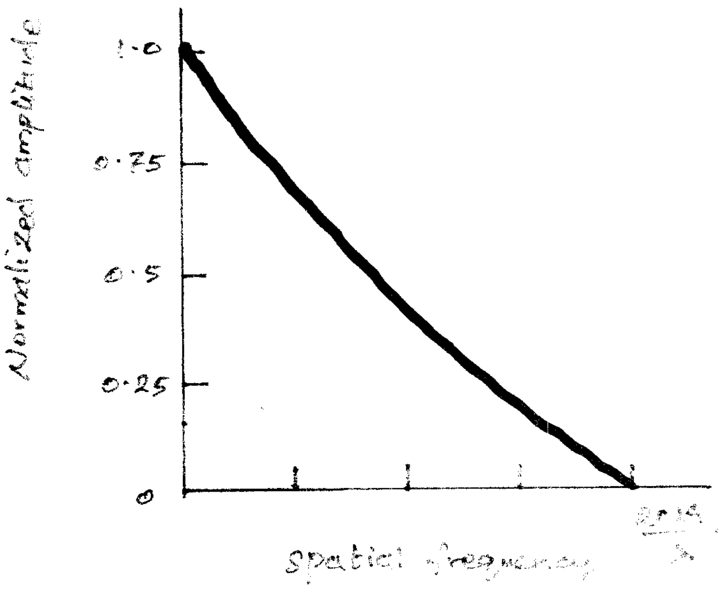
$$F(x) \leq 0$$

Where x > xo.Here x shows the spatial frequency,and xo shows the optical cut off;xo is expressed with a given NA and Wave length as follows :





Intensity distribution



Modulation transfer function MTF

$xo = 2NA$ / Wave length

As shown in figure it is the form of low pass filter(LPF).

In the case of a CD,

Wave length = 0.780 um

NA = 0.45 and

Spatial cut off frequency is $xo=1.154 * 10^{-}$

In other words,the optical system can detect pits as dense as 1154 per mm.

The smallest pit of a CD is about 0.87 micro metre at a linear velocity of 1.25 metre per second.If the track were occupied by these pits,the spatial frequency would be.

1 / (0.87micro metre * 2) = $0.581 * 10^{6}$

The wide band characteristics facilities accurate reading of the pit modulation over a wide range.

In terms of temporal frequency,the cut off frequency is 2 NA / Wave length * v = 1.44 MHz.

Where the linear velocity V = 1.25 m/s.

All the above equations are for ideal optics and ideal conditions.

ACTUATOR: -

It is attached at the end of the pick up.Since it is enough to work in the direction of tracking with a Swing - Arm system.Actuator as magnet and a objective lens are made into a single piece which is supported by a flat spring.This is moved up and down by the focus coil.ON the other hand,the slight type must have some degree of freedom in both focus and tracking direction.The requirements for this type of actuators are as,

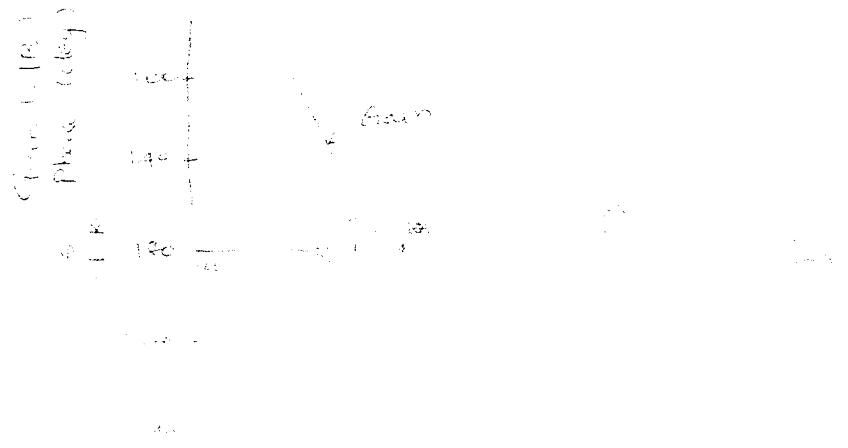
1) Cut-off frequency fo (fig iv)and the peak here must be appropriate for the servo response.

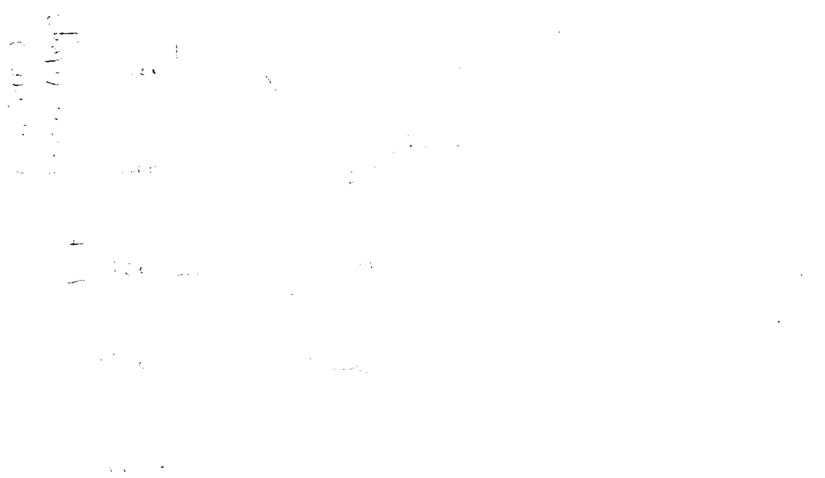
2) The secondary response frequency is much higher than the cut off frequency of the servo loop.

3) Good electrical sensitivity.(Small size,low power consumption)

4) No motion in directions that are not supported to have a degree of freedom.

Here introduce two types of two axis devices which is used to attain in focus and tracking direction.It has an off-center objective lens and an axis of rotation at the center.The lens rotation around this axis gives tracking while it moves up and down the central axis to focus.

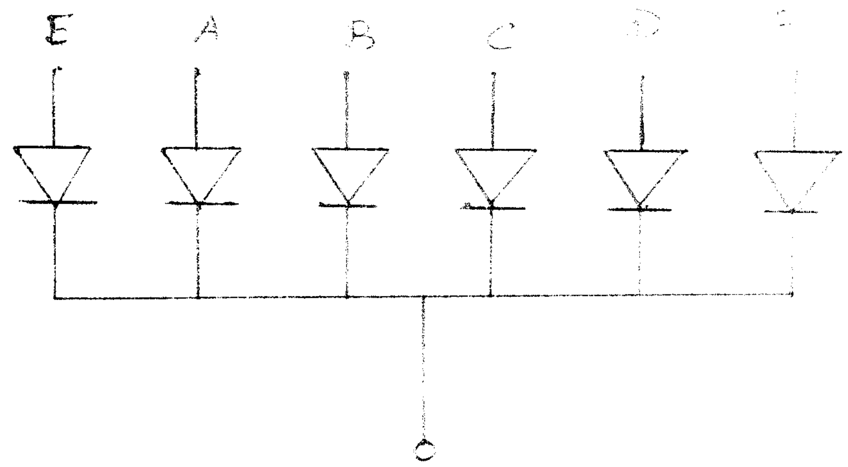

Gain

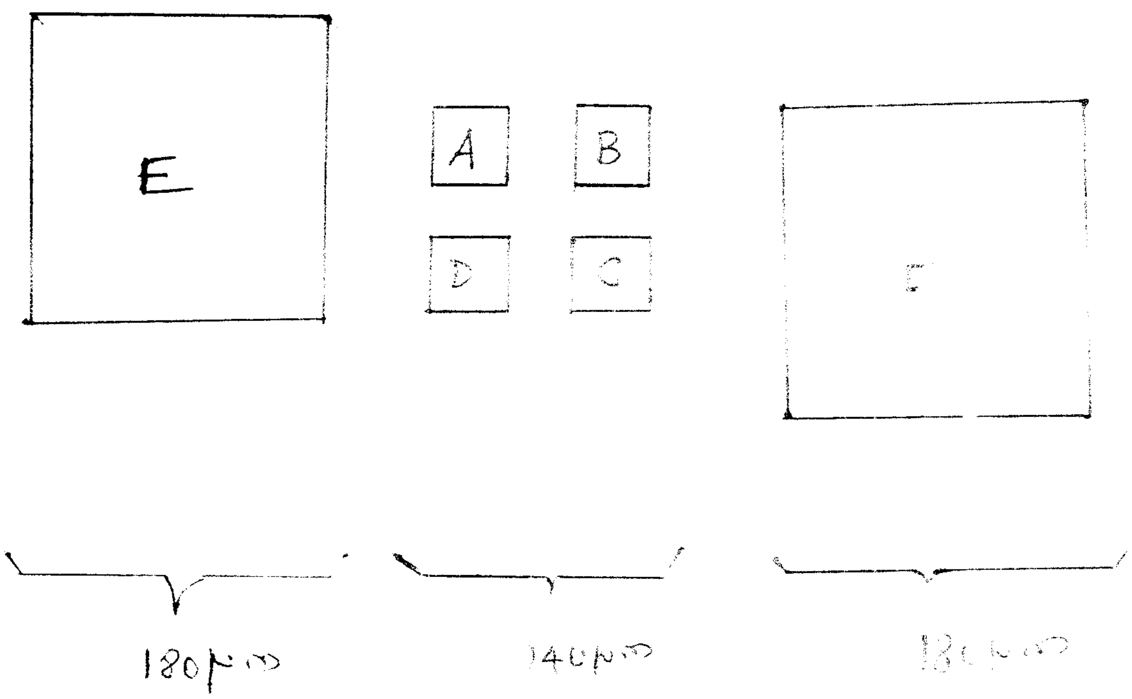


DETECTOR :-

The detector's shape and size are determined by the optical design and the methods used to operate the focus and tracking servos.Here a twin spot and astigmatisn combination that as commonly used the detection with the CD system.This is one in which a pin photo diode is divided according to the shape of returning beam.So that when the cathode is commom with a reverse bias across it,a current flows when light strikes it optical to electrical conversion.

Thus an (A + C) - (B + D) operation gives the focus error signal and (E - F) operation gives the tracking signal and (A + B - C + D) the neccessary signal current.





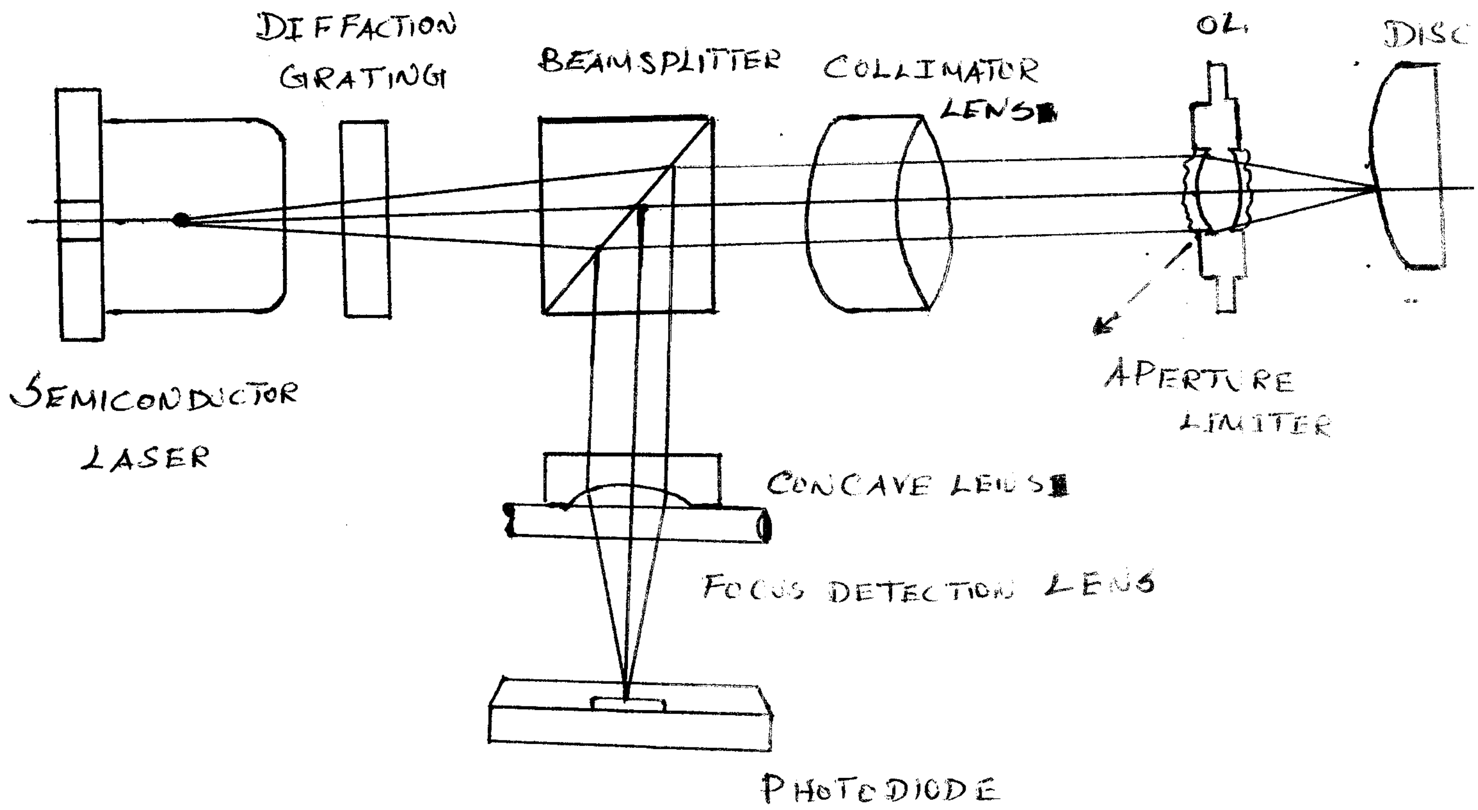
DETECTION STRUCTURE

Auto Tracking

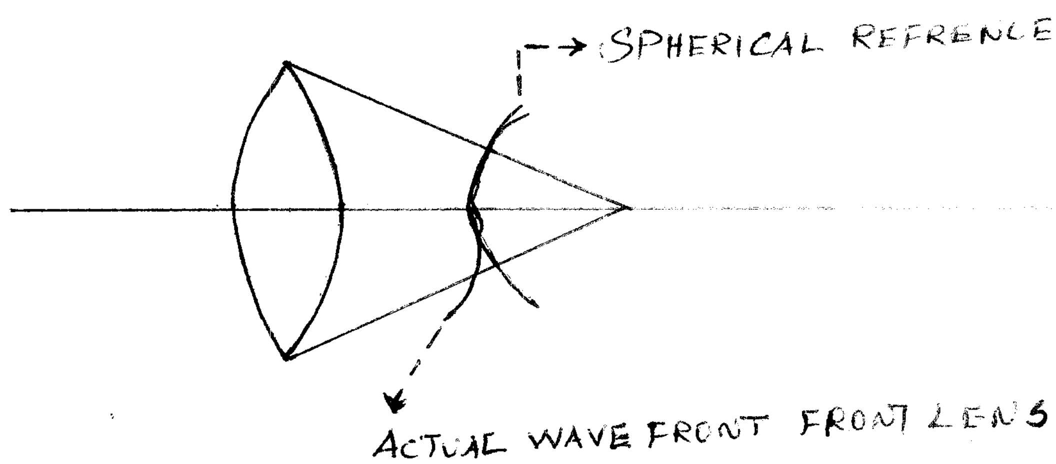
The Spiral pit track on a CD, running from a centre circle nearly to the edge, makes, 22, 188 laps over a 35.5 mm radial distance A disc uiat is off centre might exhibit track eccentricity of as much as 300 ums, which is the equivalent of 60 pit tracks. in addition vibration can challenge the pickip's ability to track within a ± 0.1 micrometer tolerance. We have clevised four methods for tracking the pit spiral (i) one beam push-pull (ii) one-beam differential phase detection (iii) one-beam high frequency wobble (iv) Three beam.

Auto focusing

Within 600.000 pits per second flying Past, the optical pickup has its work cutout for it. The disc specifications acknoledge this by allowing for a vertical deflection of ± 2 micrometer tolerance. Otherwise the phase along with audio data as well as in tracking information. The objective lens must therefore be able to refocs as the system manages this. Utilizing control electronics and a servo available for generating a focussing signal.


DIFFACTION
GRATING
BEAMSPLITTER
COLLIMATOR
LENS
OL
DISC
SEMICONDUCTOR
LASER
APERTURE
LIMITER
CONCAVE LENS
FOCUS DETECTION LENS
PHOTODIODE

FIG(i)


SPHERICAL REFRENCE
ACTUAL WAVE FRONT FRONT LENS

FIG(ii)

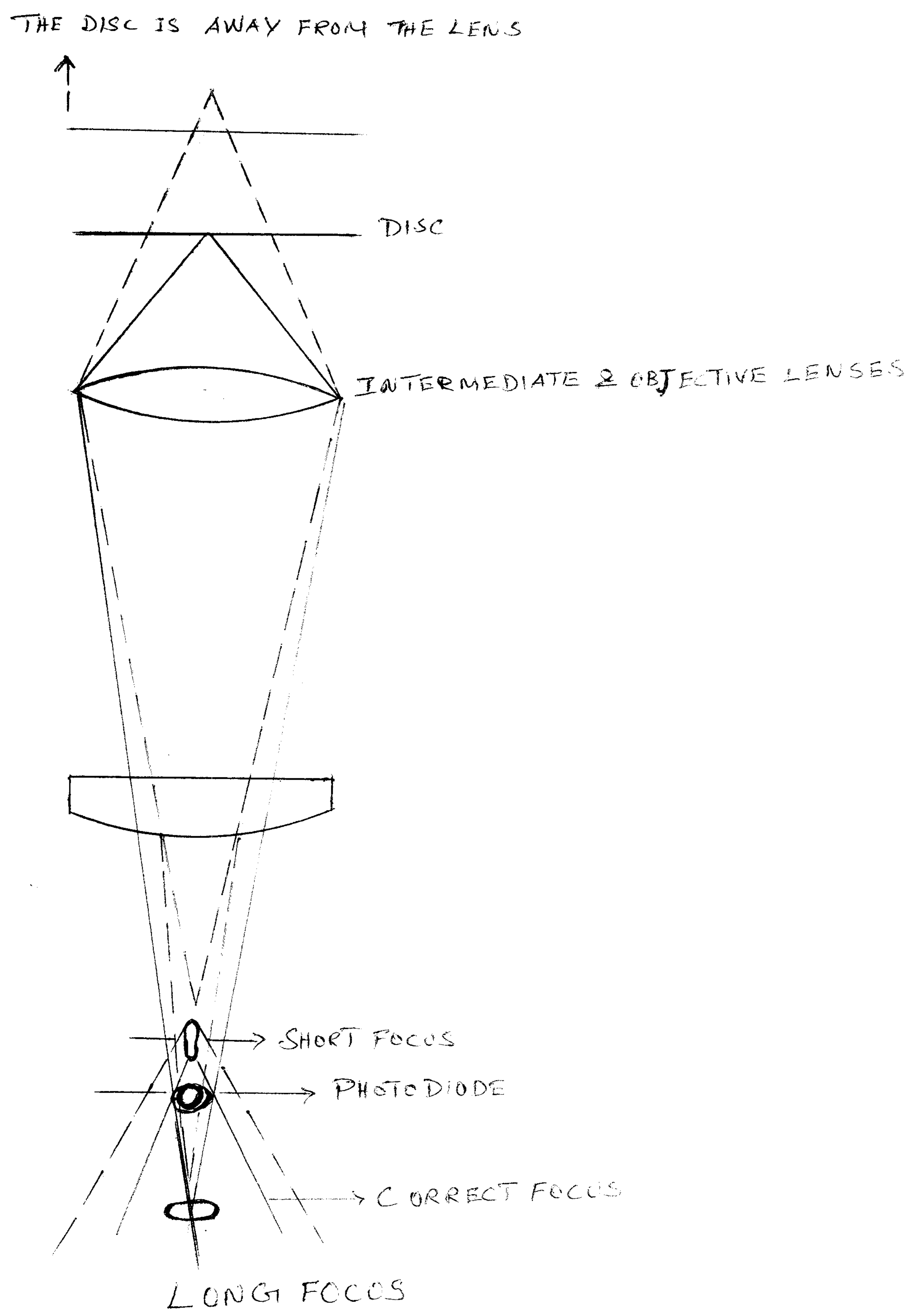
wavelength apart. The light diffracts when the resulting collection is again focused. It will appear as single, bright, concentrated beam with a series of successively less intense beams on either side. This diffraction pattern that actually strikes the disc. Three beam pickup uses the centre beam for reading data and focusing and two secondary beams for tracking only.

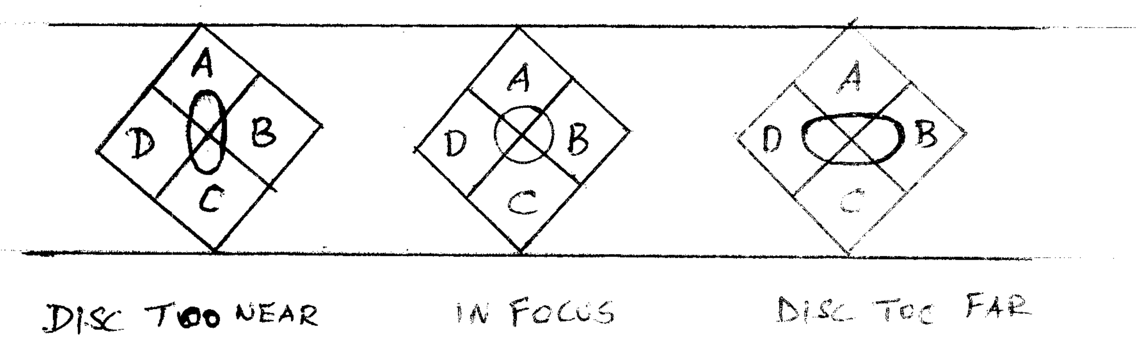
A polarization beam splitter (PBS) directs the laser light to the disc surface and angles the refelected light to the photosensor. The PBS consists of two prism with a common 45 degree face acting as a polarizing prism. The collimator follows the PBS the light then passes through a quarter wave plate (QWP) an anticotropic material that rotates the plane of polarization of the light beams. Reflected light from disc back again through the QWP will be polarized a plane right angles to of incident light. Because PBS will pass light in one plane (horizontal) and reflected light in the other plane (vertical) the PBS will properly deflect the reflected beam toward the photodiode sensor to read the digital data.

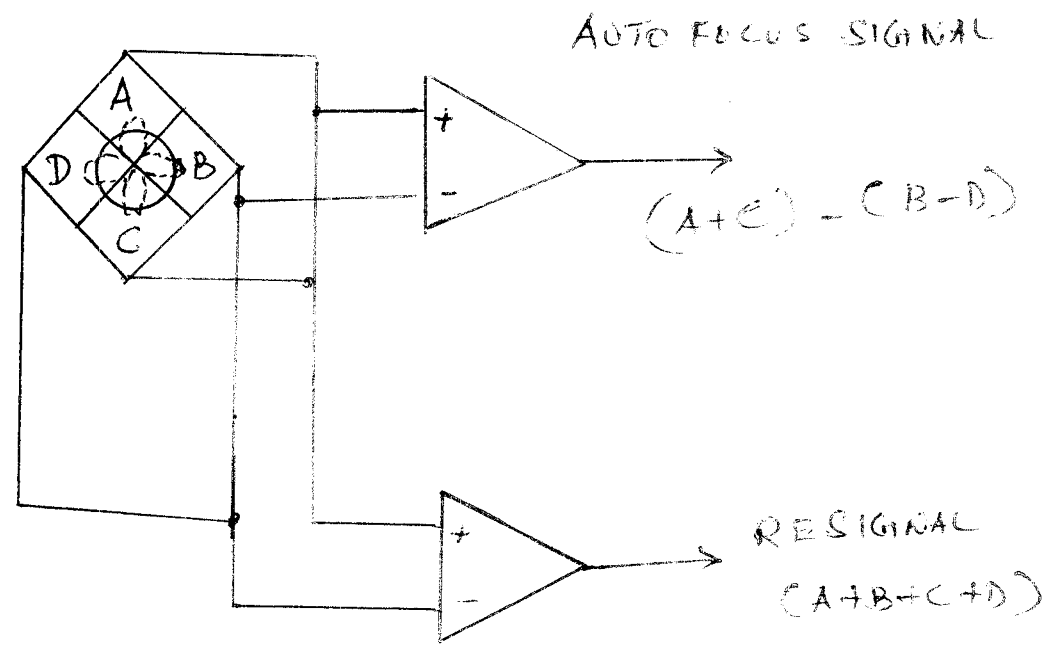
The data encoded on the disc now determines the fate of the laser light. When the spot strikes land. The smooth interval between two pits. The light is almost totally reflected when it strikes a pit with a depth in the transparent layer of about a quarter wave of the laser light diffraction and cancellation deu to interference cause light return through the objective lense. The QWP collimator lens and the PBS. Finally they pass through a single lens and a cylinderical lens enroute to the photodiode.

Three beam Auto Focus

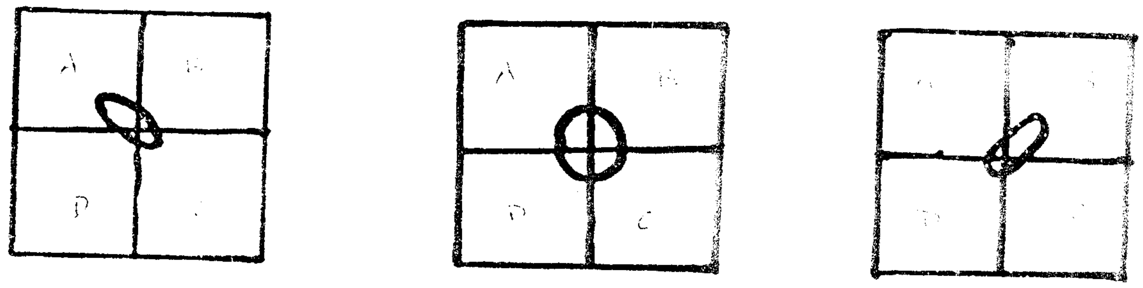
In a 3 beam player, the unique properties of astigmatism are used to acheive automatic focusing. The cylinderical lens just prefacing the photodiode. The change in image on the photodiode generates focus correction signal. The four-quadrant photodiode uses the light's intensity level from each of the quadrants to generate focus correction voltages. An elliptical pattern mainly striking quardrants A&C. When the disc is in focus no net voltage is created from the round Pattern. An elliptical pattern striking quardrants B&D indicates the disc is too far which creates a negative voltage. The centre beam is used to convey

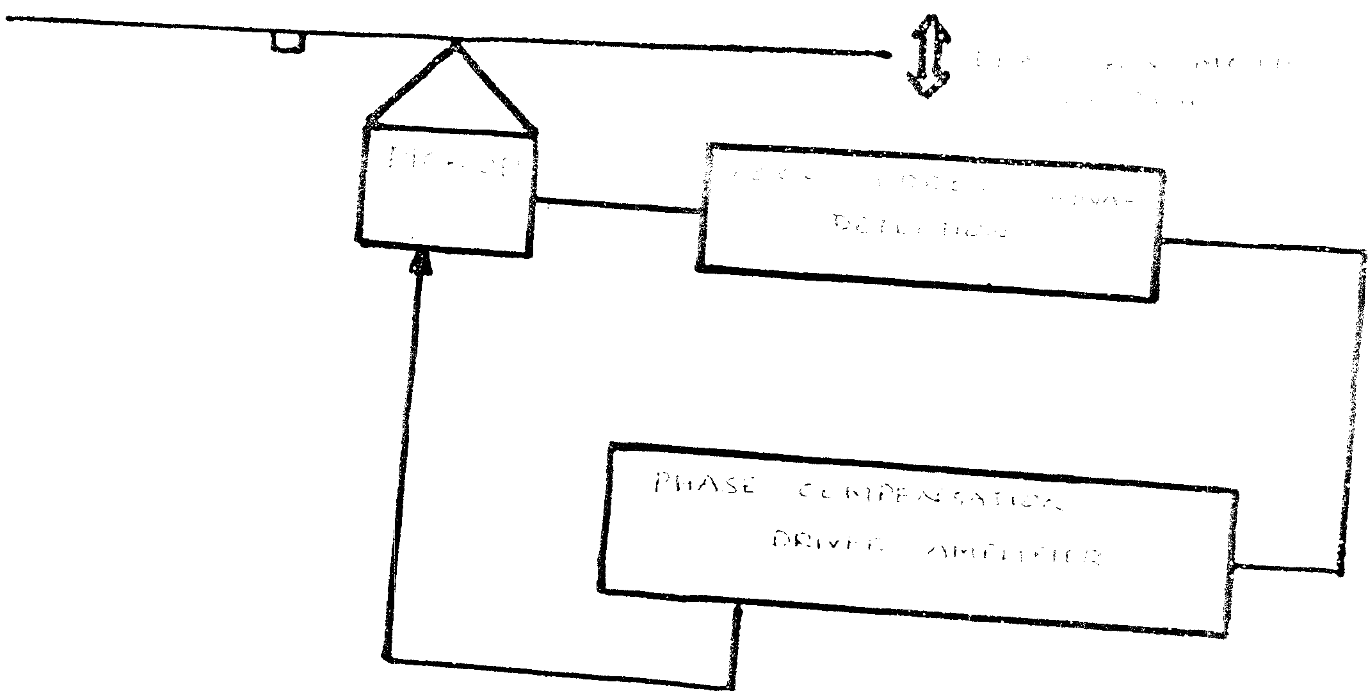

THE DISC IS AWAY FROM THE LENS
DISC
INTERMEDIATE & OBJECTIVE LENSES
SHORT FOCUS
PHOTODIODE
CORRECT FOCUS
LONG FOCUS


A
B
C
D
DISC TOO NEAR
IN FOCUS
DISC TOO FAR


AUTO FOCUS SIGNAL
A
B
C
D
+
-
(A+C) - (B-D)
RE SIGNAL
(A+B+C+D)

Fig( )





Fig( )

FOCUS SERVO BLOCK DIAGRAM

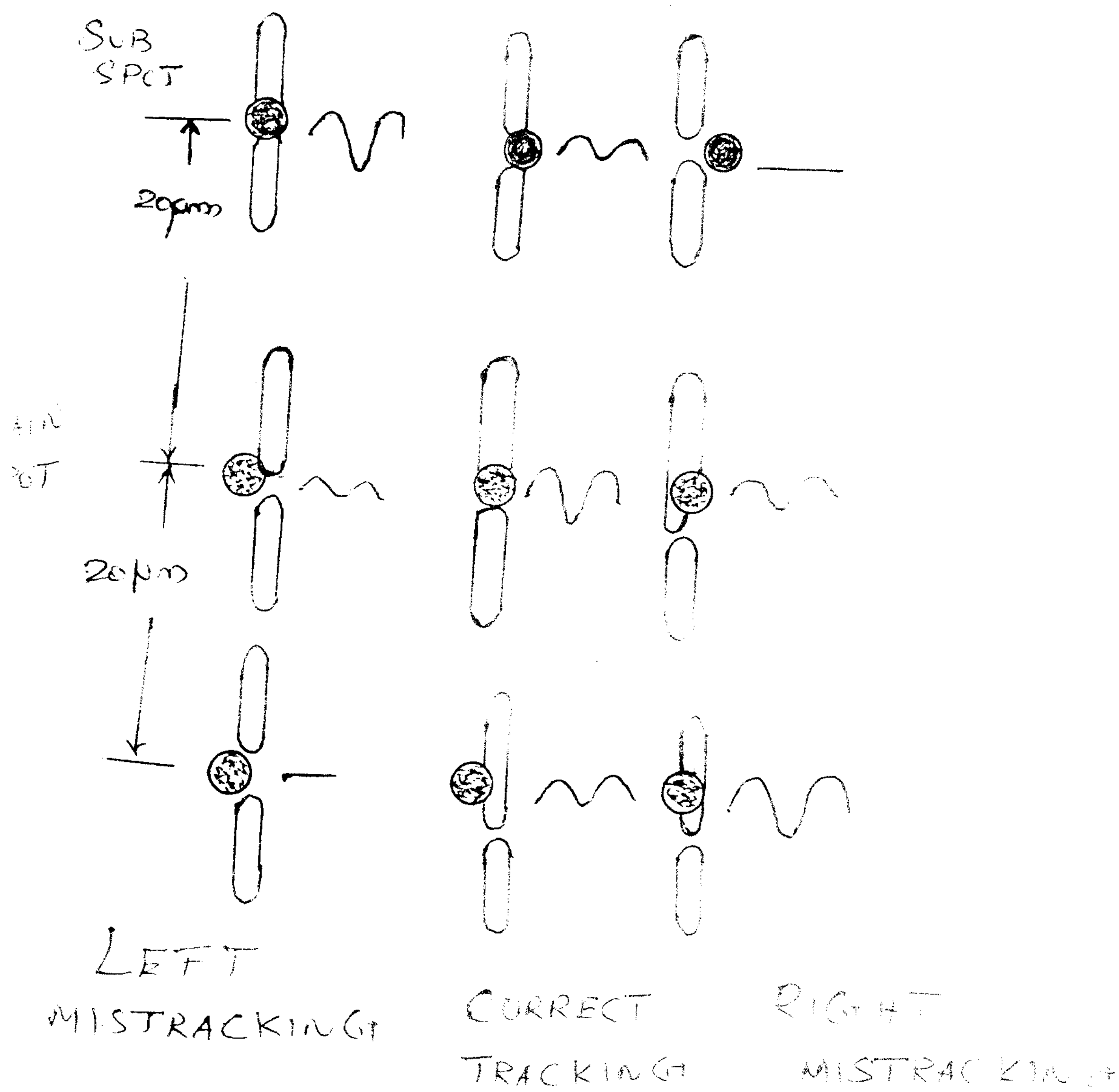
data from the disc using a signal summed from the four-quadrant photo diode (A-B+C+D).

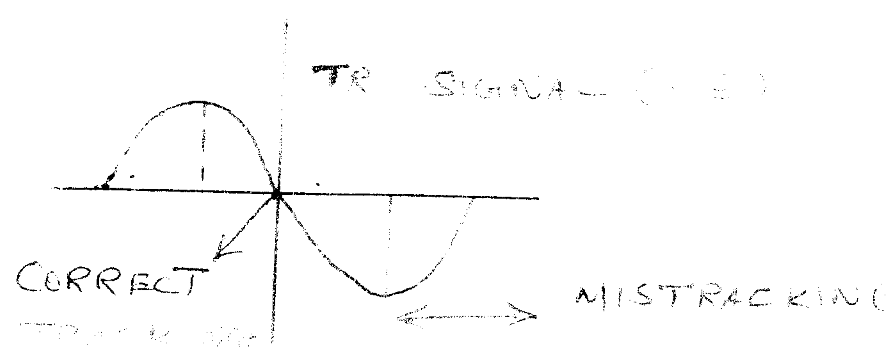
The correction voltage also generates a control signal in the form of [A-(A-(B-D))]. This control controls input the mechanical motion of the servomotor. When a disc is first loaded initial focus is acheived by injecting a signal objective lens. To move up and down several times until a focus condition is found. The focus maintaining with a tolerance of approximately ± 0.5 micrometer.

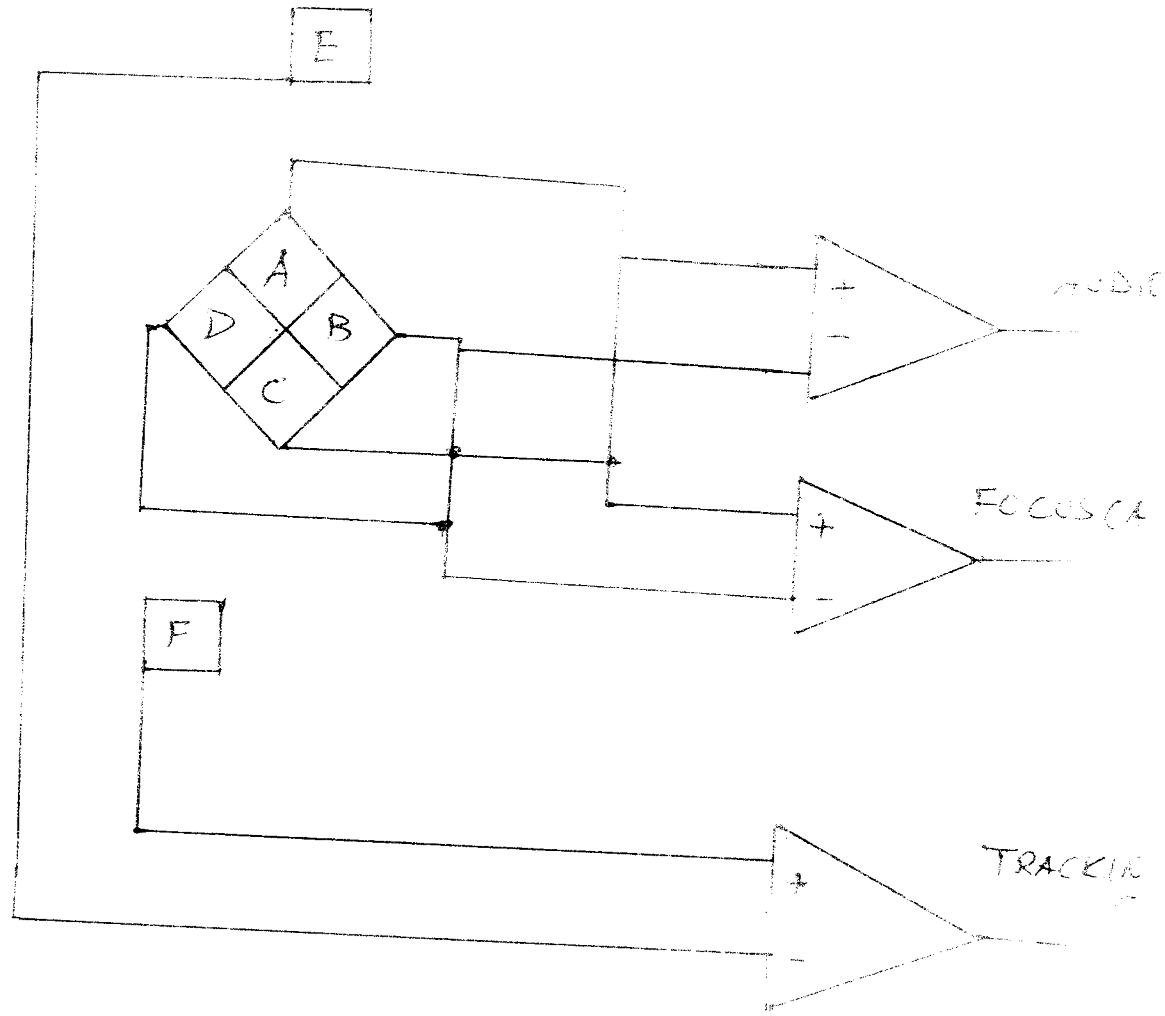
THREE BEAM AUTO TRACKING

The secondary beams are used for auto tracking. The centre beam which carries the information from the disc is accompanied by the two secondary beams from the point of generation at the diffraction grating. The tracking beam also strikes the data surface and are reflected. Their varying intensitives are used at two. Separate photo diodes mounted alongside the four quadrant photo diode.

The centre beam spot covers the pit track. While the two tracking beams are aligned above, below and to either side of the contre beam. Their relative position is firmly fixed. When the beam is tracking the disc properly,part of each tracking beam is aligned on the pit edge. The other part covers the mirrored land between pit track. The 3 beam are reflected through the QWP and PBS. The main beam strikes the four quadrat photodiode


SUB
SPOT
20µm
20µm
LEFT
MISTRACKING
CORRECT
TRACKING
RIGHT
MISTRACKING


TR SIGNAL
CORRECT
MISTRACKING



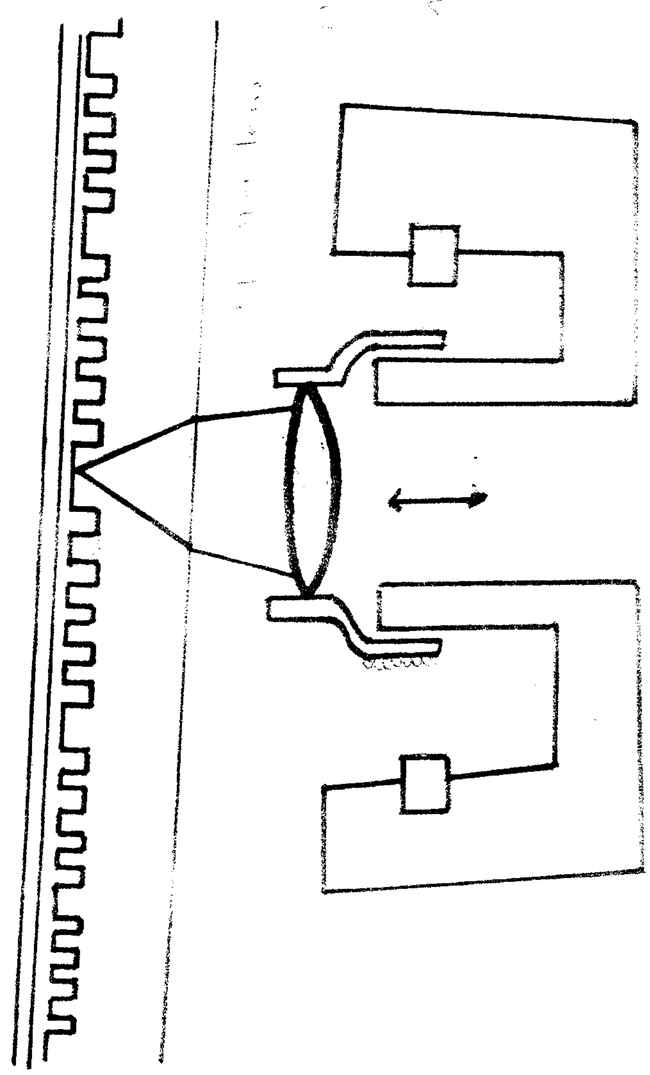
E
A
D
B
C
F
+
-
FOCUS
TRACKI

As the three spots drift to either side of thepit trade the amount of light reflected from the tracking beam varies. There is less average light intensity reflected by the beam which encounters more pits are and greater reflected intensity from the beam which encounters less pit area. The relative output voltage from the two tracking photo diodes from a tracking correction signal (F-E).

The varying output from the tracking photo diodes is converted into a useful correction signals. Because the tracking beams are aligned to different areas of the disc. The signal from the leading beam is delayed so micro seconds to permit comparison on the basic of the same pit as if they were reading the same pit simultaneously. If the tracking signal is perfectly aligned the difference is zero. if the beams drift a differential signal is generated varying positively for a left drift and negatively for a right drift.

THREE BEAM PICKUP MECHANISM

In many designs tracking and focus difference signals are applied to a two axis actuator. Within a focus tracking coil and a permanent magnet. The top assembly of the pickup is mounted on the base with a circular magnet ringing it. A circular yoke supports a bobbin containing both the focus and tracking coils. Control voltages from the focus circuit are applied to the bobbin focus coil and it moves up and down with respect to the fixed magnet. The OL thus maintains its proper depth of focus. In other



axis of movement from side to side is used to achieve tracking accuracy. When the tracking different signal is applied to the coil the bobbin swings around a shaft to move the lens laterally in the direction directed by the polarity of the signal so the main laser beam is again concentrated and the tracking signal is again zeroed.

A three beam pick requires a linear movement because the three beams must stay in a fixed position relative to the pit track. The three beam pickup is thus mounted in a sled that moves radially across the disc. The sled can be driven by a lead screw the pitch of the screws thread is a compromise between the fine thread required for normal tracking and the coarse thread needed for fast forward and reverse. in some design a moderate thread is used to achieve disc access and a movable mirror is inserted into the laser's path to detect the beam quickly to handle small differences. Such as track eccentricity. Alternatively, many designs use linear motors to provide smooth and rapid movement of the pickup across the disc surface.

Integrated Pickup

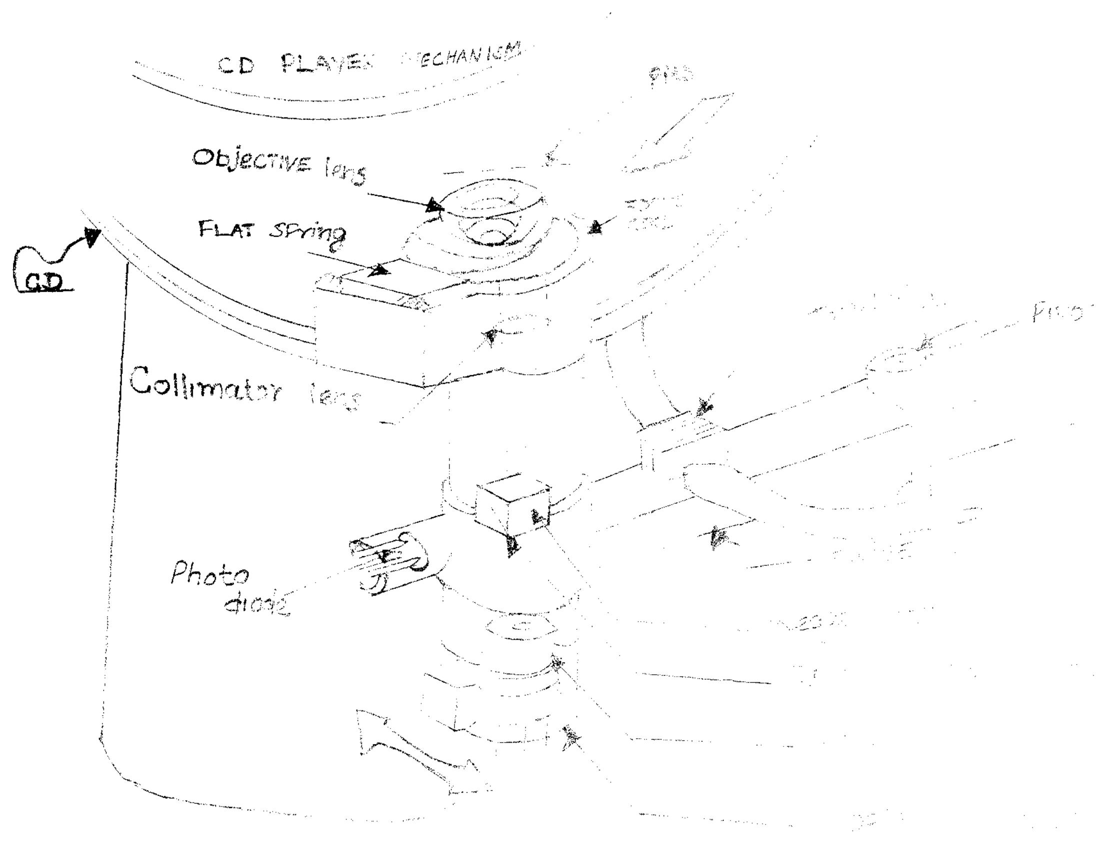
This is an area in which research is very active and it means in use of that is known as OIEC Pickup. The natural flow of things, including CD pickups is toward miniaturization and rather than aligning semiconductor. A diffraction grating is made using semi conductor processes and the light is focussed on the disc. A laser diode is attached and even through the concentration performance has not come up to the level that at which the CD signal can be read, there is no doubt that this form of pickup will emerge.

SLIDE MECHANISM

Depending upon the objective there are various ideas for slide mechanisms but the following three are commonly used in CD players.

1) Swing Arm

What is shown in Fig., is a conceptual diagram of an actual mechanism. As was mentioned earlier this differs from the other two and the tracking mechanism is a single unit, With the track taking the form of a circular arc. Because of this it is necessary for the servos to operate from a single beam, and in this example focus error detection uses the Focault method and tracking error detection the Push-Pull method. The mechanism is one in which an arm with relatively low inertia is supported in the middle by a pivot, with the pickup on one end and a counterweight on the other to maintain the balance. It uses a moving coil and

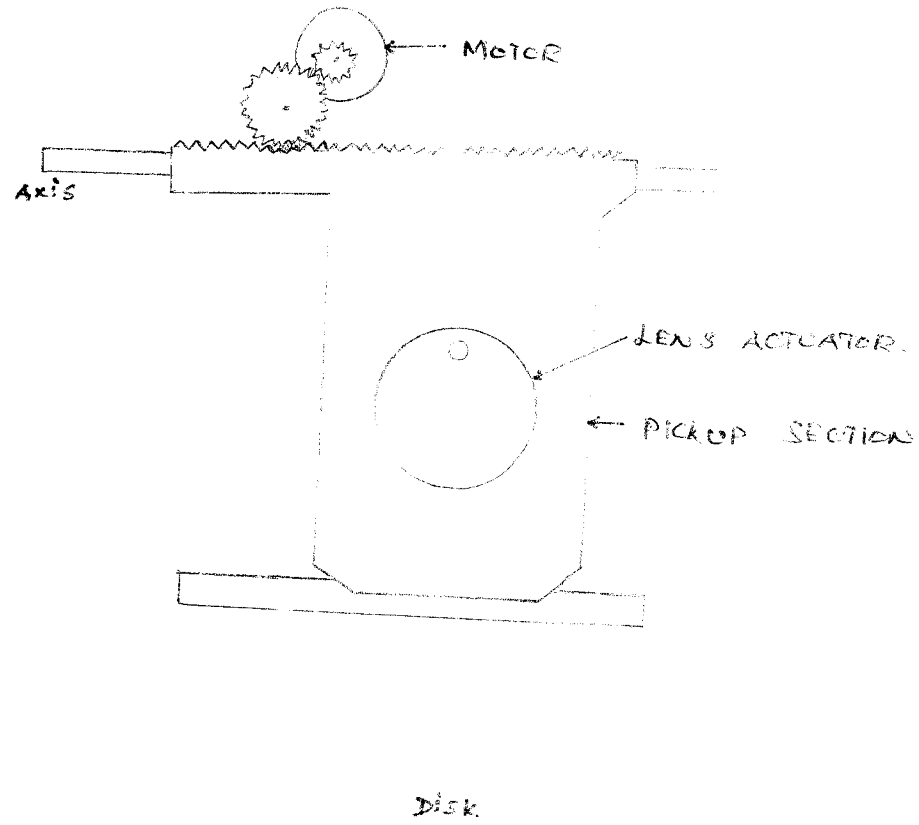


SWING ARM SYSTEM

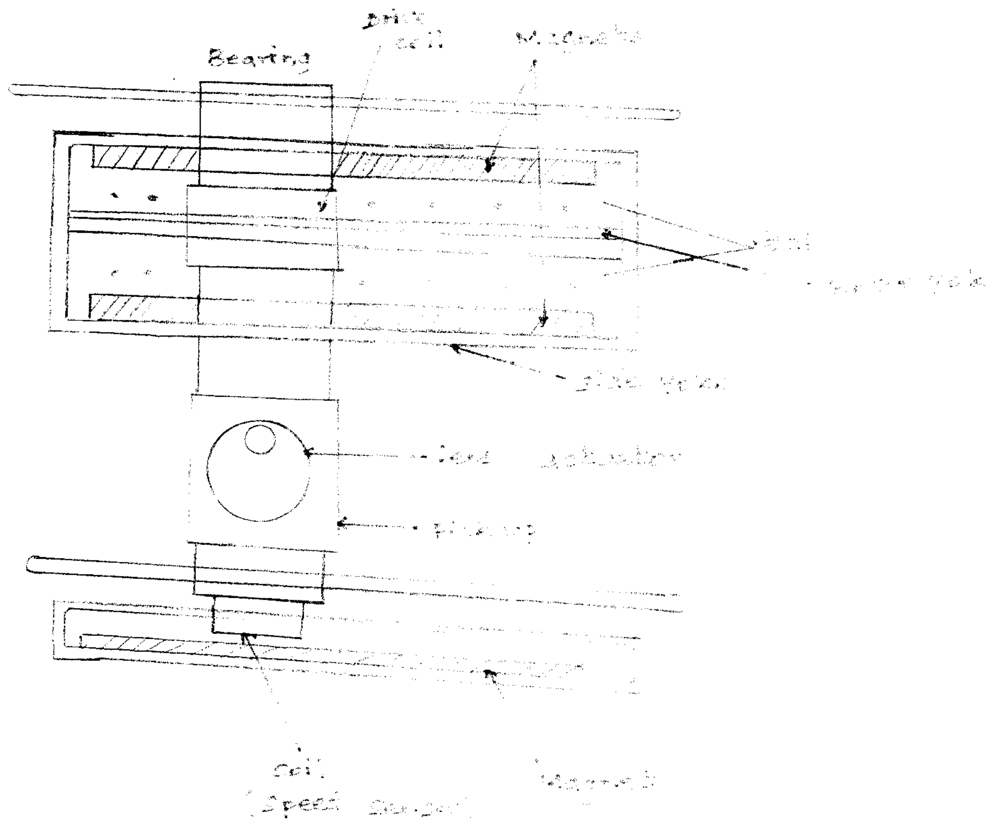
the drive coils housed in the arm. The permanent magnet is attached to the chassis. So that it takes the form of a linear motor. Since there is only one moving part in this construction the reliability of the mechanism is good but since the entire pickup is moved a large drive mass is inescapable. Given the response of the device, the design is complicated by the fact that low frequency secondary resonation is easily produced.

2) Rack-and-pinion or Screw methods

These use a two-axis device for tracking and are machine in which the entire device is moved by means of a rack-and-pinion set-up or device screw. When this mechanism is used to move the slide, a spring on the neutral point support for the two-axis device serves to produce the DC component and the slide is moved by means of this DC component. The two axes usually make for a movement that follows the track and keeps to center of the optical axis. With this system, care must be taken so that there is no backlash with the gears or screw. If there is, outside vibration con easily cause the tracking servo to slip, or there could be an area in which the servo lacks sensitivity. Compared with other mechanisms these have slow access, but since they are inexpensive they are often used in products for general use.



RACK-AND-PINION MECHANISM.



LINEAR MOTOR MECHANISM.

3)linear motor system

This is used when great importance is attached to access in things like CD-ROMs. The construction is one in which two axes support a moving section in which the optical pickup is mounted and the drive coil is attached to the section, as shown in Fig. Magnets are attached to slide yokes and two magnets have like poles (N for example) facing each other. Magnetic flux leaves the magnets forming the center and side yoke loops and creates a gap between the magnets and the yoke. The drive coil is situated in this gap and power is generated by the flow of current according to Fleming's rule (F=Bil B:density of magnetic flux. I: Current, l: effective length of coil). In fig., There is another coil on the side opposite the drive coil, and this acts as a speed sensor. That is, since it gives rise to the equation e=Blv (e: electromotive force generated by the speed sensor, l: effective length of coil, v: velocity of linear motor) the speed of the linear motor can be monitored by means of the current at this terminal. When something is accessed using a linear motor, a speed profile (in order to make the move in the minimum time, the time-speed curve for the distance to be moved is optimized) leading to the desired track is established and the profile and to apply breaking to the moving mechanism. The way in which the linear motor follows the DC component like a slide mechanism is the same as in (2).

Spindle Servo (Constant Linear Velocity). The specified CD linear speed typically is 1.25 m/s, but a speed of 1.2 to 1.4 m/s is permissible. On the other hand, a CD player must rotate the disk at exactly the same speed as when the signal was recorded in mastering. Spindle servo control is accomplished in two sequential stages as follows (Fig. 9.30):

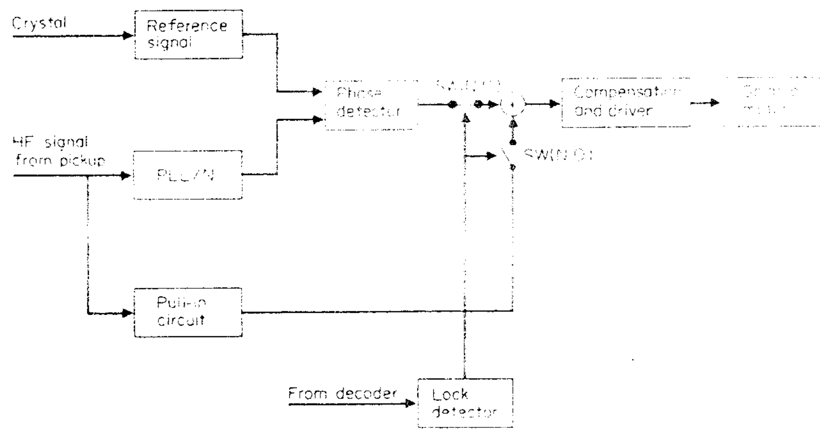


FIG. 9.30 Spindle servo (CLV).

1. *Pull-in stage:* The disk spindle motor is controlled by some means so that it rotates within the capture range of the phase-lock loop (PLL) used for clock recovery. In most cases this is done by detecting the T_{max} (the longest signal length) or T_{min} (the shortest signal length).

2. *Lock stage:* After confirming that the PLL is in *lock* condition, the spindle motor is locked to the reference signal from the crystal in the digital signal processor.

Summary of the servo circuit in a CD player

1. Foucs servo circuit : This servo circuit controls vertical movement of the 2 axis device and guarantees that the focal point of the laser beam is precisely on the mirror surface of the compact disc.

2. Tracking Servo circuit : This circuit controls the horizontal movement of the 2-axis device and forces the laser beam to follow the tracks on the compact disc.

3. Slde servo circuit : This circuit drives the sled motor which moves the optical block across the compact disc.

Spindle motor:

4. Disc motor servo circuit : This circuit controls the speed of the disc motor, guaranteeing that the optical pick-up follows the compact disc track at a constant liner velocity.

The focus servo circuit

Detection of the correct focal points. The reflected laser beam is directed to the main spot detector, an array of four photodiodes labelled A, B, C and D when the focus is on, the beam falls equally on the four diodes and the four error signal (Atc- (B+D) is zero on the other hand, when the beam is out of focus an error signal; is generated because the beam passes through a cylindrical lens, which makes the beam elliptic in shape the result focus error signal from the main spot detector (ATC) - (BTD) is therefore no zero.

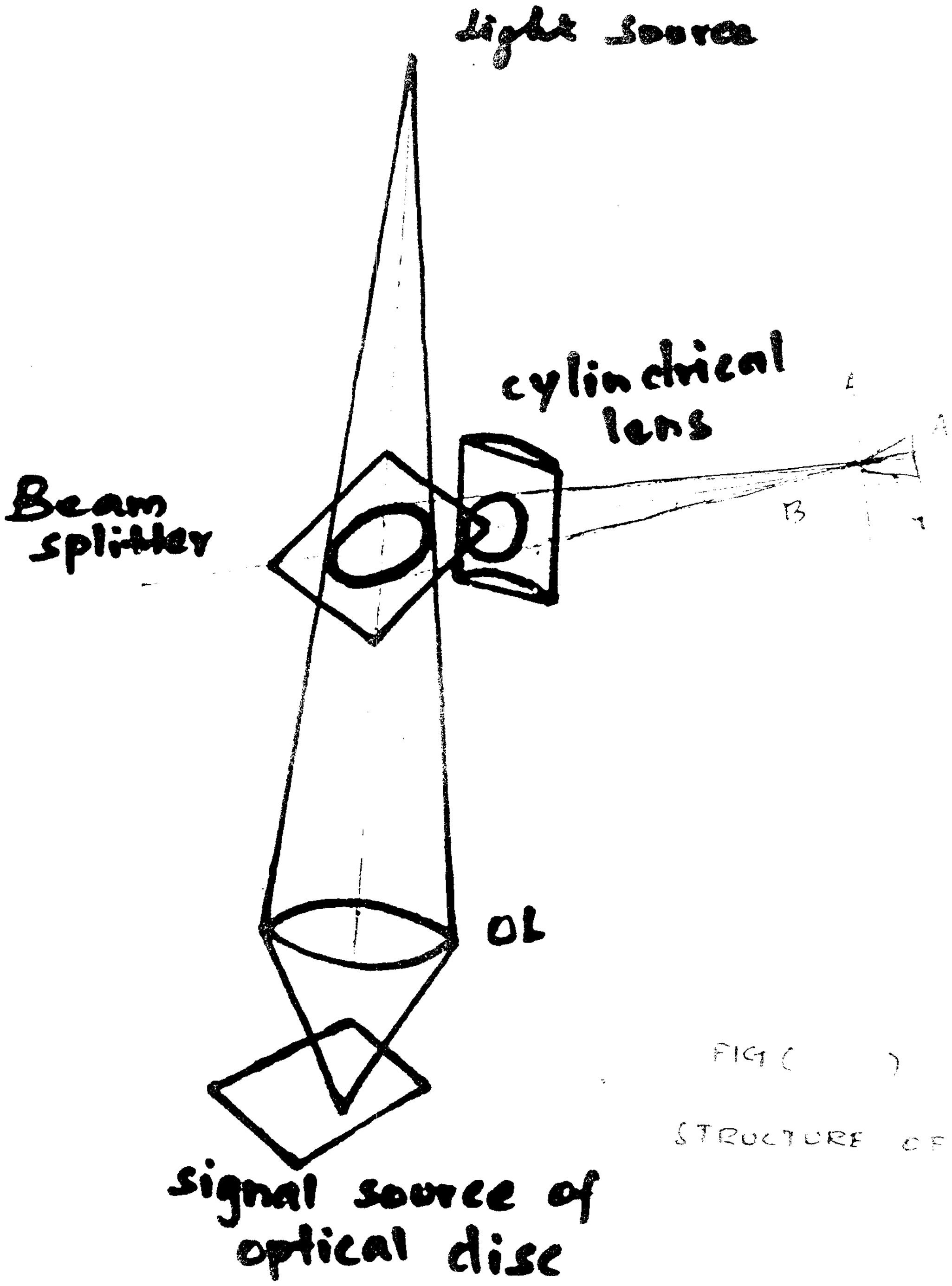


FIG (   )

STRUCTURE OF ASTIGMATISM SYSTEM.

The focus search circuit

Where a disc is first loaded in the player the distance between the 2 - axis device and the disc is too large. The focus emx signal is zero and the focus servo circuit is inactive.

Therefore, a focus search circuit is based which, after the disc is loaded, moves the 2 axis device slowly closer to the disc output of the four photodioder are combined in a different wacf (ie AtBtCtD) to form a radio frequency (Rf) signal, which represent the data bits read from the disc when the Rf signal exceeds a threshold level, the focus servo is enable and now controls the 2 axis device for a zero focus error signal.

Figl shows the three possible tracking situations as the optical pick-up follows the disc track. In fig-1a and 1b the main spot detector is not currently tracked and so one or other of the side spot detector gives a large output signal as the pit the transversed in fig , on the other hand, the main spot detector is correctly track and both sides spot detector gives small output signals.

DIGITAL SIGNAL PROCESSING

Digital signal processing (DSP) is a powerful technology and in fact without DSP the CD system would not be a viable reality It is DSP that permits error correction, digital filtering. low-bit D/A conversion and other crucial CD player operations. Before we tackle the important topics of digital filtering and low-bit conversion. let's take a look at DSP.

DSP is a technology used to analyze, manipulate or generate signals in the digital domain. A signal is a time-based sequence in which the ordering of values is important. For example a digital audio signals makes sense and can be processed properly. only if the sequence is properly preserved. DSP is thus a special application of general data processing.

DSP is the workhorse of modern engineering. Its extensive applications in telecommunications includes modems cellular systems data transfer between computers, vocoders and transmulti-plexers. DSP image processing is used to process photographs received from orbiting from orbiting satellites. Television studios use digital signal processing techniques for manipulating picture signals. other applications include robotics, machine vision, pattern recognition, digital television, computer anima-tion, flight simulators, missile / torpedo control, vibration analyzers, navigation systems, automobile anti-lock brakes, active suspension and motor and emissions controls.

Of course, digital audio systems also present rich possibilities for DSP. Error correction, multiplexing, sample rate conversion, speech and music synthesis, data reduction, volume /fader/balance, filtering,adaptive equalization, dynamic compression and expansion, crossovers,reverberation, ambience processing, time alignment, acoustic noise cancellation, mixing and editing and acoustic analysis can all be performed with digital signal processing. An audio DSP system is in reality a computer dedicated to the processing of audio signals.

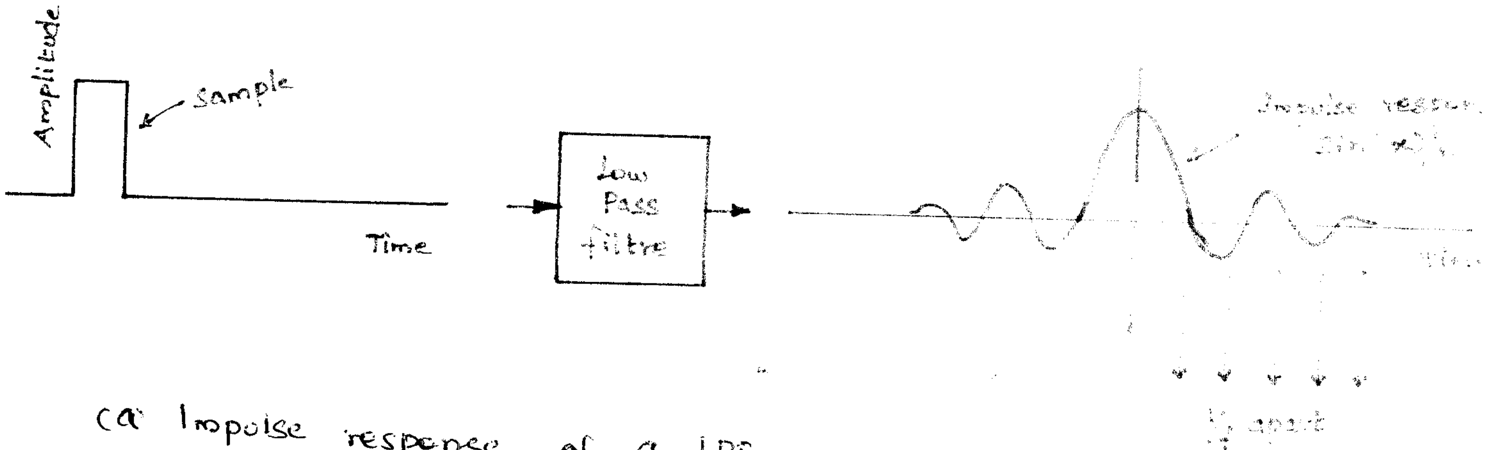
Advantages and disadvantages

In theory, digital signal processing like digital audio in general, provides many performance advantages over analog signal processing. Digital processing is more precise and repeatable, and it can perform operations that are impossible with analog techniques. Noise and distortion can be much lower with DSP thus audio fidelity is much higher. In addition, whereas analog circuits age, lose calibration and are susceptible to damage in laser environments. DSP circuits cannot lose calibration and are much more robust. In that respect the technology is well suited for consumer applications where everyday use over long periods puts great wear on products.

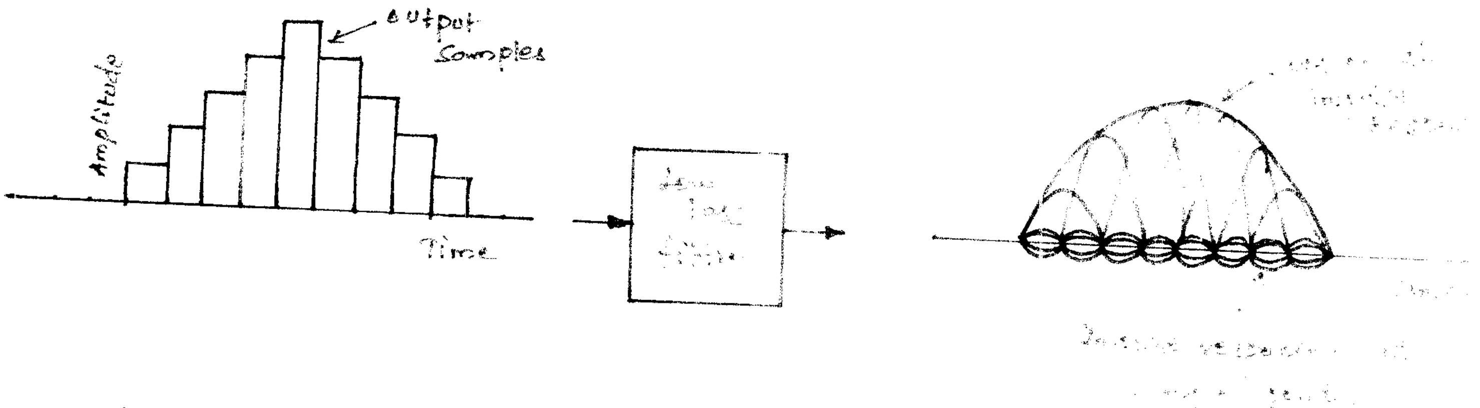
However DSP is not a simple technology to implement. The challenge becomes apparent when design engineers begin to a mass the hardware and software needed to process the signal. Aside from the sheer volume of computations needed, a number of difficult problems can arise. For example, truncation errors occur when the numerical result of a calculations such as addition is limited to a certain number of bits without regard to the value of the number. If a number such as 2.88 was simply truncated to 2.0 a considerable error would occur and the error would be compounded as many calculations take place, each using truncated results. The resulting numerical error would be manifested as a distortion of the output signal. To avoid this all DSP computations on audio signals must be of extremely high accuracy. This requires long word lengths ; many DSP chips employ digital words that are 32 bits in length of longer.

In addition even simple DSP operations may require several intermediate calculations and complex operations may require hundreds of operations. To accomplish this, the hardware must execute the steps very quickly. Because all computations must be accomplished in real time-that is the span of one sample period (perhaps 22 microseconds) - the processing speed of the system is crucial. DSP chips must often process twenty or thirty million instructions per seconds in fact DSP chips are among the fastest processors in use.

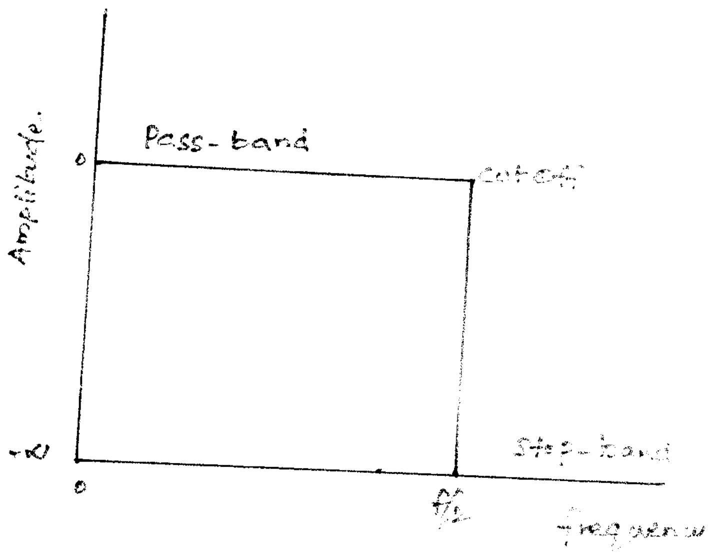
(a)



(a) Impulse response of a LPF



(b) Summing of impulses forms the output



(c) Ideal LPF Characteristics

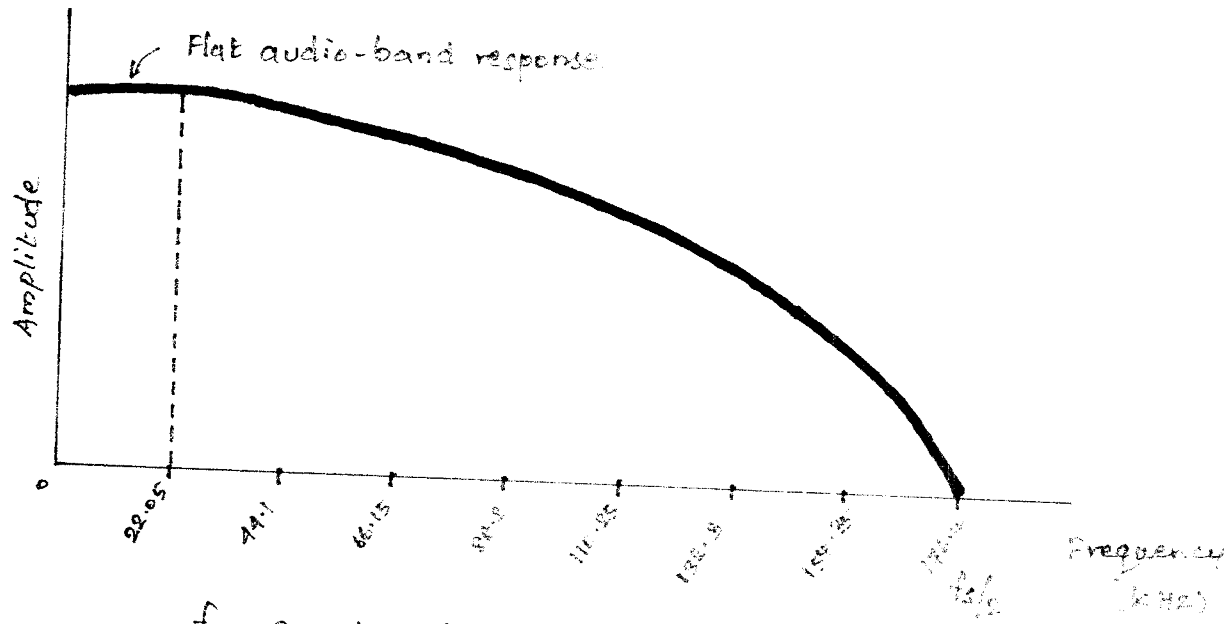
FILTER DESIGN

Classic digital design dictates an analog brickwall filter at the input and output of the digitization system. Today, most systems, employ digital filters. However to understand the purpose of filtering and the advantages of digital filtering, our discussion must begin with a look at analog filters.

Analog Filters

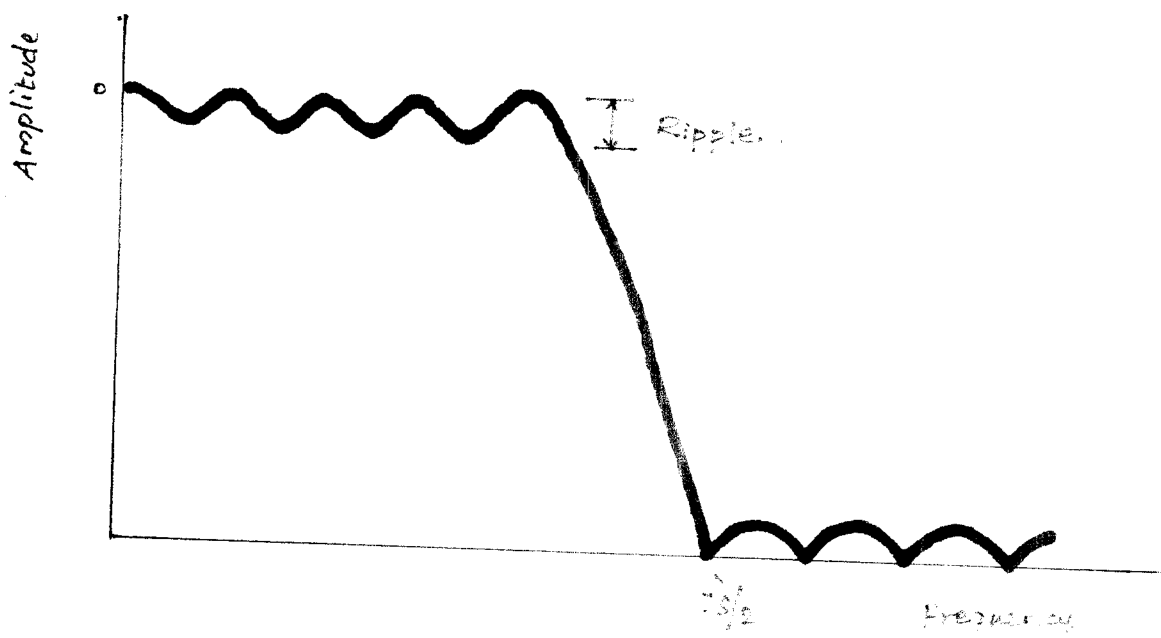
Filtering is a fact of life for digital systems. An input anti-aliasing filter must precede the sampler to uphold the Nyquist theorem's criteria for bandlimited and thus lossless, sampling. Similarly, the output anti-imaging filter must filter out all frequencies above the half-sampling frequency. The analog signal at the output of the digital-to-analog converter is a pulse amplitude modulation waveform whose staircase appearance is easily spotted by waveform watchers. These sudden shifts in amplitude represent high-frequency components not present in the original analog waveform; such artifacts of sampling must be removed to create a smooth waveform that eliminates high - frequency components. Even when digital filters are employed, a gentle analog smoothing filter is still required at the output.

It is the output filter that reconstructs the audio waveform. Although the idea of smoothing the output samples to remove high frequencies is entirely correct. A more mathematical analysis shows the inner workings. A low-pass filter has a sin x)/x impulse response.This means that a single rectangular audio sample (of infinite width) passing through an ideal low-pass filter would emerge with a waveform in the shape of this sin(x)/x



f_s = Sampling frequency = 352.8 kHz

(a) Problems associated with brickwall filters can be overcome with more gentle filter characteristics.



(b) Actual LPF's exhibit ripple in both the Pass and Stop bands.

curve, as shown in fig., . If the low-pass filter has a cutoff frequency at the half-sampling point ($f_s/2$), then the $\sin(x)/x$ curve passes through zero at multiples of $1/f_s$.

When a series of audio samples passes through the filter, the resulting waveform is the summation of all the individual $\sin(x)/x$ contributions of the individual samples, as shown in fig . Each sample's impulse response is zero at the $1/f_s$ position of any other sample's maximum response. The output waveform has the value of each sample at sample time and the summed response of the impulse of all samples in between sample points. This superposition of individual impulse responses results in an output waveform that is identical to the input low-pass filtered audio waveform. It is thus the filer's impulse response to the audio samples that reconstructs the original waveform.

Sharp - eyed readers might question the need to worry about frequency such as 88.2 or 176.4 KHz. since they lie so far above the limits of audibility. Why waste money on a filter when the ear itself is rather effective at filtering everything beyond 20 or 25 KHz? The original waveform is reproduced without filtering but the accompanying image bands could stress the player's output amplifier or cause intermodulation in downstream analog equipment through which the signal passes. For example, the high-frequency components might create intermolecular components with base-band frequencies of with the high-frequency bias current of an analog tape recorder. That, in turn could add audible distortion to the signal. Filtering is therefore mandated.

IDEAL FILTERS

Because engineers have been designing analog filters for a long time. We would expect little trouble with this particular assignment. Both the input and output filters can share similar designs and the design criteria can be easily summarized. Ideally we would like to attenuate all audio frequencies above the half - sampling frequency without affecting the lower audio instantaneously so that the usable bandspace is extended as far as possible in order to yield an extended and flat frequency response. Thus an ideal filter, such as the one in fig., would have a flat pass-band (the audio range the filter passes) an immediate or brickwalls filtering characteristic and a stop-band (the frequencies the filter eliminates) attenuated to below the system's quantities resolution. In addition to these frequency response criteria, an ideal filter would not affect the phase or any time-domain characteristic of the signal.

ACTUAL FILTERS

Although an ideal filter may be approximated, in practice its implementation presents a number of engineering challenges. A brickwall design requires compromise in other specifications, such as flat pass-band and low phase distortion. To alleviate these problems inherent in a brickwall response we could design filters with more gradual cutoff which would not exhibit phase nonlinearity. However the frequency of the half-sampling point would have to be increased to make sure that it was placed in a sufficiently attenuated part of the filter characteristic. This is the case shown in fig., . Consequently a higher sampling frequency would be needed to achieve the same flat audio frequency response. To limit the sampling rate and make full use of the bandspace below the half-sampling point, a brickwall filter at both input and output of the system, is the only alternative unless, as we shall see later, the technique of oversampling is employed to increase the sampling rate.

Let's consider an analog output filter design, such as those found in some early compact disc players. With a sampling rate of 44.1 KHz, the output filters (one for each channel) are usually designed for flat response from DC to 20 KHz. This provides a guard-band of about 2 KHz to ensure that attenuation is sufficient at the half-sampling point. The pass-band undoubtedly exhibits some frequency response irregularity, called ripple, which is typically specified to be less than ± 0.1 dB, as represented in fig. . The stop-band's attenuation is designed by word length. Ideal a 16 bit system would require a stop-band attenuation of greater than 95 dB. The stop-band also typically exhibits ripple.

CASCADE FILTER DESIGN

Although there are many types of filter designs, only a few are able to achieve the performance criteria required of a high fidelity digital audio system. One filter type often employed is an elliptical filter, sometimes called a Cauer filter. The design generally uses a low-pass filter combined with a band-reject notch filter to sharpen the cutoff. The elements of an elliptical filter interact with each other, making them challenging to design. High precision parts are also required.

Basic filter elements of various types can be cascaded (repeated in series) to sharpen the cutoff. As the number of cascaded stages is increased, the filter characteristic steepens until the ideal filter frequency response is approximated. Unfortunately as the cutoff steepens, the phase shift increases as well. Compact disc players might require ninth-order filters with characteristics such as those indicated in fig. . The cutoff shown in fig looks like the north rim of the Grand Canyon but the phase shift shown in fig., might exceed 90 degrees at 20 KHz. This phase shift introduces nonlinear time delay across the audio band.

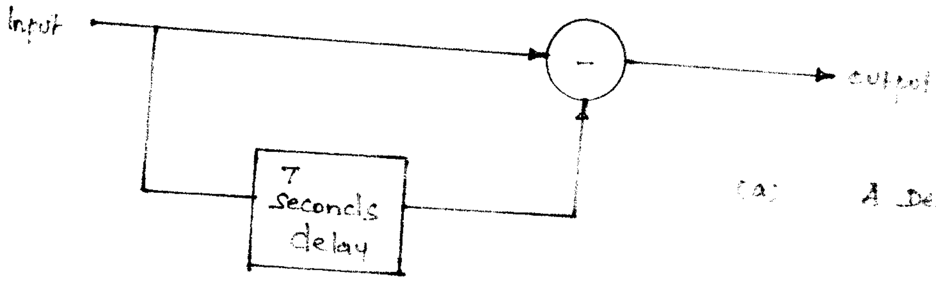
The resulting group delay shown in fig., which measures the change of phase shift with respect to frequency, causes high frequencies to be shifted in time (delayed) relative to lower frequencies. The delay increases toward the cutoff frequency of an analog filter in a CD player. This creates a kind of time smear in the signal. (Ultrasonic frequencies are theoretically delayed "forever" which is how they are filtered out).

Absolute non-frequency dependent phase shift is inaudible. Recordings of Enrico Caruso may be ninety years old, which is a lot of time delay. but the CD suffers nc distortion on that account, because all of the frequencies of the recording have been delayed is different by the amount. But frequency-dependant group delay is different. Conceptually we might compare such delay to the worst case example of an extremely strange loud-speaker in which the tweeter is located a mile behind the woofer. If properly equalized, steady-state tones would sound fine. But when the tone ended we would hear sound emanating from the tweet-er a full 5 seconds after the sound from the woofer had ended. Conversely, at the onset of a tone, the tweeter's attack would arrive 5 seconds. Such long frequency-dependent delays would obviously be audible. What about a real brickwall liter, in which delay at 20 KHz might be 300 microseconds relative to 0 Hz? Is that audible? The answer has yet to be fully determined, but manufactures have certainly hesitated to feature the relevant specification in full color advertisements. obviously we would prefer to avoid any such group delays in frequency response. The point is that the analog filters necessary in CD players are not trivial design exercise.

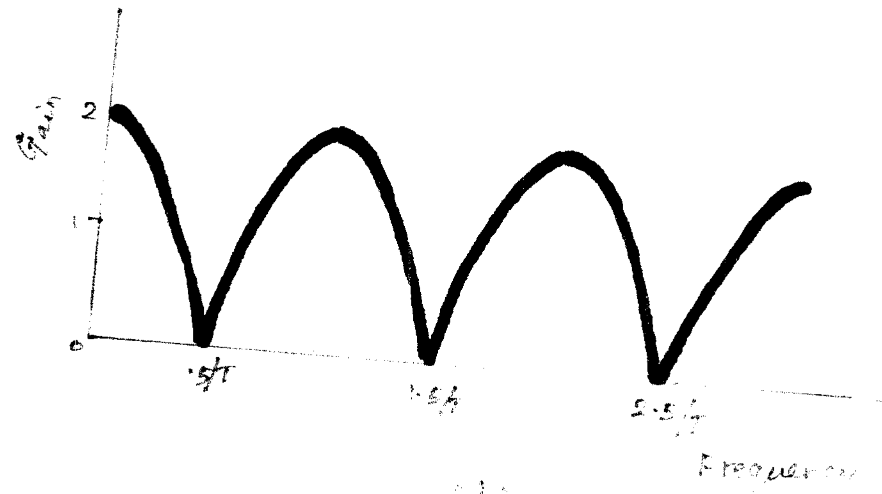
DIGITAL FILTERS

The digital filter used in today's CD players is a circuit which accept audio samples and outputs audio samples. In other words, it is a DSP application. Filtering takes place precisely because the values of the output samples have been altered to produce filtering. In this case the digital filter simulates the process of low-pass analog filtering and the waveform reconstruction it provides. Moreover, a technique called oversampling is utilized in which additional sample values are computed by interpolating between original samples. Because additional samples have been generated (perhaps two, four, or eight times as many) the sampling rate of the output signals is greater than the input signal.

Each intermediate sample is multiplied by the appropriate sin(x)/x coefficient corresponding to its contribution to the overall impulse response of the filter. The multiplication products are summed together to produce the output the output filtered sample. It thus digitally simulates the impulse response of an analog filter. the spectrum of the signal is changed, with the images appearing at multiples of the new (oversampled) sampling rate. Because the distance between the baseband and sidebands is larger, a gentle analog filter can be used to remove the images without causing phase shift or other artifacts.

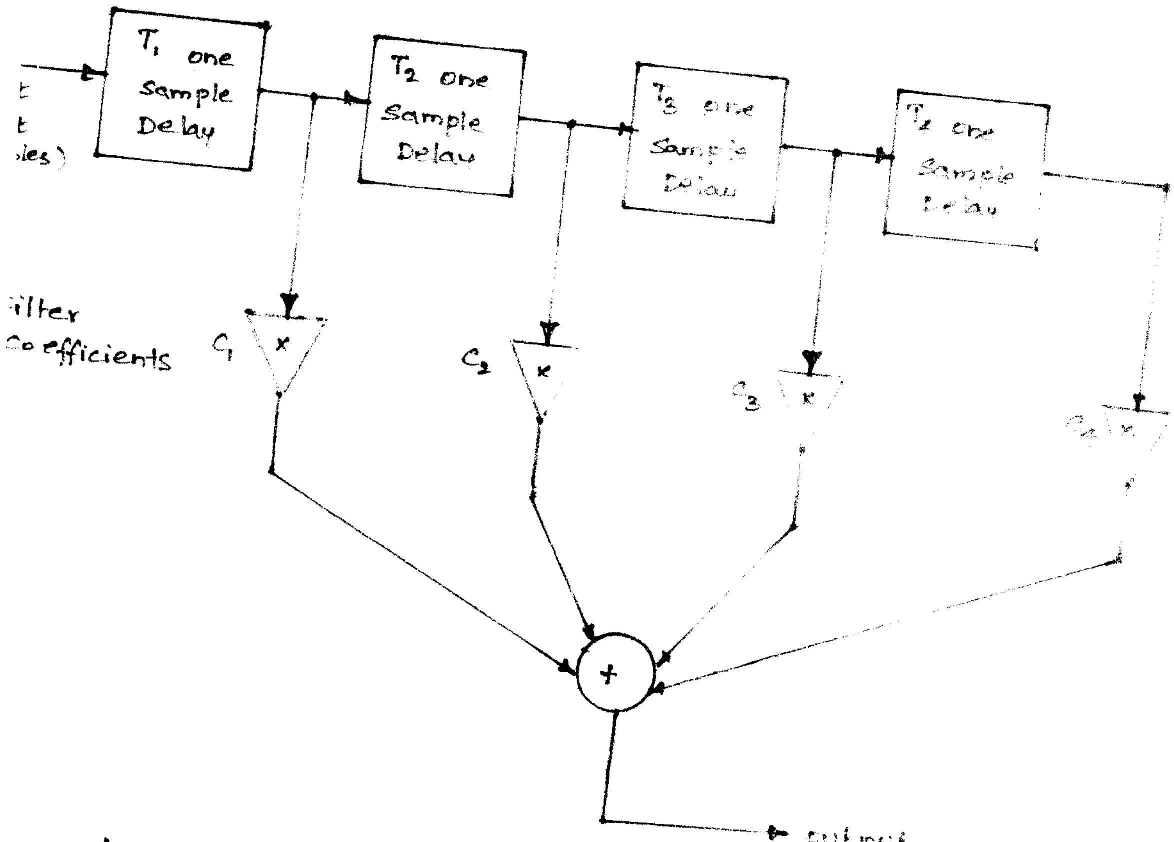


(a) A Delayed signal added to



(b)

Results in a filtered frequency response



A transversal filter uses a shift register, coefficient multipliers, and an adder to produce a filtered output

Time Delay

Rather than using a brickwall filter to suppress the high frequency image components after the signal has been converted to analog form. it is possible to process the digitized signal before D/A conversion using a digital filter. The basic mechanism used in the digital filters found in CD players is time delay. A delayed signal added back to itself results in a wholly new frequency response called a comb filter response. If a signal is delayed by time T, as shown in figure and mixed with the original undelayed signal, the two will be out of phase, causing cancellations at odd multiples of the frequency, that is 1/(2T). For example, a 1 millisecond delay will produce notches at 500 Hz 1500 Hz 2500 Hz etc.,

In the same way, digital filters use summed delays to produce phase cancellation in the signal and consequently, a filtered frequency responses.

Although the notion of filtering use summed delay line might seem unusual the phenomenon is common in the acoustical world. For example a sound traveling to a microphone can take a direct path or a reflected (delayed) path lengths creates a comb filter response.

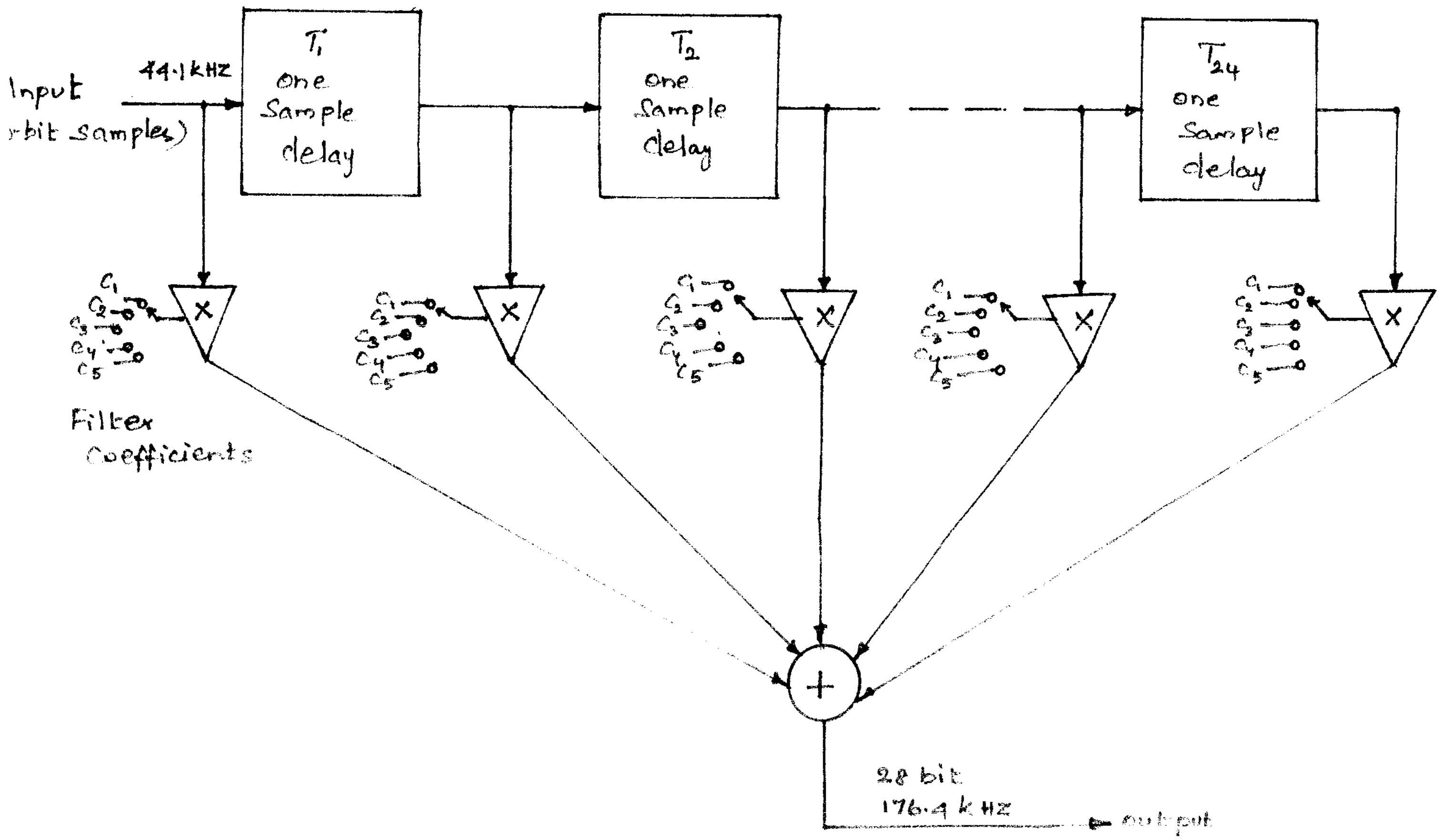
In a digital filter delay lines composed of shift registers are used to create time differences in the signal. Furthermore multiple delays are taken from the line. For example delays of 3,4, and 5 microseconds will yield notches at odd multiples of 166, 125 and 100 KHz. In addition, the notches will overlap.

attenuating all frequencies in that band. In the case of a CD player's digital filter, the delays are greater in number and specifically calculated to attenuate image spectra above the audio band.

TRANSVERSAL FILTERS

Remembering that each sample point of the filtered output signals is the sum of many samples. We can demonstrate how a digital filter works. We may employ a shift register with output taps after each delay element, as shown in fig., . Input samples enter the shift register. The output of each tap is multiplied by a coefficient associated with the impulse response and the product is summed with other products to yield the new output sample. The coefficients are taken from the sin(x)/x curve and thus impose that characteristic on the audio data when it is multiplied low-pass filtered data. To access many such samples, each time a new sample is input, the samples in the shift register are shifted one delay to the right and the new sum of products is recalculated. Because of the movement of the samples across the shift register, this design is often called a transversal filter.

A step-by-step example may illustrate the filter's operation. First (referring again to fig) assume that data has already shifted through the filter. We observe that the output sample is equal to the T_1 input sample times the C_1 coefficient plus the T_2 input sample time the C_2 coefficient plus the T_3 input sample times the C_3 coefficient plus the T_4 input sample times the C_4 coefficient. When the next new sample is entered the previous

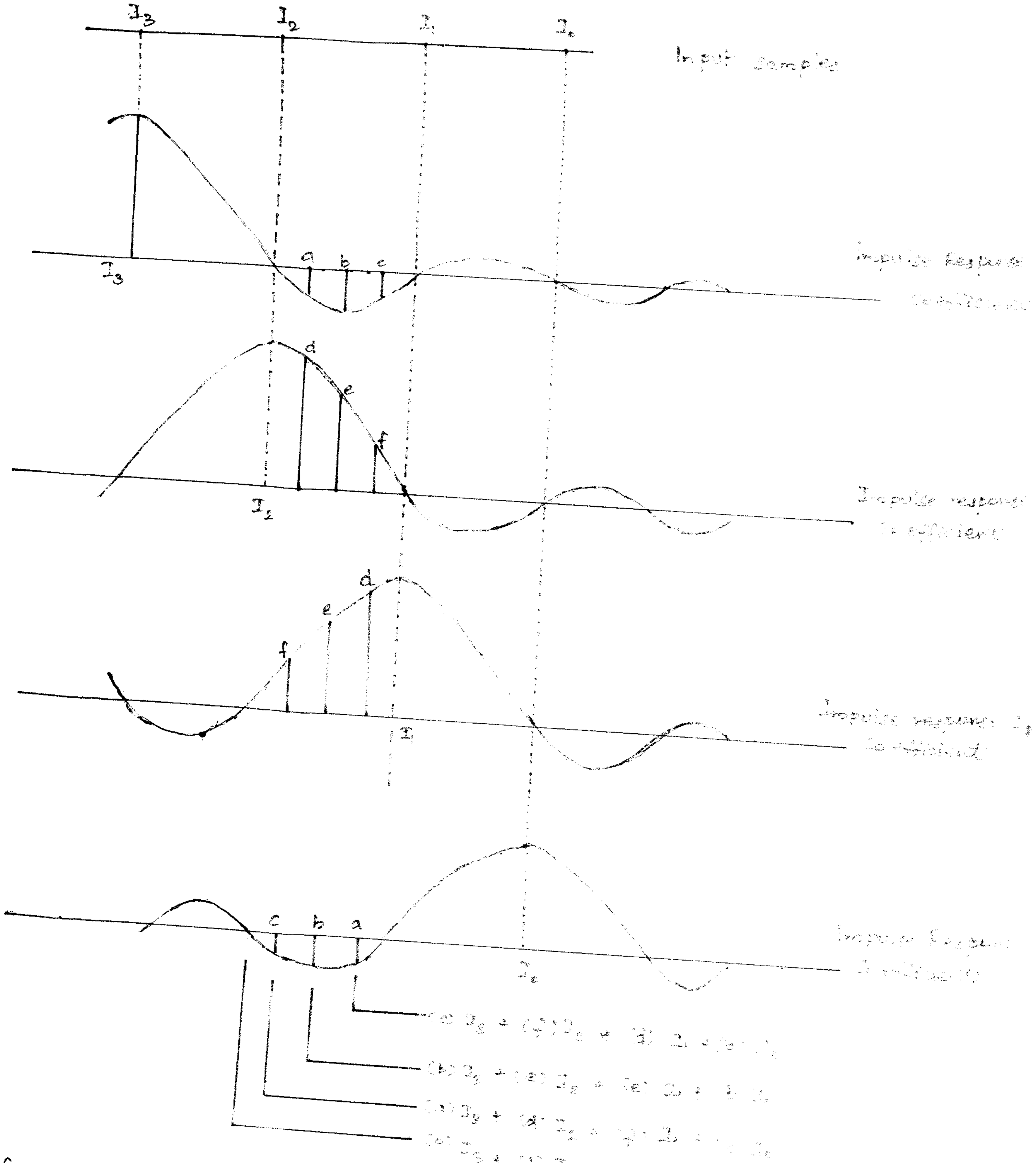


A Practical four times over sampling transversal filter may use twentyfour delay elements and four set of coefficients.

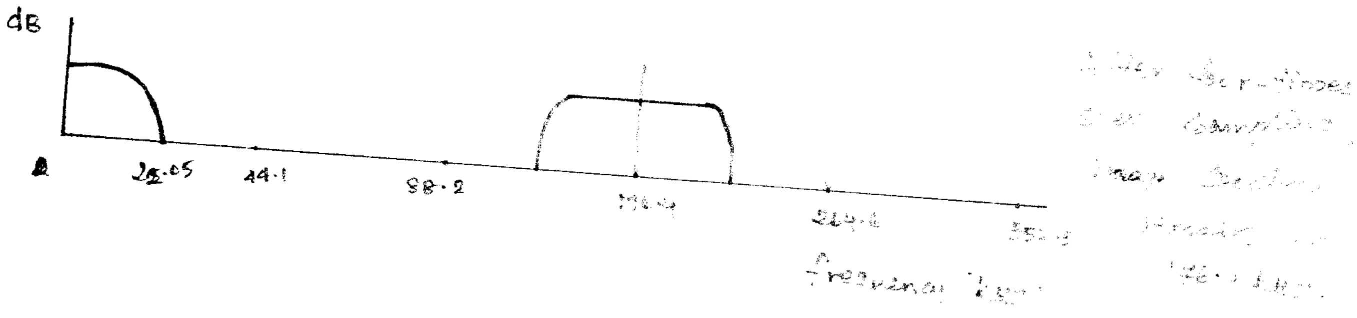
A step-by-step example may illustrate the filter's operation. First (referring again to fig) assume that data has already shifted through the filter. We observe that the output sample is equal to the T_1 input sample times the C_1 coefficient plus the T_2 input sample time the C_2 coefficient plus the T_3 input sample times the C_3 coefficient plus the T_4 input sample times the C_4 coefficient. When the next new sample is entered the previous

samples are shifted one delay to the right to make room for it and the calculation is repeated. Thus in this example each sample shifted through the filter requires four muiltiplications and four additions.

In practice a number of considerations determine the design of implemented filters. The analog signal is the sum of the sin(x)/x waveforms resulting from each sample. The sin(x)/x waveform extends to infinity in both the positive and negative directions, so theoretically all of the values of that infinite waveform would be required to reconstruct the analog signal. (In our example, above, only four coefficients were used). Fortunately we can find a point on the waveform where neglecting any further response results in an error less than the system error. We can disregard the rest of the response values. Such a design



A four times over sampling filter treats input samples as [illegible] sin(x)/x curve, thus synthesizing [illegible]



is called a finite impulse response (FIR) filter.

As noted the number of samples must be increased through oversampling. To achieve oversampling a transversal filter could for example be constructed so that each delay is now one_ fourth sample period. This would result in four times the number of output samples. With the filter calculating three new samples values for each input sample. To provide enough impulse response values to permit a good response after summation. Perhaps ninety six delay elements would be required. It would input new data four times each sample period. Every fourth sample would be an audio sample from the disc, and the other three would be zero intermediate samples. Thus only twenty four of the ninety_six elements would be filled with non_zero samples at any time. The filter would output data at a rate of 176.4 KHz each new sample the sum of twenty-four non-zero multiplications. The filter thus calculates three new sample values at the location of the zero input samples. However this oversampling design is inefficient the same result can be obtained with fewer delay elements.

A more cost-effective approach is shown in fig . It shows the architecture of a practical four-times oversampling digital filter generating three intermediate samples between each input samples each delaying a 16 bit samples for one sampling period. Thus each sample remains in each element for a sample period. During this time each 16-bit sample is multiplied for four times by a 12 bit coefficient stored in ROM, a different coefficient for each multiplications. In total, the four sets of coefficients are applied to the samples in turn, thus producing four output values. The twenty-four multiplication products are summed four times during each period and are output from the filter. The data is shifted one place, and the process is repeated. After multi-plication and summation four times as many samples are present after oversampling with new intermediate values calculated by the filter. The sampling frequency is increased four times, to 176.4 KHz.

Readers still uncertain as to the working of digital filters or suspicious of their ability to simulate the effect of analog filters in terms of waveform reconstruction are directed to fig . a graphical summary of oversampling. It shows how intermediate samples are computed in a four times oversampling filter. The input samples I_3, I_2, I and I are treated as sin(x)/x impulses placed relative to the center of the filter. Their sin(x)/x impules response amplitudes are equal to the original sample amplitudes and the width if the impulse responses are determined by the response of the filter. In this case a filter with a cutoff frequency at the Nyquist frequency of the input samples. The summation of their unique contributions forms the interpolated samples (in practice, as noted many more than four samples would be present in the filter). Each of the three interpolated samples is formed by adding together the four products as shown in the fig. Original disc samples pass through the filter unchanged by using one set of filter coefficients that contains three zero coefficients and one unity coefficient. In this way the output sample that coincides with the input sample is unchanged. By multiplying each group of samples (four in this case) with this coefficient set and three others interpolated samples are output at a four-times rate.

The filter's coefficients produce a transition region between 20 and 24.3 KHz and again around 154 KHz. Fig., shows the effect of the filtering. The values of the intermediate samples, obtained by the calculation process determine the filter characteristic; the bands centered at 44.1, 88.2 and 132.3 KHz have been largely suppressed.

Following the multiplication occurring during oversampling the word length is much longer than the original 16 bits. Oversampling reduces the audio-band quantities noise by distributing it over a wider bandwidth. Four times the bandwidth in the case of four-times oversampling. However the output word cannot simply be truncated to 16 bits this could increase distortion. Sometimes employed is a special rounding-off mechanism known as noise shaping. Quantization noise and the distortion from truncating are distributed over the entire oversampling spectrum. Thus the distortion in the audio band is a fraction of the total. With noise shaping the dynamic range of the system can be extended. Noise shaping is discussed in more detail later in this chapter.

detail later in this chapter.

DIGITAL-TO-ANALOG CONVERTERS

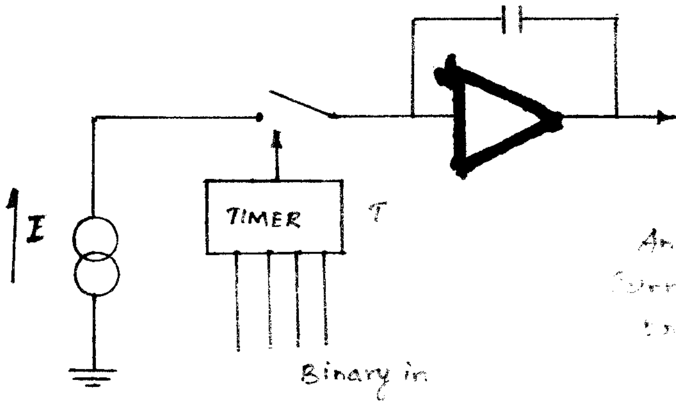
All compact disc players have digital-to-analog converters, otherwise known as D/A or DACs. Increasingly, however CD players optionally output their digital information directly to preamplifiers and power amplifires with their own D/A converters. Ultimately all - digital stereo systems will appear. The analog signal that our ears requite will not appear until the last possible moment, at the loudspeaker. Until then players contain D/As to convert the digital bit stream to an analog signal, so that analog preamplifiers and power amplifiers can process it.

A D/A must produce an analog voltage corresponding to each digital audio sample. The sum total of these sample voltages creates an analog waveform-actually one waveform for each stereo channel. The signal output from the D/A (following the output sample - and - hold) circuit in the recording side of the chain. There are several excellent ways to accomplish this task.

Dynamic Element Matching D/A Converter

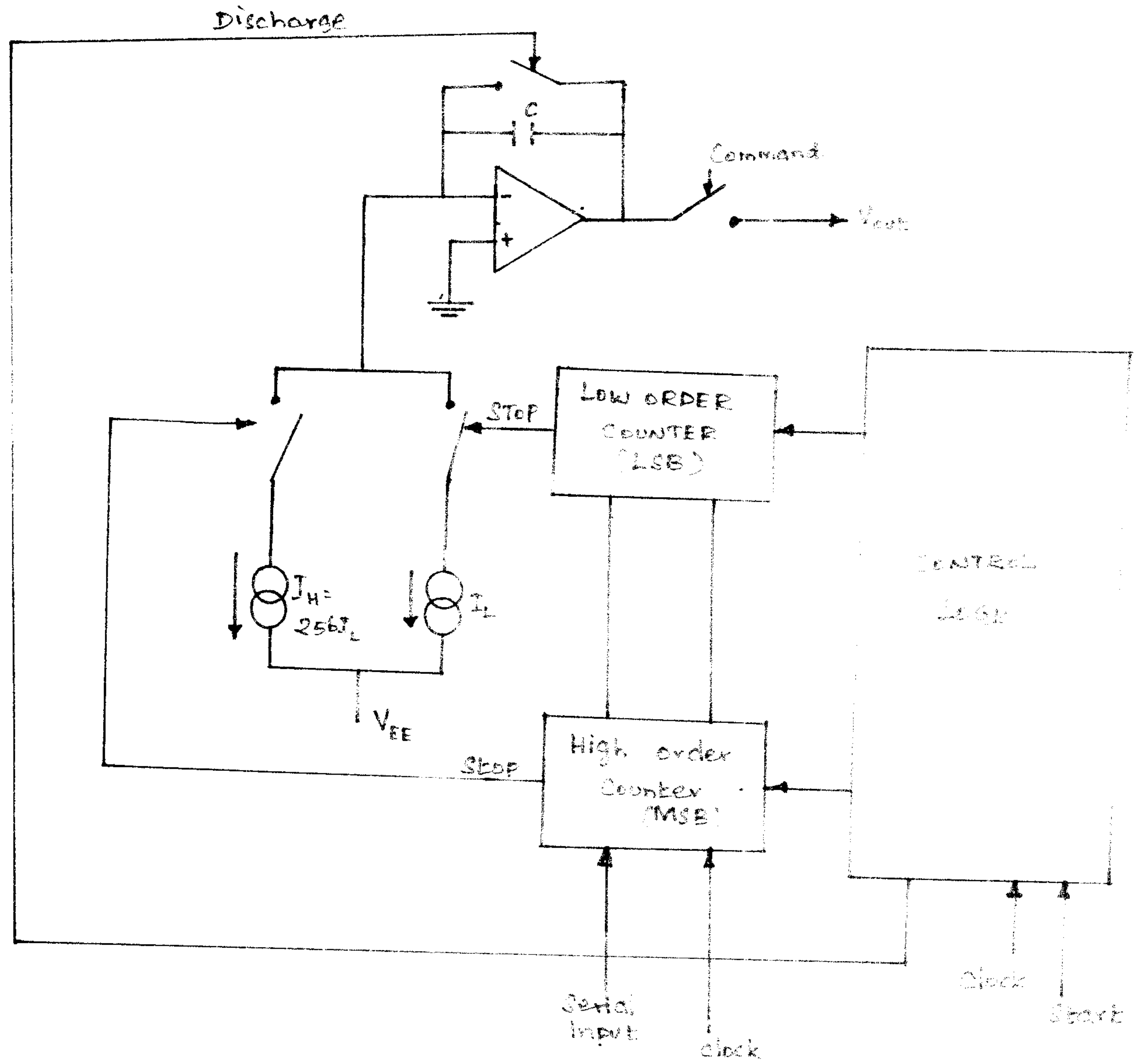
The D/A design, called dynamic element matching (DEM) uses the principle of current adding. This is illustrated in fig., and accurate current source is obtained by dividing a reference current source with a pair of resistor, then by averaging their outputs with a pair of switches and capacitors. By cascading these or other types of current-dividing stages, a binary weighed series of currents can be obtained. Their selected summation accomplishes the task of conversion. Although a DEM converter may require more external parts than other designs it is cost-efficient because it requires no calibration.

Whichever method is employed, the D/A converters used in most CD players do not pose insurmountable design problems. Contemporary players utilize one or two converters. Typically converting a 16 bit word. In some designs 18 or 20 bit D/A converters are employed the longer word lengths present following oversampling permits this technique, which yields an increase in the system's S/N ratio.



$V_{out} = \int_0^T I\,dt$

An Integrating D/A converter [illegible] current source integrator, [illegible] [illegible] [illegible]



An dual slope integrator [illegible] [illegible] and uses two current sources to produce [illegible] voltage.

D/A Converter Resolution and Linearity

The number of bits that a converter process is not a good means of determine quality. In reality, the accuracy of the converted bits is far more important. For the most part, 16 bit D/A converters are being superseded by converters that provide 18 or even 20 bits of resolution. In fact, such resolution is considered to be de rigueur in even medium-quality players.

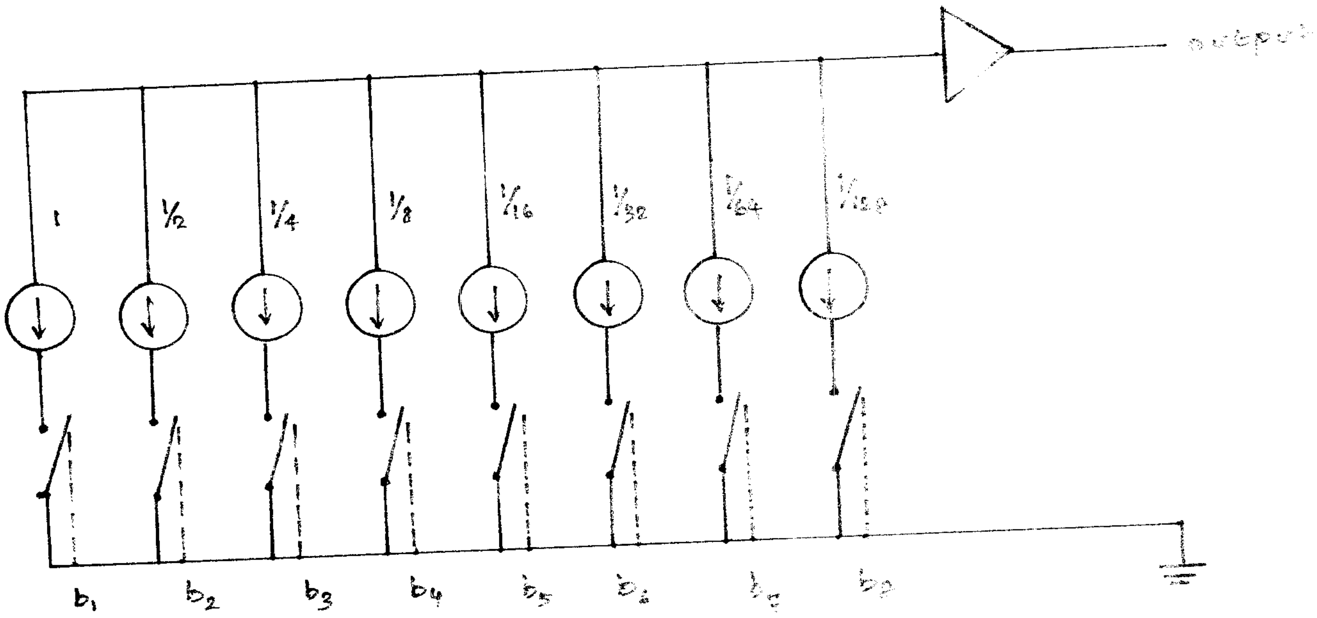
The choice of storing 16-bit words as the CD standard was determined primarily by the availability of linear 16 bit D/A converters. However new conversion methods offer a change to improve conversion quality. While no D/A technology can yield more than 16 bits of fidelity from a 16-bit medium the use of new conversion methods can provide more linear conversion of those 16 bits. When correctly done alternative conversion of the medium's 16 bit signal. Looked at in another way it may be said that in practice linear 16-bit D/A conversion is insufficient for 16 bit strong or processing. The reason for this lies in flaws inherent in D/A converters. Except in theory 16 bit converters cannot fully decode a 16 bit signal without a degree of error.

A D/A linearity test measures a converter's ability to reproduce analog signals particularly low-amplitude signals. Ideal for example at -80 dB tone on a disc would be reproduce at -80 dB. However a nonlinearity of +2dB at this level would result in a tone reproduced at -78 dB. A deviation such as this manifests itself as distortion in the audio signal.

Converter non-linearity can be traced to a number of causes, but the most critical factor is the accuracy of the converter's most significant bit (MSP). It accounts for a change in fully one-half of the analog signal's amplitude while in comparison the least significant bit (LSB) in a 16 bit converter accounts for an amplitude of one part in 65,536. Many D/A converters thus provide a means to calibrate the MSB and in some cases several of the most significant bits can be calibrated. This permits the CD player manufacture to compensate for differences in chips. Because these most significant bits steer the converter's output proper calibration of individual must be done with great care. For example when trimming the MSB of a 16 bit converter an error of 0.01 percent would negate the contribution of the two least significant bits. An error of 0.1 percent in the MSB would offset the combined value of the five least significant bits.

Unfortunately many audio critics have found cases of calibration trimpots consistently set in the middle of their adjustment are accurately performed calibration is often lost over time as the player is exposed to changes in temperature, humidity, and simple aging. Given the accuracy needed in the calibration of the MSBs this is not surprising.

One solution to the limitations of 16 bit conversion is a D/A converter with more of conversion. An 18 bit D/A converter has 262.144 levels and as a 16 bit converter. Any nonlinearity would be correspondingly smaller, and increasing the quantities word length at the conversion stage results in an increase in S/N



MSB LSB

Fig (a)

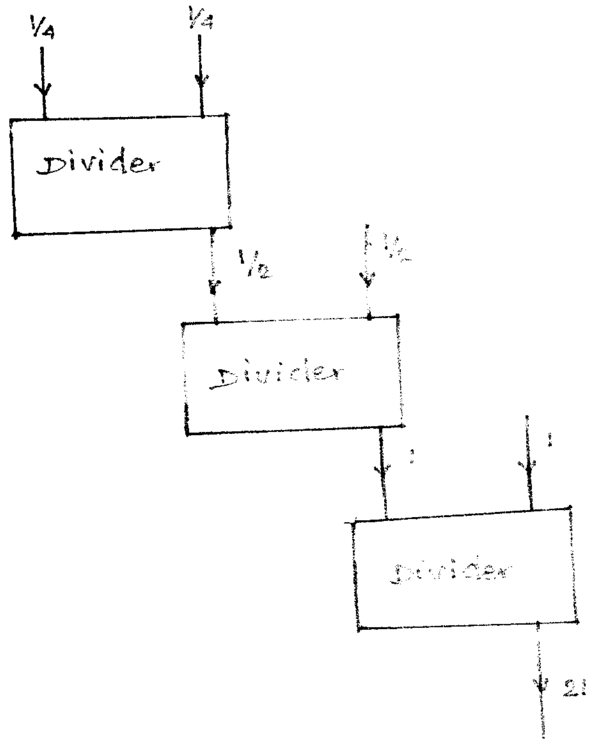


Fig (b)

A dynamic element matching D/A converter uses current adding (a)

The current sources ~~and~~ obtained from a current source and current dividers (b) ratic. Simultaneously any quantization artifacts are diminished. In other words, a higher bit converter would give a better 16 bit conversion. In fact the extra bits of a linear high-bit converter would not even have to be connected to yield improved 16 bit performance.

The intent of 18 or 20 bit D/A conversion technology is similar to that of oversampling. While the sampling rate is increased the method doesn't create new information. It merely attempts to make better use of existing information. not coincidentally it is oversampling that makes high-bit conversion of 16-bit data possible. It solves the obvious dilemma of coming up with the extra bits. When the output from the medium is only 16 bits. When a 44.1 KHz 16 bit signal is oversampled [illegible] sampling frequency and number of bits are increased, the former because of oversampling and the latter because of the multiplication which must take place. For example, the output of an oversampling filter may be 176.4 KHz and 28 bits. Normally only the 16 most significant bits are used for conversion through a 16-bit D/A converter and the rest are discarded or used for noise shaping.

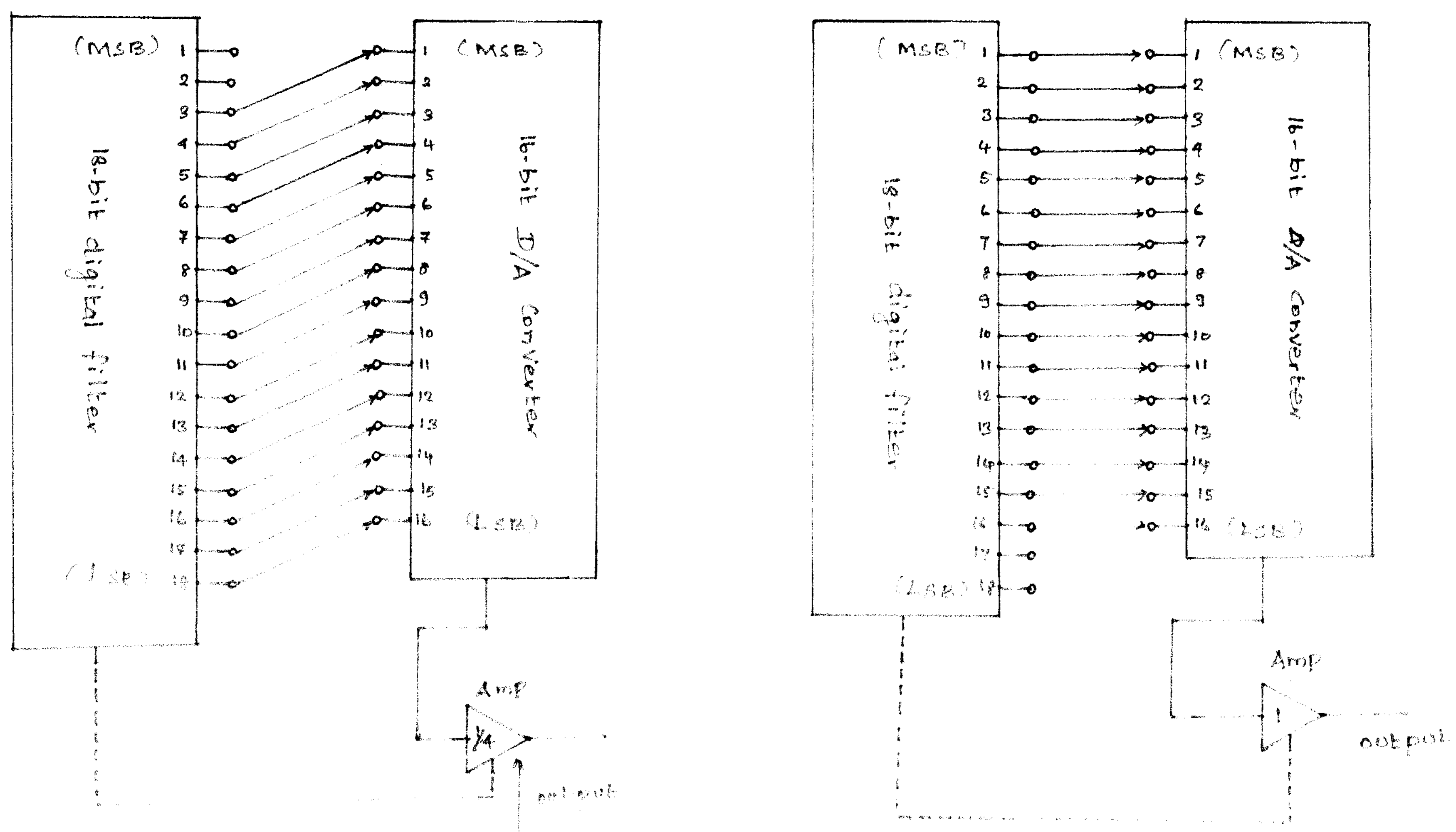
The use of extra bits for output conversion requires either a longer word length D/A conversion of a switched 16 bit converter. In the latter design, the parallel bits from the oversampling filter's output are wired through switches to the inputs of a 16 bit D/A converter. An example of an 18 bit shifted bit D/A converter system is shown in fig. . When all 16 bits are used to convey a signal (a high-amplitude signal) the upper 16 bits are applied to the 16 bit converter. However when the two upper bits from the oversampling filter are not being used to convey a signal (a lower amplitude signal) the 18 bits are shifted down-ward, the unused bits are ignored and the 16 lower bits are utilized instead. This adaptive scheme makes sense because in music recording the 2 upper bits are rarely used, and then often for a brief period of time. Through bit shifting a 16 bit con-verter may thus handle an 18 bit input.

However because we have shifted the word by 2 bits the signal's amplitude changes. in the binary system, a shift of one place (one bit) results in a doubling of value. Thus when the lower bits are shifted up, the amplitude is four times too large thus the output must be attenuated by one-quarter. An analog gain block downstream of the D/A handles this chore.

This adaptive 18 bit conversion scheme may be considered a noise reduction scheme because the signal is expanded at the D/A converter. The benefits result from the fact that the residual noise of the converter as well as its conversion non-linearities will be proportionally reduced. Looked at another a four-times (12 dB) higher analog output has been achieved without increasing the D/A's residual noise and conversion error. When the gain is reduced by one-quarter to bring things back to normal, the noise and conversion errors are reduced by one -quarter. There is an increase in S/N of 12 dB and distortion reduction by one-quarter both equivalent to using an 18 bit converter.

Of course as with any clever scheme there is a pr[illegible] paid for the benefit accrued. Specifically, when the bits are shifted it is difficult to immediately and simultaneously shift the gain of the analog output to compensate. Furthermore, any static offset introduced by component tolerances in the attenuator could become apparent when the switching takes place. It would not be significant over the first 14 bits affecting only the attenuation ratio. But when switched off (or later on) the difference between the attenuator and the absolute value of the output could create a glitch in the waveform where the attenuated and nonattendance signals are joined. The glitch would be present at -12 dB regardless of signal frequency. In practice, the increase in S/N ratio far outweighs the disadvantages of [illegible] nonlinearity.

Recent work has resulted in linear 18 bit D/A converters which are both very affordable and highly sophisticated one D/A converter contains a test circuit to monitor settling time of the current source. This minimizes glitches to the extent that resultant distortion is below measurable levels and can be analyzed only through computer modeling. Although a conventional binary-weighed architecture is employed in this particular converter, several special design consideration were instituted to attain the desired accuracy. The three most significant bits are made up of unit-valued current sources which feed bits through R-2R



(a) Gain lowered by 1/4 (b)

Eighteen bit performance can be achieved with a 16 bit D/A converter by using two lower bits from the oversampling filter during low audio levels (a).
The bits are shifted when high levels occur (b).

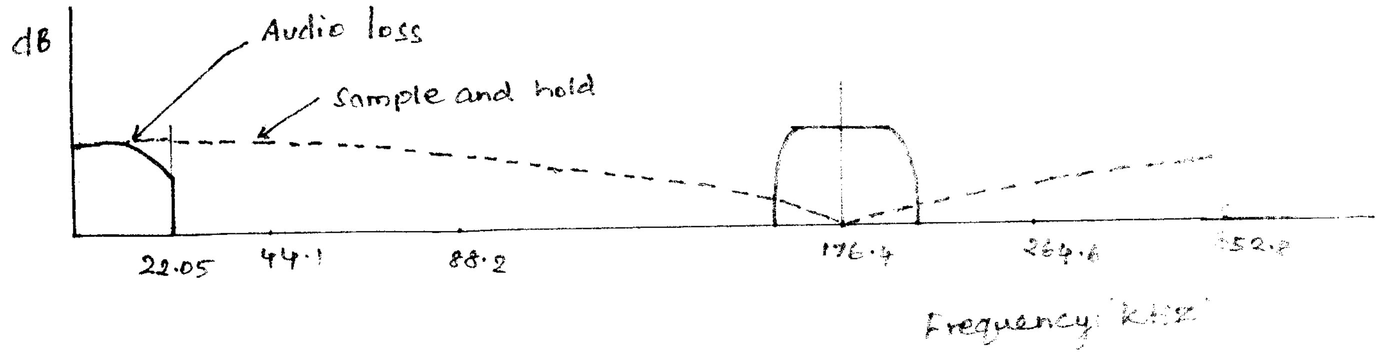
ladder network. Currents for bits 17 and 18 are derived from the unit-valued source. The relative gain of the upper bits can be adjusted against the total of the 15 lower R-2R bits by trimming the scale-down network.

The chip itself is a 40-pin hybrid with a special layout provide full control over laser trimming of the upper and lower bits. Supporting circuit is located outside the chip to help maintain the chip's thermal balance, a critical concern in D/A design. Accuracy to the least significant bit requires precision equivalent to four parts per million.

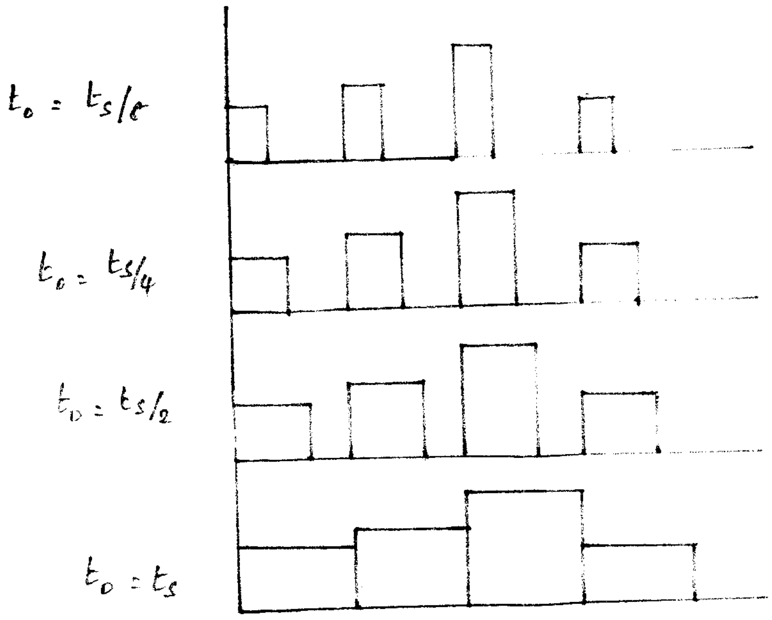
The architecture of a linear 18-bit D/A design is markedly simpler than a pseudo 18 bit method. The use of 18 bit D/A chip in player specification clearly superior to 16 bit designs. As described above an 18 bit converter has four times greater resolution than a 16 bit converter. In other words 18 bit conversion of a 16 bit signal provides for a 12 dB increase in S/N ratio while processing the data. This improvement is clearly measurable with 18 bit conversion. Of source there is no risk of gain matching errors as with pseudo 18 bit conversion methods.

OUTPUT SAMPLE-AND-HOLD

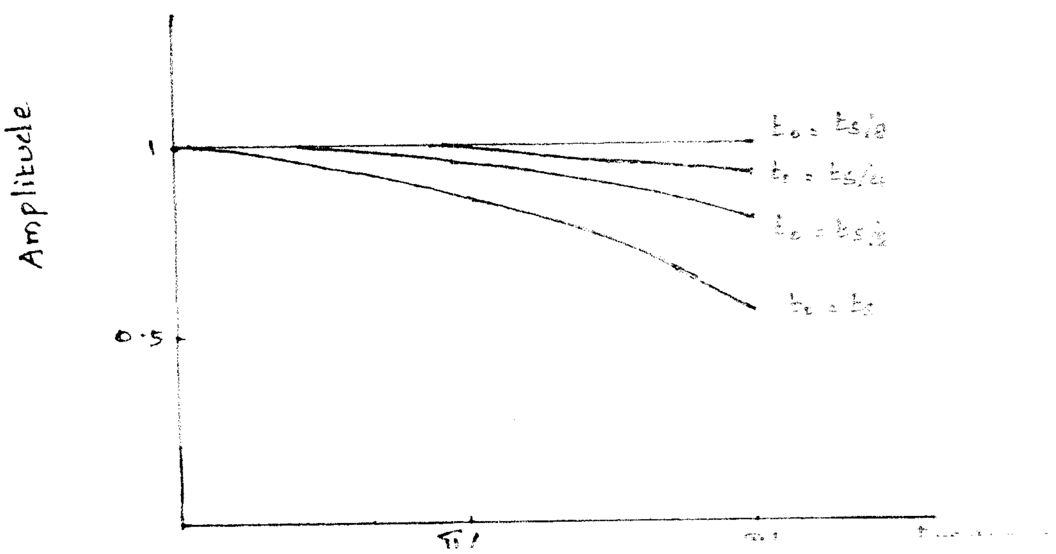
The sample-and-hold (S/H) circuit following the D/A which is sometimes contained on the D/A chip performs a hold function to create a pulse amplitude staircase. The impulses in the output staircase waveform are the width of a sampling period but reconstruction theoretically requires the impulses of infinitely short



Aperture error results in a slight attenuation of audio frequencies.



Shorter duration out samples minimize ape error resulting in a flat frequency respon



duration. This is impossible to achieve because it would require among other things, infinitely large current flow. Because of the finite duration of the output samples a filtering effect occurs in which the amplitude response (a sin(x)/x curve) gradually falls to zero at the sampling frequency, which can be seen in fig . This is beneficial because image spectra are attenuated. However the filtering results in a slight loss at the Nyquist frequency. This is sometimes called aperture error. To minimize this loss, the impulses can be narrowed closer to their original characteristic closer to the originally flat response for the pulse amplitude signal. Fig. [illegible] shows the sample pulse narrowed by four amounts. The aperture time A is defined by [illegible] t/t where t is the sample time and t is the duration of the pulse. Fig shows the resulting frequency roll-off due to aperture effects. The digital filter design can compensate for any slight remaining high-frequency attenuation via an inverse x/sin(x) characteristic.

In addition to its role in correcting aperture error, the sample-and-hole circuit is also useful in removing glitches from the staircase waveform. The D/A converter output is not a smooth transition from one analog level to the next. Instead inaccurate transition voltage known as glitches are often generated. [illegible] though they are memory and disappear when the D/A 's output value settles to its correct voltages these glitches would increase distortion in the output signal. A sample-and-hold circuit can [illegible] used to gate them out. This circuit is timed to wait for them to settle before it samples the D/A output. In this way only accurate values are propagated. Other types of de-glitcher circuits have also been devised to solve this problem. For example a shunting circuit could switch the output to ground during glitches transitions.

Following the S/H Circuit some image spectra energy will still remain. This must be suppressed by an output analog filter. This anti-imaging filter is tame compared to brickwall filters. Since the remaining band is so high in frequency, we can use a filter with a gentle. 12 dB/octave response and a -3 dB point between 30 and 40 KHz. it is a noncritical design, and its low order guarantees good phase linearity. Phase distortion can be reduced to $\pm$ 0.5 degrees across the audio band.

LOW-BIT D/A CONVERSION

The advent of multiple D/A conversion technology set a new standard for digital audio conversion. In many cases, the primary sonic limitation in CD reproduction using linear 18 bit D/a converters is the quality of the equipment used to make the master recording itself.

On the other hand although 18 bit converters can reduce nonlinearity error below levels mandated by 16 bit converters the technology is not perfect. For example the reference voltage supplied to an 18 bit converter must be highly accurate and stable the converter must be carefully calibrated and extreme care must be taken to prevent drifting. The difficulty can be accessed by recalling that a 16 bit converter must be calibrated to an accuracy of one part in 65.536 whereas an 18 bit converter demands one part in 262,544. In other words, although it is more precise than a 16 bit converter an 18 bit converter has all the real-world problems of a 16 bit converter, only more so.

One response to the limitations inherent in multiple converters lies in a clever conversion method which follows the philosophy that sometimes less is more. Instead of converting parallel conversion an analog process by performing conversion in the digital domain using DSP. Today many CD players use this third generation converter technology.

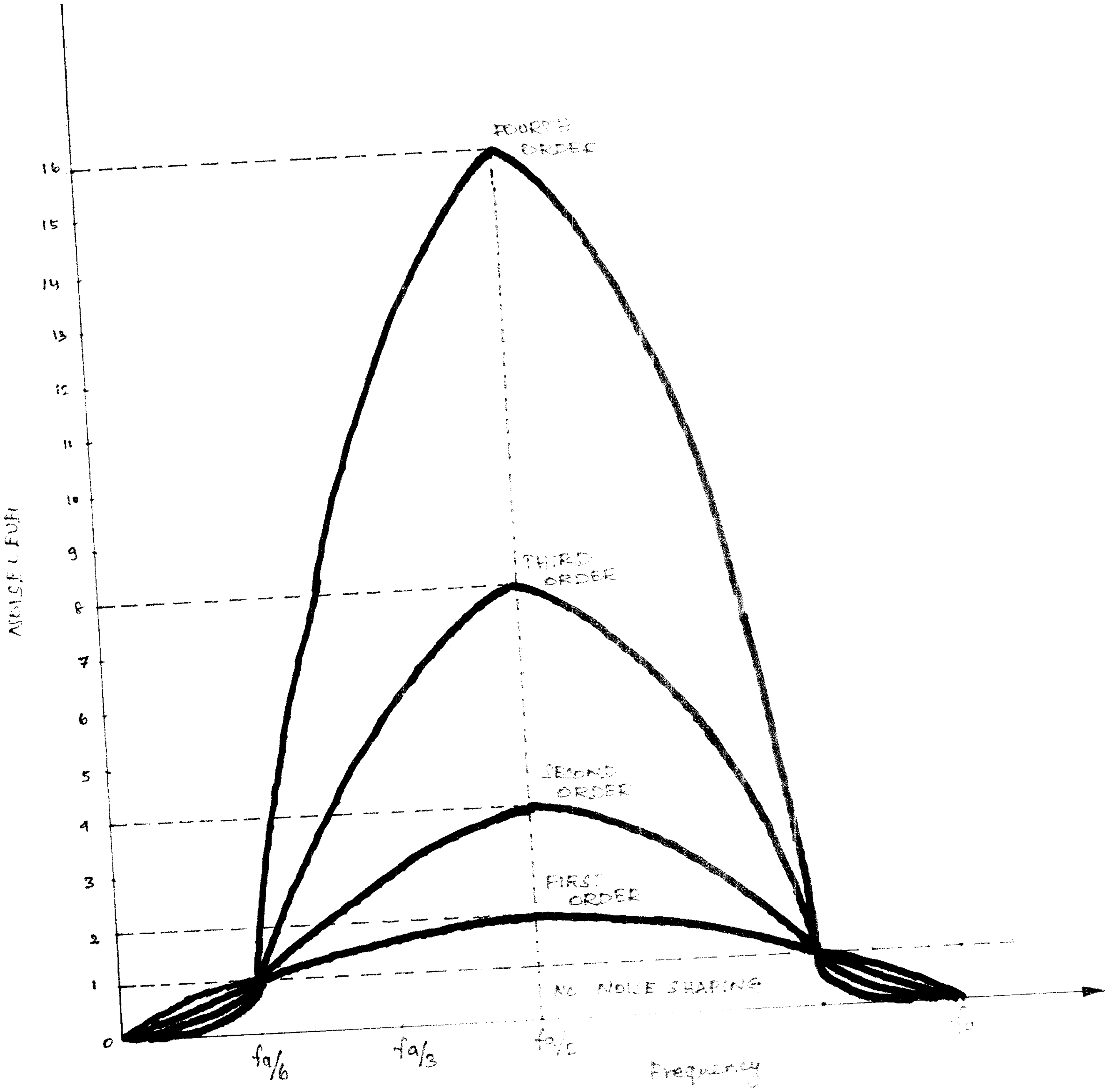
Fundamentally a multiple converter represents the analog waveform as an amplitude signal, processing information that measures the amplitude sample by sample. however, the method is flawed when quantization introduces errors in amplitude representation. Moreover because a multiplicity of bits are used to form the representation and because each bit has an error unequal to the others the overall error varies with each sample, making compensation almost impossible.

By contrast a low-bit system takes a radically approach. It employs only a small number of bits (perhaps 1 bit) at a very high rate to represent the signal, instead of an amplitude representation a low-bit system uses time division to represent the signal. When this representation is low-pass filtered , the result is an analog waveform just as in the multibit method. because only a few bits are used, any amplitude error is essentially constant and hence there is no relative error. In theory neglecting jitter an error would produce only an offset and thus could easily be removed from the signal. Because time division is inherently more accurate than an multibit converters.

The way in which a 1-bit (high-or low-level) pulse signal can accurately reconstruct an audio signal is not intuitively obvious. A simple analogy helps illustrate the method. If a light switch is turned on and off quite rapidly any intermediate light intensity can be created. Similarly, a 1-bit D/A converter can reproduce an audio waveform. All 1s would be a full positive signal all 0s would be a zero-level signal, and other variations would create intermediate levels. Fig shows how a signal bit with either a high or low level, can represent in this example, a smooth sine wave. In practice as described below, noise shaping is required to reduce in-band noise, and in some designs several bits are used instead of one bit.

NOISE SHAPING

A principle objective of low-bit conversion is reduction in the word length required to represent a signal. however the bit reduction required to form a low-bit signal from a 16 bit source greatly degrades the signal's dynamic range. For example to accurately convert a maximum-amplitude 16 bit word a 1 bit system would have to change polarity 216 times per conversion period with a sampling rate of 44.1 KHz this would demand a rate of



Depending on the order of the noise shaping [illegible]
In-band and out-of-band noise [illegible]
relatively [illegible]

approximately 2.9 GHz. This is too fast for current hardware limitations. 16 bit performance is lost and noise shaping is prerequisite in any low-bit system to reduce noise in the audible spectrum and hence improve dynamic range.

A low-bit converter may be viewed as a wide-band spectrally weighted noise source in which the noise is more benign than multibit converter nonlinearity error. As its name implies, noise shaping shifts noise away from the audio band thus lowering audio-band noise. Fig shows a 2 KHz test tone and the noise floor with low-bit conversion. Clearly with noise shaping noise is decreased in the audio band (up to 2 x 10 4 in the fig) at the expense of boosting noise beyond the audio band, because the noise beyond the audio band can be safely removed through filtering with noise shaping 16 bit performance (or much more) can be achieved with a single bit.

Noise shaping reduces quantization error by using the known characteristics of quantation error to modify the error and decrease its effect. Efficient noise-shaping algorithms use recursion to spread the noise power across a greater spectrum. In particular the method places the error information generated in the bit reduction process back into the signal much like negative feedback is used to reduce distortion in analog amplifies. In effect, noise shaping attempts to cancel error by subtracting it from the input signal. Feedback in this discretely sampled system can operate only if a delay element one sample or more in duration is placed in the feedback loop. By placing the quantization error signal in a feedback loop, the frequency response of the error signal is altered, shifting much of its energy outside of the audio band. Moreover the noise is shaped by the approximate inverse of the loop transfer function when a low-pass filter is placed in the loop, the noise spectrum rises with frequency. In other words the noise-shaping loop low-pass filters the signal and high-pass filters the noise. if the system is designed so the signal's frequency content is less than the filter's cutoff frequency the signal will not be affected. What's more oversampling performed prior to noise shaping spreads quantization noise over a wider bandwidth hence noise is further diminished in the audio band.

The real innovation associated with low-bit systems comes in the design of noise-shaping algorithms. Simply put the more complex the algorithm the lower the noise in the audio band. More specifically higher orders of integration in noise shaping decrease the in-band noise level. Fig shows four noise curves-for no noise shaping. Higher-order noise shapers can be designed with cascaded digital differentiations. The greater the number of cascaded stages, the greater the slope of the shaping function this in turn increases the suppression of audio-band noise. However third-or higher -order noise shapers may suffer operational instability due to overloading of the integrators. Thus in practice, a multi stage circuit is often employed as described below.

As the order is increased, the noise-shaping characteristic changes and in-band noise is decreased. However out-of-band noise is greatly increased; this could overly burden output analog low-pass filters. The dynamic range can be optimized through a combination of oversampling rate and order of the noise-shaping filter. As the oversampling rate is increased the portion of the curve in the audio band is relatively reduced, that is moved toward 0 Hz. In other words, although the shape of the noise curve remains the same, high oversampling rates relatively decrease in band-noise. A successful noise-shaping circuit thus seeks to balance a high oversampling rate with noise-shaping order to reduce in-band noise and shift it away form the audible range.

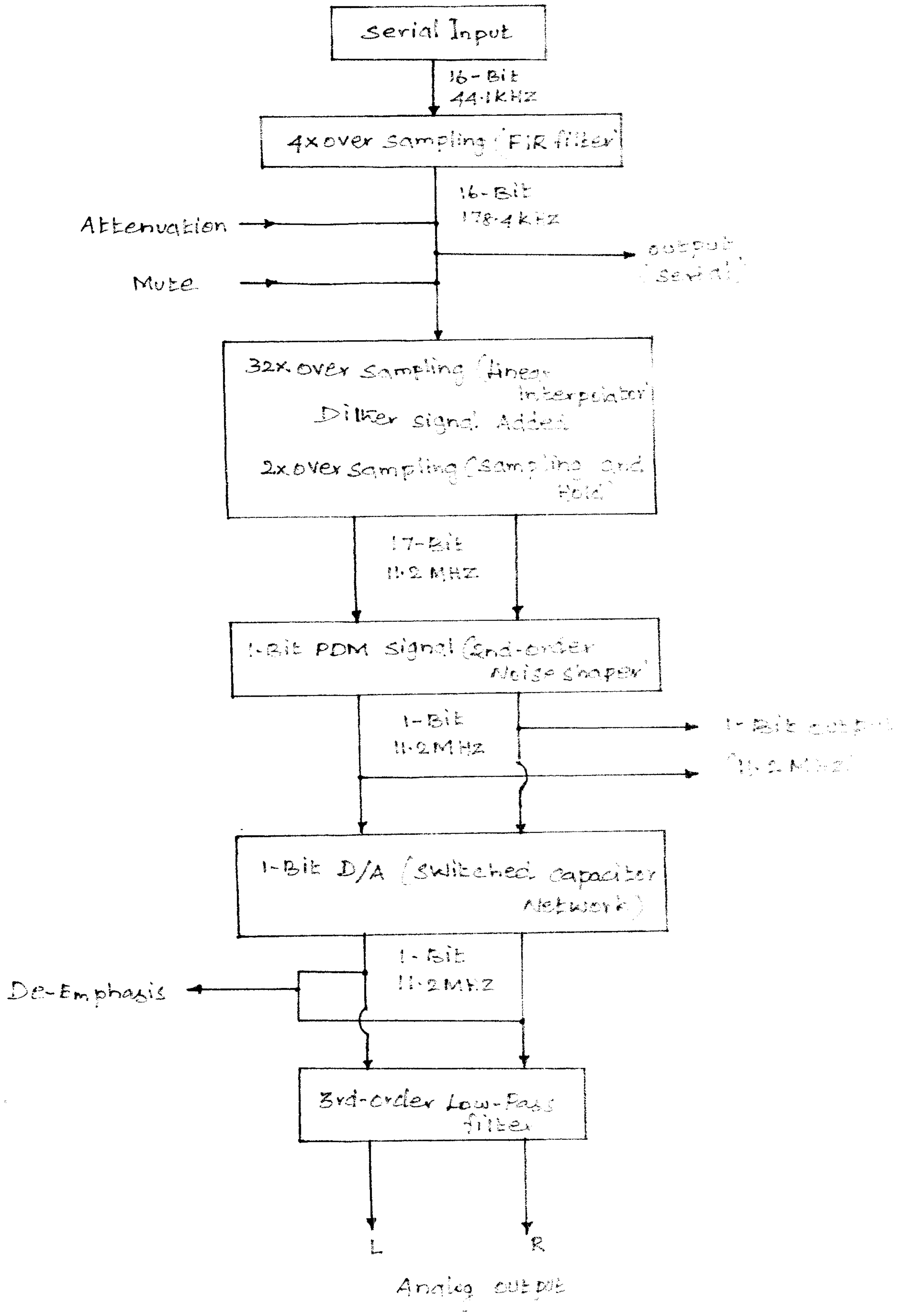
Noise-shaping processing can also result in the output of correlated patterns when the processor is idling with low-level audio signals. To overcome this, dither is added to the data so the circuit always operates with a changing signal even when the audio signal is zero or otherwise unchanging. A digital dither signal applied inside the shapping loop is identical to a high-pass filtered dither signal applied at a point outside the loop prior to the quantizer. In other words , the quantizer and the dither signal noise are both shaped by the loop.

Low-Bit D/A Conversion with Second-Order Noise Shaping

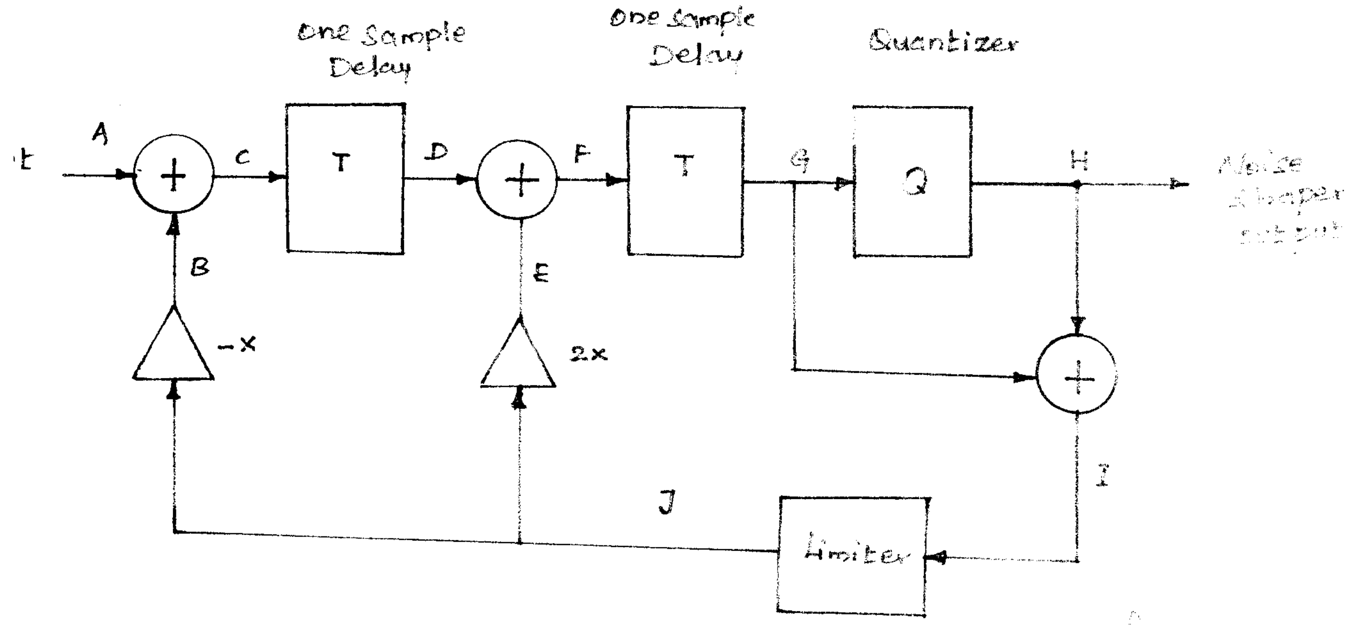
one implementation of a true 1-bit D/A conversion method is composed of three elements oversampling filters, second order noise shaping and pulse density modulation output. This Bit stream converter was developed by Philips. The sampling rate is increased from 44.1 KHz to 11.2896 MHz-an increase of 256 times. At the same time, the 16 bit signal is converted to a 1-bit signal. It is this fast, 1-bit output that reproduces the audio waveform. The re-quantization error of the output signal is corrected by second order noise shaping.

The output bit operating at a rate of 11.2896 MHz is converted to an analog signal using a simple switched capacitor network. The network's operation is very accurate and hence the error of the signal is low. There are only positive and negative full-scale reference points; intermediate points are determined by time averaging. Any errors in the two points would produce an offset error but not a linearity error. The offset error can easily be removed.

Fig shows the complete system. The first of the three oversampling stages uses a non-recursive FIR interpolation filter to perform four-times oversampling to attenuate image spectra; in addition first order noise shaping is performed in the filter. The signal is downscaled to prevent clipping during any signal overshooting. The gain of the filter is selected to compensate for the effects of other stages in the conversion path In particular, a high-frequency rise is used to compensate for the



BLOCK DIAGRAM OF THE SIGNAL PROCESSING IN A 1-BIT D/A CONVERTER EMPLOYING SECOND ORDER NOISE SHAPING.



	A	B	C	D	E	F	G	H	I	J
t_0	0·5	-0·4	0·1	0·5	0·8	1·3	-0·6	-1	0·7	[illegible]
t_1	0·5	-0·3	0·2	0·1	0·6	0·7	1·3	+1	[illegible]	[illegible]
t_2	0·5	0·3	0·8	0·2	-0·6	-0·4	0·7	+1	[illegible]	[illegible]
t_3	0·5	-0·6	-0·1	0·8	1·2	2·0	[illegible]	-1	0·6	[illegible]
t_4	0·5	-1·0	-0·5	-0·1	2·0	1·9	2·0	+1	[illegible]	[illegible]

An example of a Second-order noise-shaping [illegible] with numerical values in [illegible] processing [illegible] aperture error (-3 dB at 60 KHz) presents in the output analog filter. The second stage performs 32 times oversampling through linear interpolation. A dither signal (-20 dB at 352 KHz) is added to prevent idle patterns from causing nonlinearity. Two-times oversampling is performed in the third stage through a sample -and-hold operation. This 17 bit signal (dither adds 1 bit to the original 16 bit signal) undergoes second-order noise shaping as described above, and a single bit is output from the quantizer. Finally D/A conversion is accomplished at a 1-bit D/A converter via PDM (pulse density modulation) outputing 1-bit data at 256 times oversampling or 11.2896 MHz. A third-order low-pass filter removes out-of-band high-freaquency components.

The noise shaping performed in this D/A converter is complicated (it is an example of DSP at its best). but we can get a feel for its working through fig., . This circuit performs second-order noise shaping through a feedback loop and generates a low-bit signal for conversion into an analog waveform. In particular this noise shaper consists of two integration (filter) loops to reduce in-band quantization noise. The 1-bit code from the quantizer is the output sign bit; only one bit is output, with a logical error, after a limiting operation designed to prevent overflow. Second order noise shaping is performed by adding double the error and the negative value of the error to the previous two samples.

Operation of the circuit can be explained by following the values of audio samples during several sample periods at various points in the noise shaper. These values are shown in fig . In this example a maximum 0 dB value is represented by a 1 (in practice, to prevent overload, an input is scaled down in the digital filter). The successive inputs in this example (column A going top to bottom) are a constant 0.5 (-6 dB); for correct operation the output of the circuit should thus be a 1-bit output with an average value of 0.5. Arbitrarily initial values for the contents of the delay registers in fig are selected as D=0.5 and G=-0.6 output at H is 0.5 and the value at G is negative; thus the 1-bit output at H is -1. The output of the quantizer is +1 if its input is positive (MSB = 0) and -1 if its input is negative (MSB = 1).

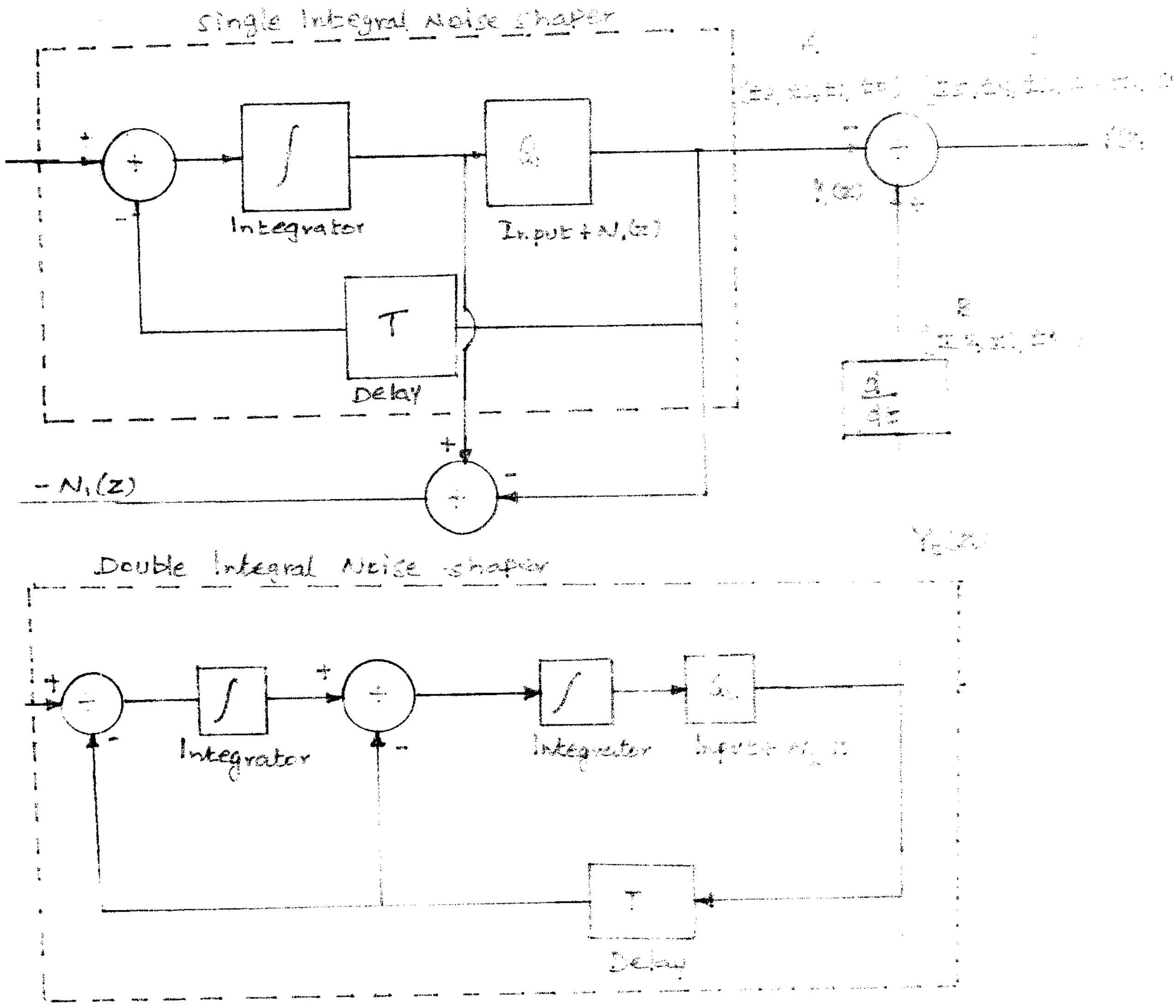
The quantization error signal is formed by taking G-H; this error is fed back into the double integration loops. The value of I is 0.4 and j=-0.6 + (-) - 1 = 0.4. The value of E is 2 x I = 0.8 and C = A-J= 0.1. The value of F=D + E =1.3. Similarly values pass through the circuit at each sample time; for example, D takes the previous value of C. It can be seen that the values inside the loop are larger than the unit value in other words , wider data buses are required. Also, if large values are input to the circuit, the limiter would be needed to prevent overloading of the loops. This example uses a constant input level but high-speed averaging applies equally to varying input music signals. In either case, the output signal from the noise shaper is applied to a 1-bit D/A converter. Which generates the appropriate analog voltage.

Two values are output at 512 times the sample rate. The signal conveys the audio waveform through the density of pulses; this pulse density modulation signal is converted in to an analog signal using a switched dual capacitor network. Specifically a capacitor is charged and discharged according to the "1" or "o" value of the data; the result is an analog waveform which reflects the encoded waveform through time averaging of the output bit.

This converter retains its accuracy over environmental extremes and a stereo converter can be contained on one chip. In addition this chip may be differential configuration.

Low-Bit D/A Conversion with Third-order Noise Shaping

Several manufactures have developed D/A conversion methods which employ third-order noise shaping. However their design details differ somewhat particularly in the noise shaping algorithms and the low-bit output signal. The MASH system is a multistage third - order noise shaping method. MASH was codeveloped by Matsushita Electric Industrial and Nippon Telegraph and Telephone Corporation. One implementation of this design accepts 16-bit words at a nominal sampling frequency and a digital filter



Q_1, Q_2 Local Quantizer

BLOCK DIAGRAM OF THE SIGNAL PROCESSING [illegible] A THIRD-ORDER MASH NOISE-SHAPING ALGORITHM stage performs 8-times oversampling and outputs 24-bit words, Noise - shaping circuits output data as an 11 value signal at a 32 times over sampling rate. D/A conversion is accomplished via PWM (pulse width modulation). Outputting low-bit data at a 768-times oversampling rate.

As with other low-bit systems, the key to the MASH system lies in noise shaping. As noted, if noise-shaping circuits are cascaded to achieve more than second-order noise shaping, they can be prone to oscillation; the MASH circuit achieves third-order noise shaping but avoids oscillation through its "multi-stage" (noncascaded) configuration. A simplified schematic for the MASH noise shaper is shown in fig ; it contains a first -order noise shaper in a main loop in parallel with a second a seccnd-order noise shaper. Input data is requalified into seven values (± 3, ± 2, ± 1 and 0) at the main loop. At the subloop the requantization error is requalified into five values (± 2, ± 1 and 0) used to correct the requantization error of the main loop. When the output values of the two stages are added together, the low-bit (4-bit) digital signal output from the circuit represents eleven vales (± 5, ± 4, ± 3, ± 2, ± 1 and 0). These values represents the audio signal.

The requantization error of the output signal is shaped with a third order differential characteristic (18 dB/octave). The final element in the system is analog conversion. The eleven-value signal is converted into pulses, each with a width corresponding to one value. This can be accomplished by applying the

4-bit output of the DSP circuit to a RCM to map eleven amplitude values into twenty-one time values with a constant amplitude, each representing a "start" and "stop" time of the pulse widths. The widest pulses translate into a large positive output, while the narrowest pulses translate into a large negative output as shown in figure The width of the pulses carries the vital information; the amplitude of this signal can only be high or low. At this point the signal has the form of 1-bit binary data.

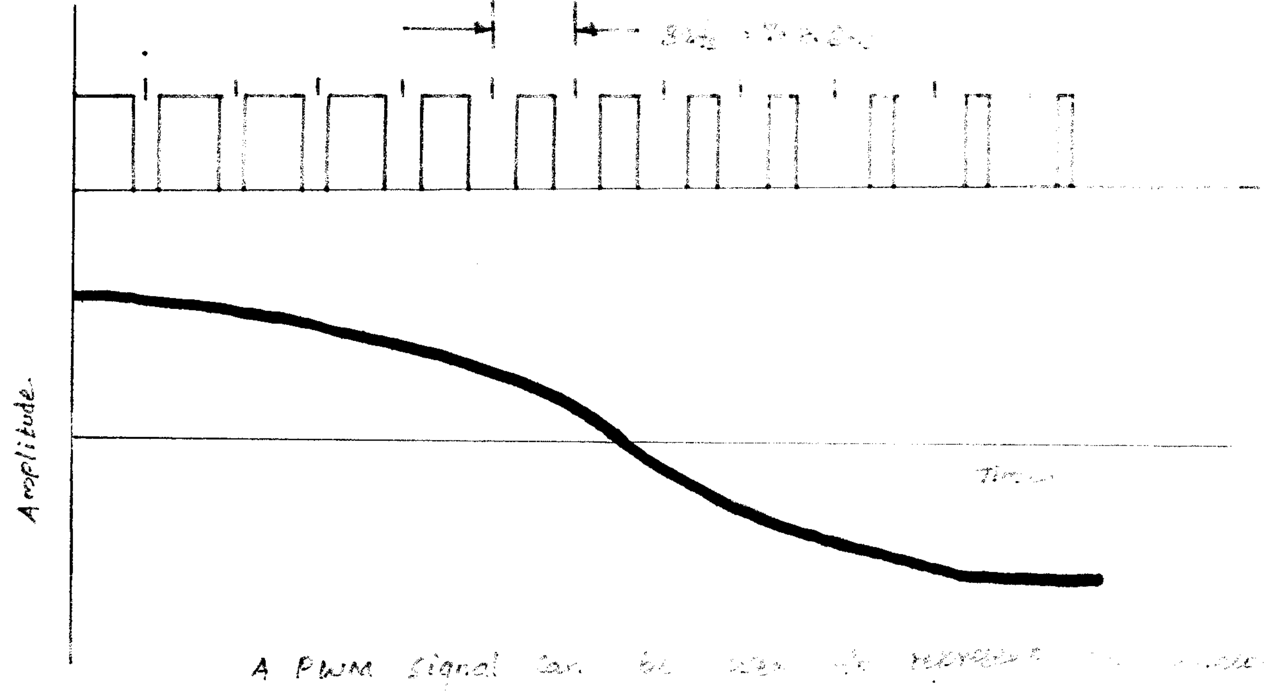
Because the signal is represented by a pulse width modulation waveform, final conversion is performed by simply passing the signal through a low-pass filter. Precise timing accuracy can be achieved through crystal oscillators; thus the widths are very accurate, and the error of the signal is low. Positive- and negative-going pulses may be output to cancel common noise. This 33.8688MHz(768 Xf_s) 1-bit data forms a PWM representation of the waveform.

Low-bit converter designs have already demonstrated the ability to provide outstanding performance. In terms of phase linearity, amplitude linearity, noise, long-term stability, and other parameters. oversampling D/A converters using low-bit architectures offer significant advantages over conventional multibit conversion methods. Moreover, the method is also applicable to A/D conversion. In short, the low-bit architecture is an exciting development in digital audio.

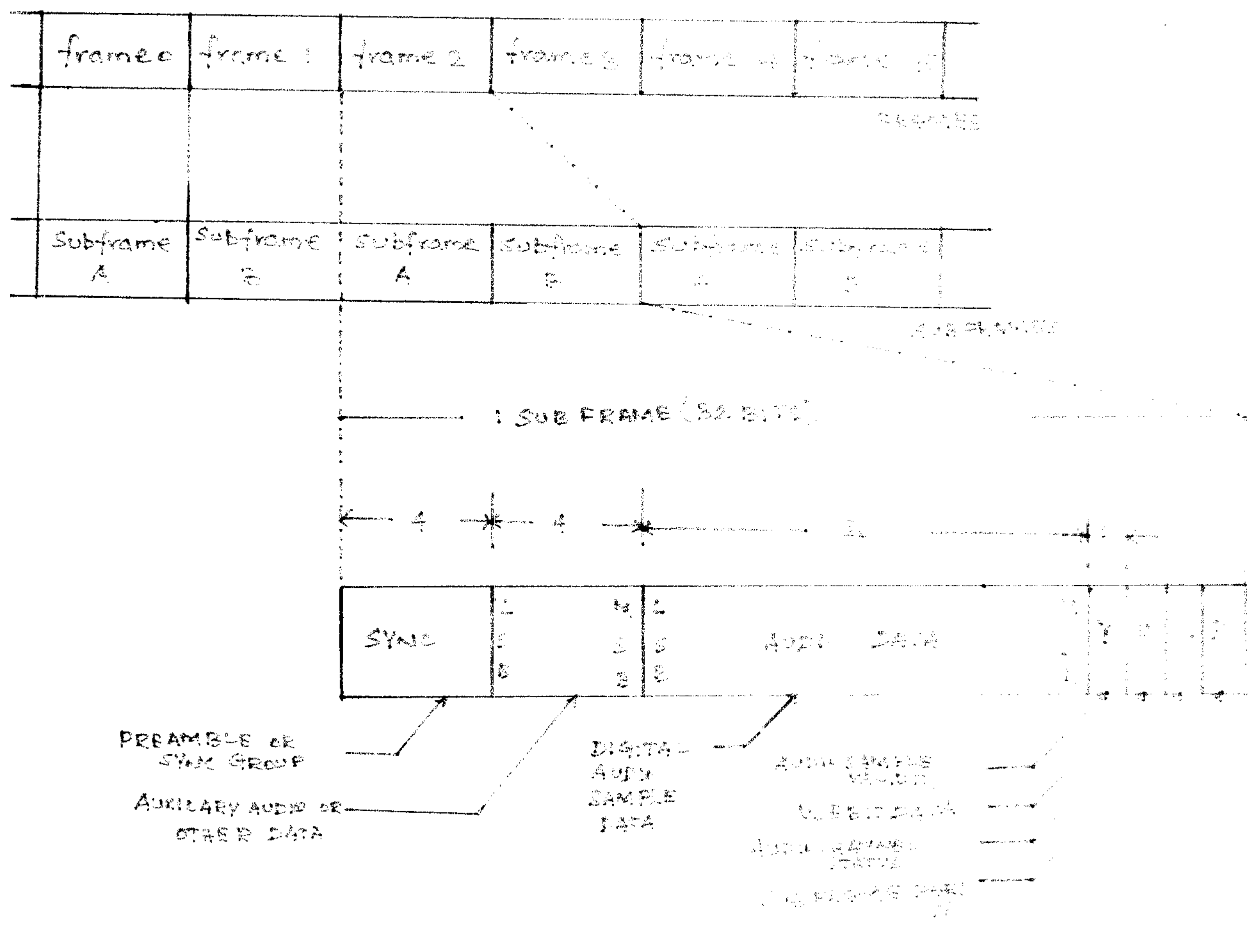
Each subframe carries data for one audio channel as shown in fig . The first 4 bits are used for synchronization and for identifying preambles, which will be described shortly. The next 24 bits carry audio data with the MSB transmitted last. Since most digital audio devices use 16 bit words the last 16 bits in the field are set aside for auxiliary audio or other data.

The last 4 bits form a control field which carries special information. The validity bit indicates if the previous audio sample is error free. The user data bit can be used to form a block of data associated with the audio channel, such as labeling of program number and duration. a high-level data link control (HDLC) communications protocol is specified for the user data channel. The channel status bit is used to form a data block; for each channel, one block is formed for the channel; status bit contained in 192 successive frames. The start of a block is identified with special forms of the subframe preamble. The parity bit is identified with special forms of the subframe: this permits simple error detection.

Three preambles are used to designate three types of events. One preamble marks the start of subframe A and the stat of a channel status data block. Otherwise, a second preamble marks the start of subframe A. A third preamble marks the start of subframe B.



A PWM signal can be used to represent ... in the output stage of a MASH D/A converter



SUBFRAME FORMAT ...

Each channel status data block consists of 192 bits (times 2) organized into two 24-byte sequences. Table 4.1 shows that these 8-bit byets convey a considerable amount of information pertaining to the transmission. The first byte specifies a consumer or profissional interface, use of pre-emphasis and sampling frequency. The second byte specifies channel mode (eg., stereo, or two-channel mono). The third byte carries word length information. The remaining channel status bytes specify information sample address, time of-day timecode, channel status validity and CRCC.

The data is modulated with biphase mark, in which a 0 corresponds to one inversion and a 1 corresponds to two inversion as in the [redacted]., . However the preambles ignore, the inversion correspondence to provide unique identification as shown in fig . The receiver input section uses this preamble portion for synchronization. The AES/EBU standard specifies that the digital waveform is transmitted over twisted pair conductors. With 3 pin XLR connectors (pin 1 shield or earth and pins 2 and 3 carrying signal).

‘CD’ Description

CD Description

This CD player signal processing and control circuit consists of the following main block.

Analog signal processor

Digital signal processor

Micro controller

Driver and

D/A converter.

HIGH-FREQUENCY SIGNAL PROCESSING :-

After the compensation of frequency response,if necessary,we can obtain the so-called eye diagram,shown in the figure.This is the result of processing by optical low-pass filter,expressed by MTF.

To convert into a two-level bit stream,it is necessary to take care of the "pit" distortion.By looking at the figure carefully,it can be understood that the center of the eye is not in the center of the amplitude.This is called Asymmetry,a kind of pit distortion.It cannot be avoided when disks are produced in large quantities because of changes resulting from variations in mastering and stamping parameters as well as differences in the players used for playback.According, a form of feedback digitizer,using the fact that the dc component of the EFM signal is zero,is recommended.In addition,the clock for timing signals is regenerated with a PLL circuit locked to the Channel-Bit Frequency (4.3218 MHz).

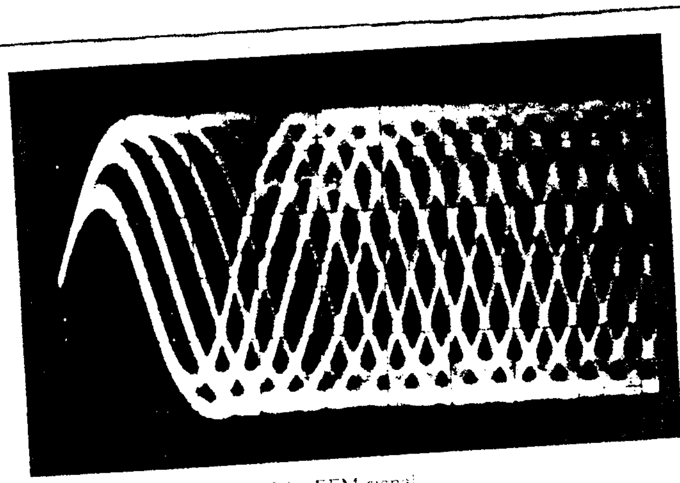


FIG. 9.32 Eye diagram of the EFM signal.

ANALOG SIGNAL PROCESSOR (ASP)

This is a bipolar analog signal processor and servo controller IC for CD players. Signal processor and minimum of external components to from a complete controller for a compact disc player.

The function of this circuit is underlined below

(i) RF amplification

(ii) Slice level control

(iii) Voltage - controlled oscillator (VCO)

(iv) VCO control amplifier

(v) Automatic laser power control

(vi) focus error amplifier

(vii) Tracking error amplifier

(Viii) Track jump amplifier

(ix) Focus servo preamplifier

(x) Track servo preamplifier

(xi) Spindle servo preamplifier

(xii) Sled servo preamplifier

(xiii) RF detection circuit

(xiv) HF level detection circuit

(xv) Defect detection circuit

(xvi) Shock detection circuit

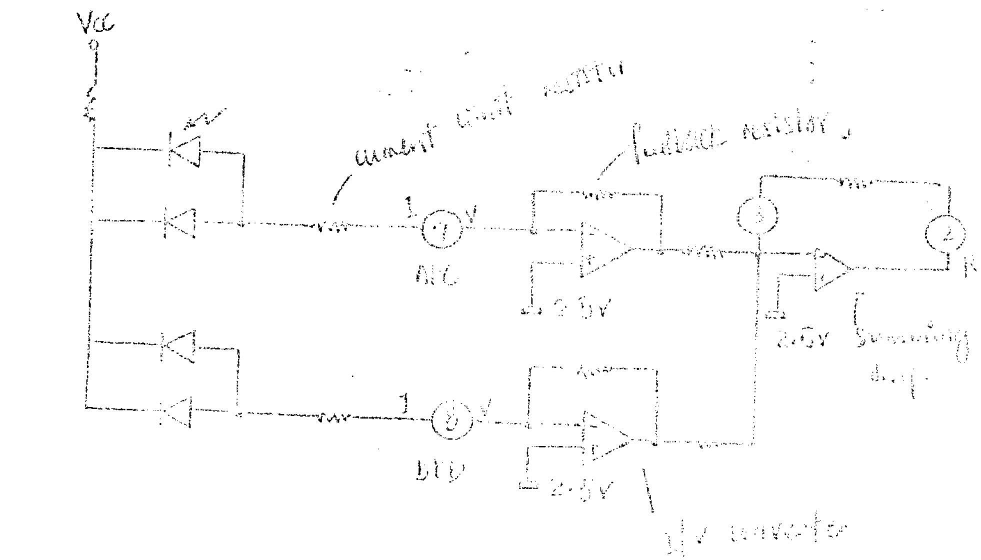
(xvii) Focus switch

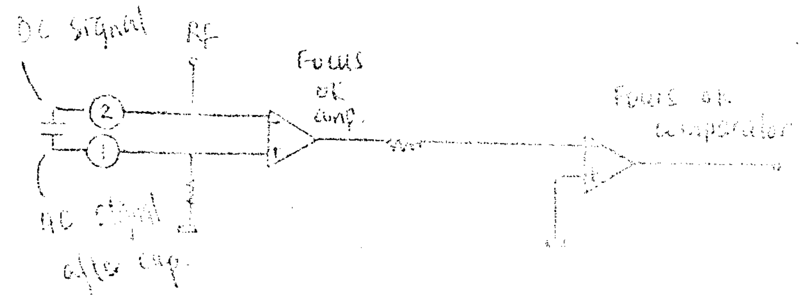
(xviii) Tracking servo gain switch

(xix) Tracking error slice comparator.

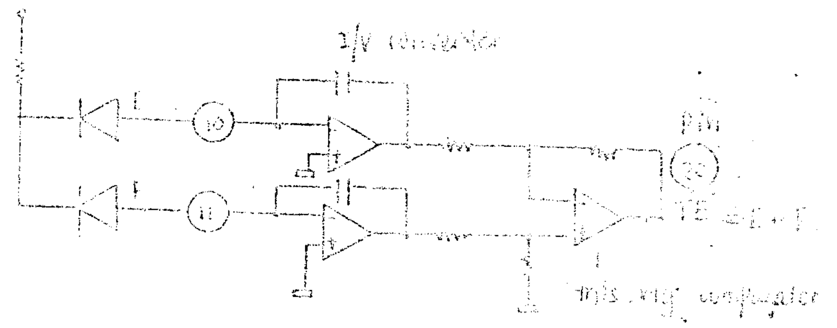
RF amp.

RF - Radio freq

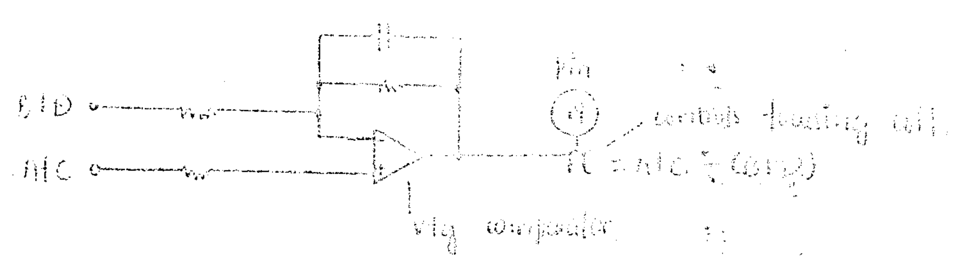




Tracking Error Amp - controls the tracking coil
E & F are signals.



Focus Error Amp.



EXPLANATION OF SP AND SURROUNDING CIRCUIT DIAGRAM

The above values are representative examples amplitude of HF signal is being decided by RNF = 33 K ohm. pass through HF coil (3T level is about 1.4 MHz compensation) for amplification and wave form shaping then enter DSP. HF amplitude at HF test point is around 2V - 2.7V. If HF amplitude is lower than that, it is necessary to check pick-up laser power. Also when measured jitter value at EFM test points. It reading should be around 20 ns and below is normal. Average value is around 14-15 ns measurement condition at this time should set at 600 ns - 800 ns for window width of jitter meter and 2.5 V for thrust level.

PLL CLOCK PLAY BACK CIRCUIT

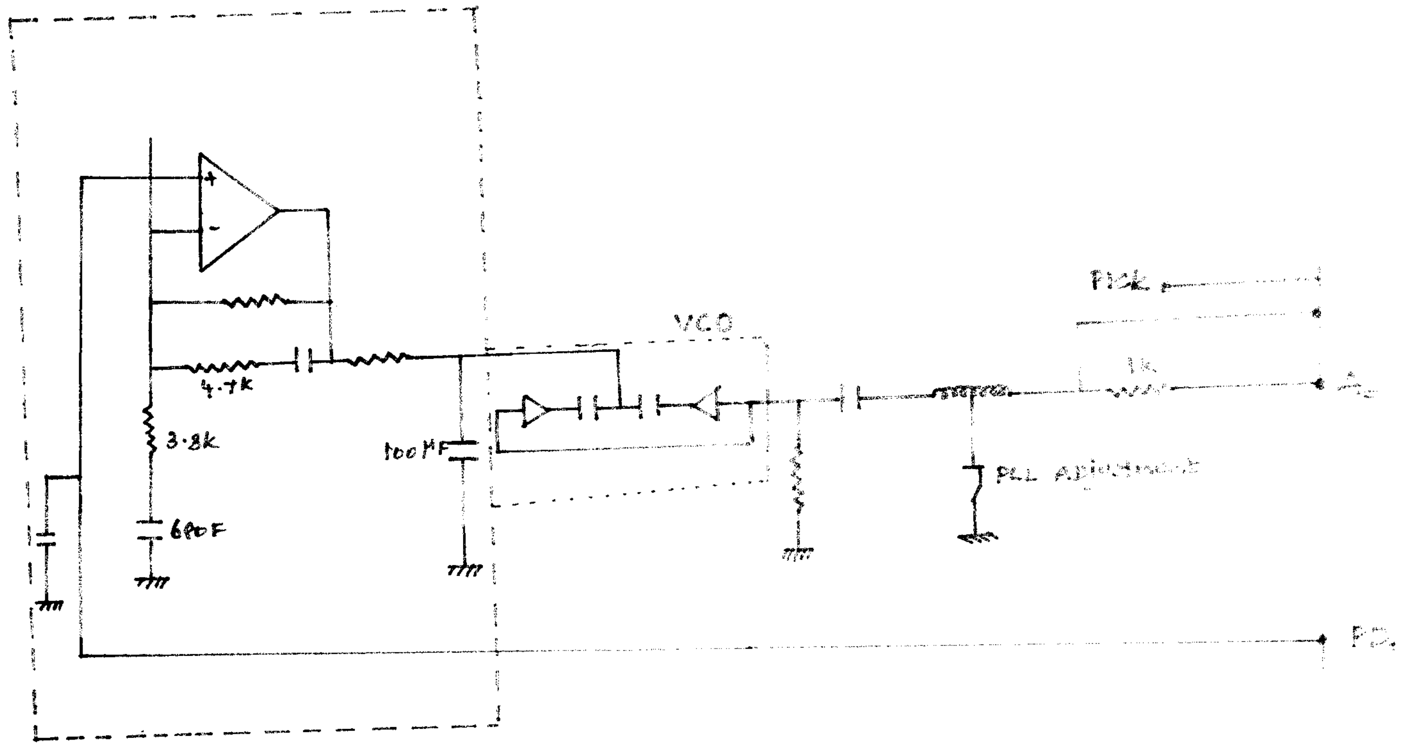
Above diagram is representative example of PLL circuit. VCO free run frequency alignment is to turn PLL coil while monitoring from PLCK terminal. Amd adjust f=4.3218 MHz. If this frequency is off-position. It will takes too much time staring a song. When search difference in between inner circumference and outer circumference is too great. allowable is about 4.31 - 4.33 MHz.

OUTER SPEED = 200rpm

INNER SPEED = 400rpm

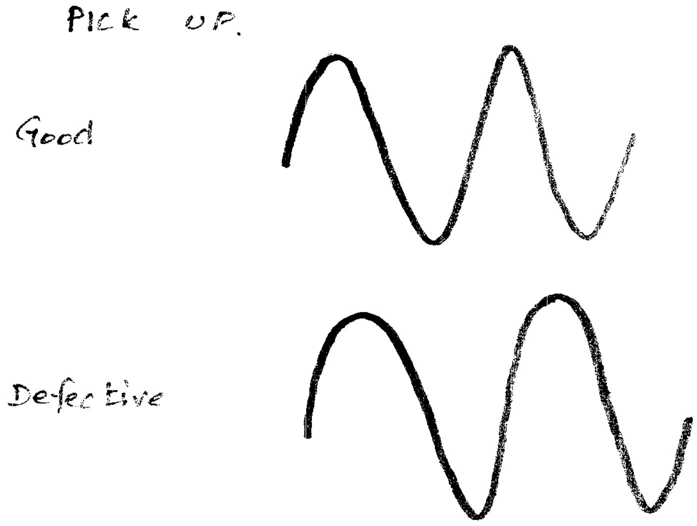
TRACKING SEARCH AND KICK PULSE

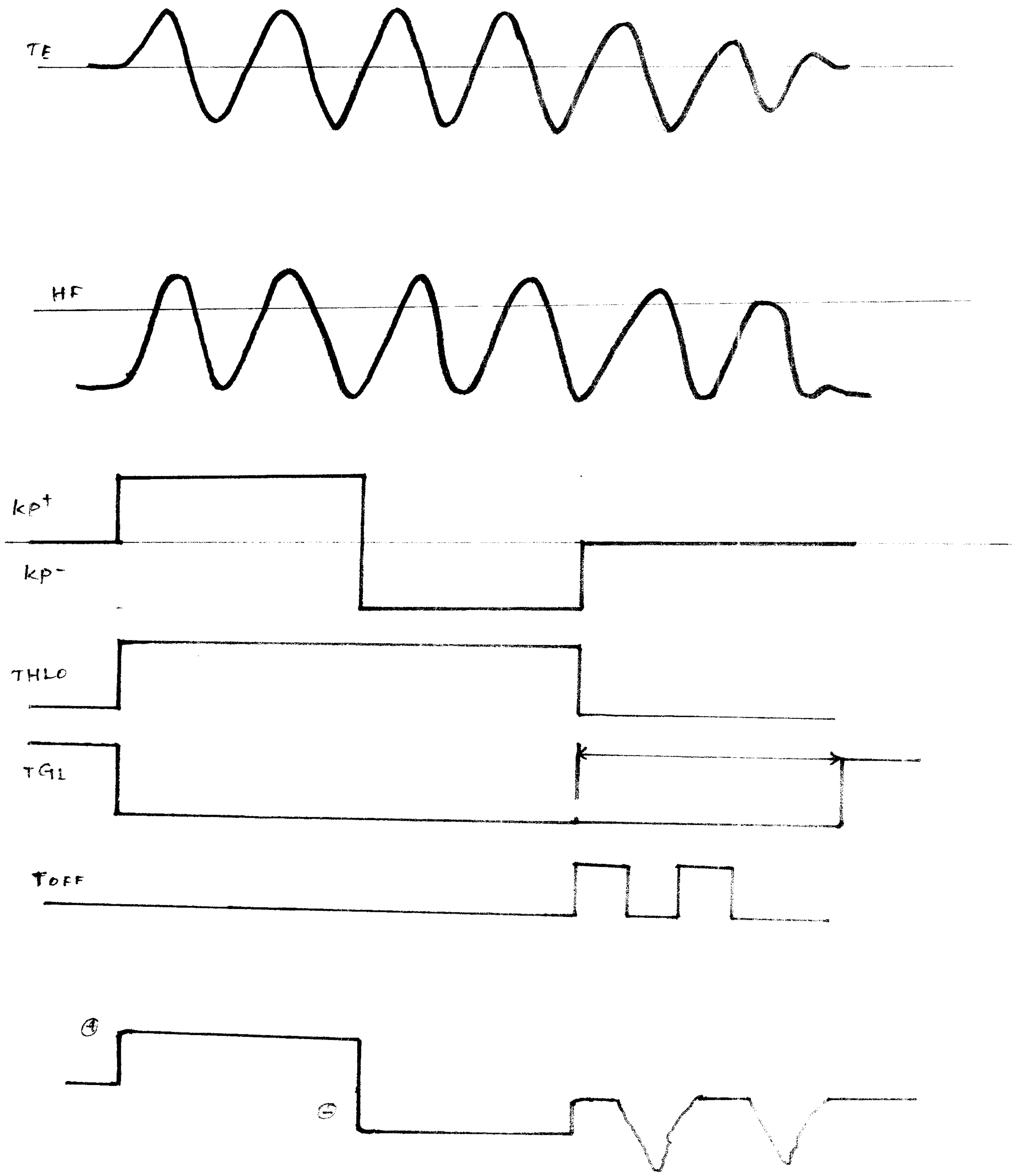
In sanyo DSP the kick pulse is controlled by command from microprocessor. It's selection is decided by sequences explained in search function item.



PLL CLOCK PLAY BACK CIRCUIT.

Pick UP.





TRACKING SEARCH AND KICK PULSE

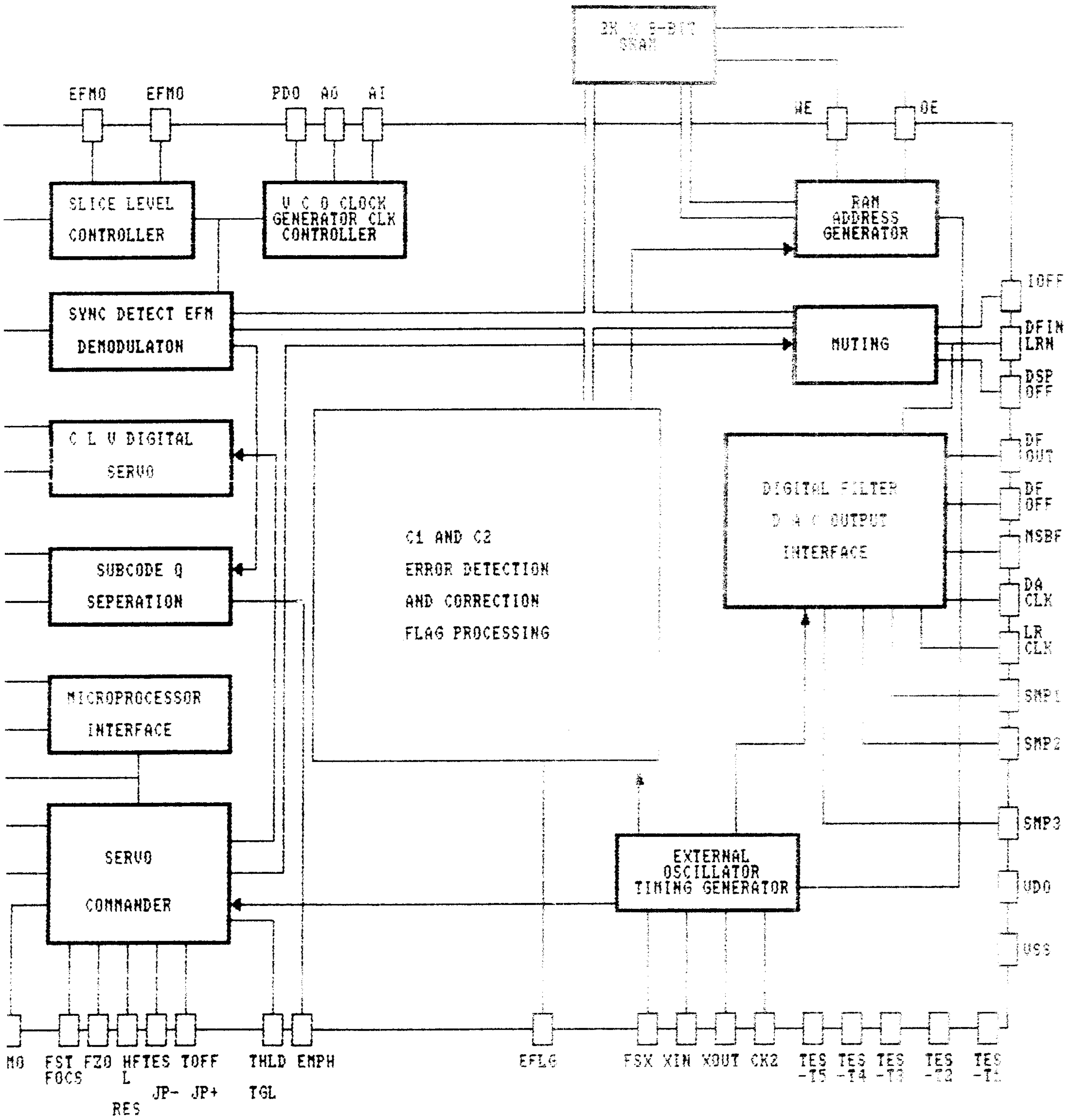
Above digram is example of 4 track JMP (external) according to command input. First speed pulse (Kp+) occur next reduce speed pulse (Kp-) occur when kick pulse appears. Loop of servo series become front position hold mode. THLD become "H" After that according to brake signal (TOFF) which is formed by servo cross signal from TES and traverse signal HFL obtained from HF. Break off unnecessary oscillation after jump of Actuatuion (coil) is being carried out. TCL terminal is normally "H" in order to gather and bundle servo easily diruing above search it become "H" increase tracking gain also TOFF at stop condition is "H".

FOCUS SEACH CIRCUIT

Focus servo when S curve of FE output is at point A of the diagram. At this time FZD falling. Focus is reset. DRF is "H" when received reflected light from the disc. This DRF signal is being input into micro-processor to judge whether has disc or not. In micro-processor by sending command of Focus start to DSP the above Focus search function is being carried out interval etc, is decide by content of micro -processor software. And if there is no disc pick-up will of up and down 3 times and then stop, Amplitude of focus coil voltage is about 1.2 V now It will shift $\pm$ 0.84 mm by pick up will if there is no pit on disc surface. It becomes no focus.

The driving method of focusing servo resembles a dynamic speaker, when voltage is supplied to focus drive coil as shown in diagram above, the objective lens will move up and down in vertical direction.

For tracking servo, there are suspension on four side of the objective lens, when voltage is supplied to tracking drive coil which is installed. The side of the bobbin, the objective lens will move in horizontal direction.



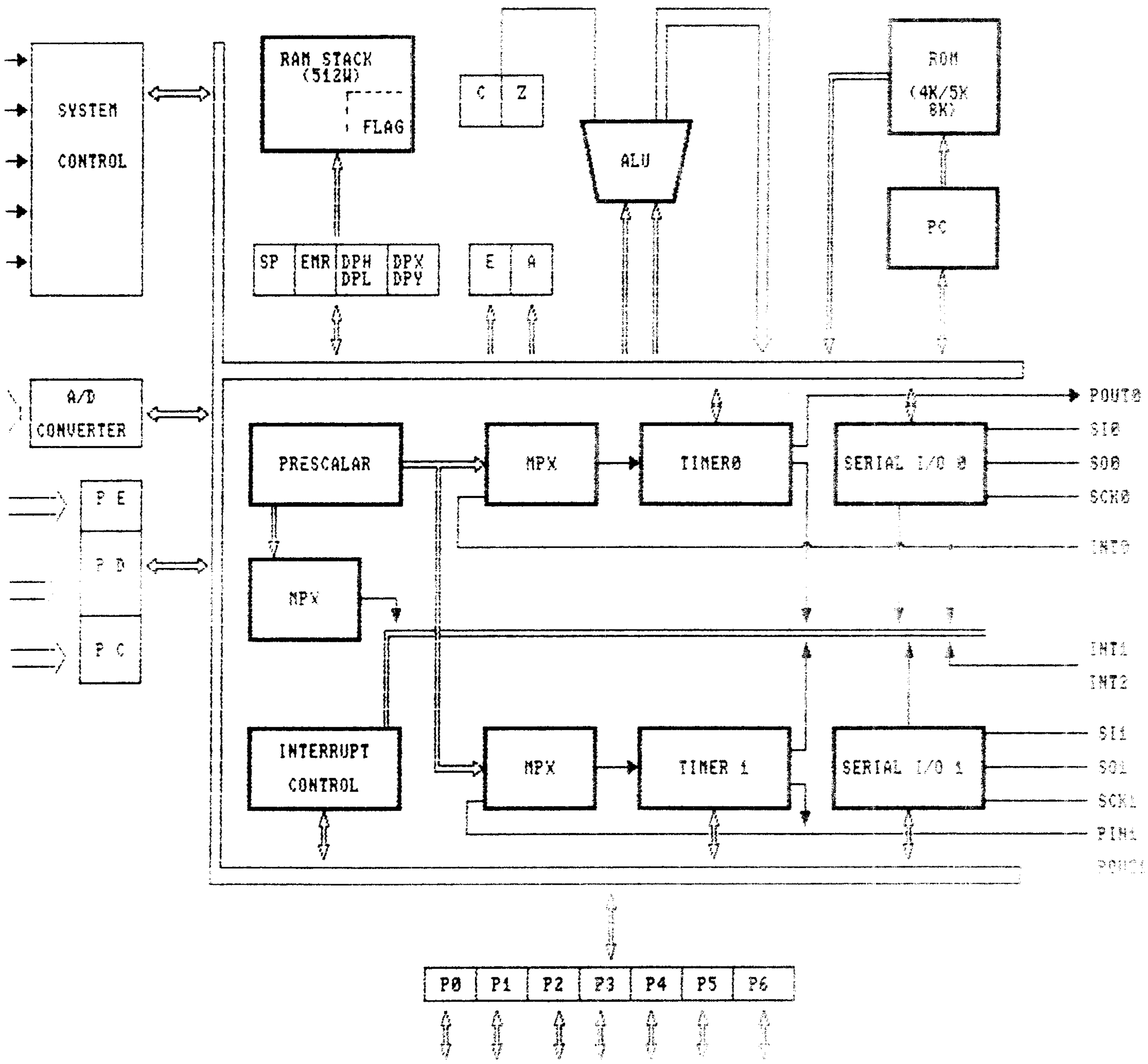
BLOCK DIAGRAM OF DIGITAL SIGNAL PROCESSOR.

DIGITAL SIGNAL PROCESSING :-

Figure shows a block diagram of digital signal processing.The demodulation of EFM can be accomplished by using the conversion process shown in the figure.This provides the digital audio data and parities for error correction (CIRC).At the same time ,the subcoding that directly follows the synchronization signal is demodulated and sent to the control and display block.THe data and parities are then temporarily stored in a buffer memory (2K bytes) for the CIRC decoder circuit.The parity bits can be used here to CIRC is one of the most powerful error-correcting codes,if more errors than a permissible maximum occur ,they can only be detected and used to provide estimated data by linear interpolation between preceding and new data.

At the same time the CIRC buffer memory operates as the de-interleaver of the CIRC and is used for time-base correction (TBC).If the data are written into the memory by means of the recovered clock signal with the PLL and then read out by means of the crystal clock after a certain amount of data has been stored,data can be arranged in accordance with a stable timing rate .In this way wow and flutter of the digital audio signal are reduced to a level equal to the stability of the crystal oscillator.

SYSTEM BLOCK DIAGRAM



Microcontroller

This is a 42-pin package type COMS 4 bits Single-Chip micro-controller. This controller contains

ROM

RAM

I/O PARTS

A/D Converter

Timer

The main function of this controller is

1) System Control

2) Subcoding control

3) Pickup system

The main system controls are

Volume control(up or down)

Power On/Off control

Sled motor control

Play

Repeat

Subcoding

Command data

Sub-Q Data control

Sub Q trigger control

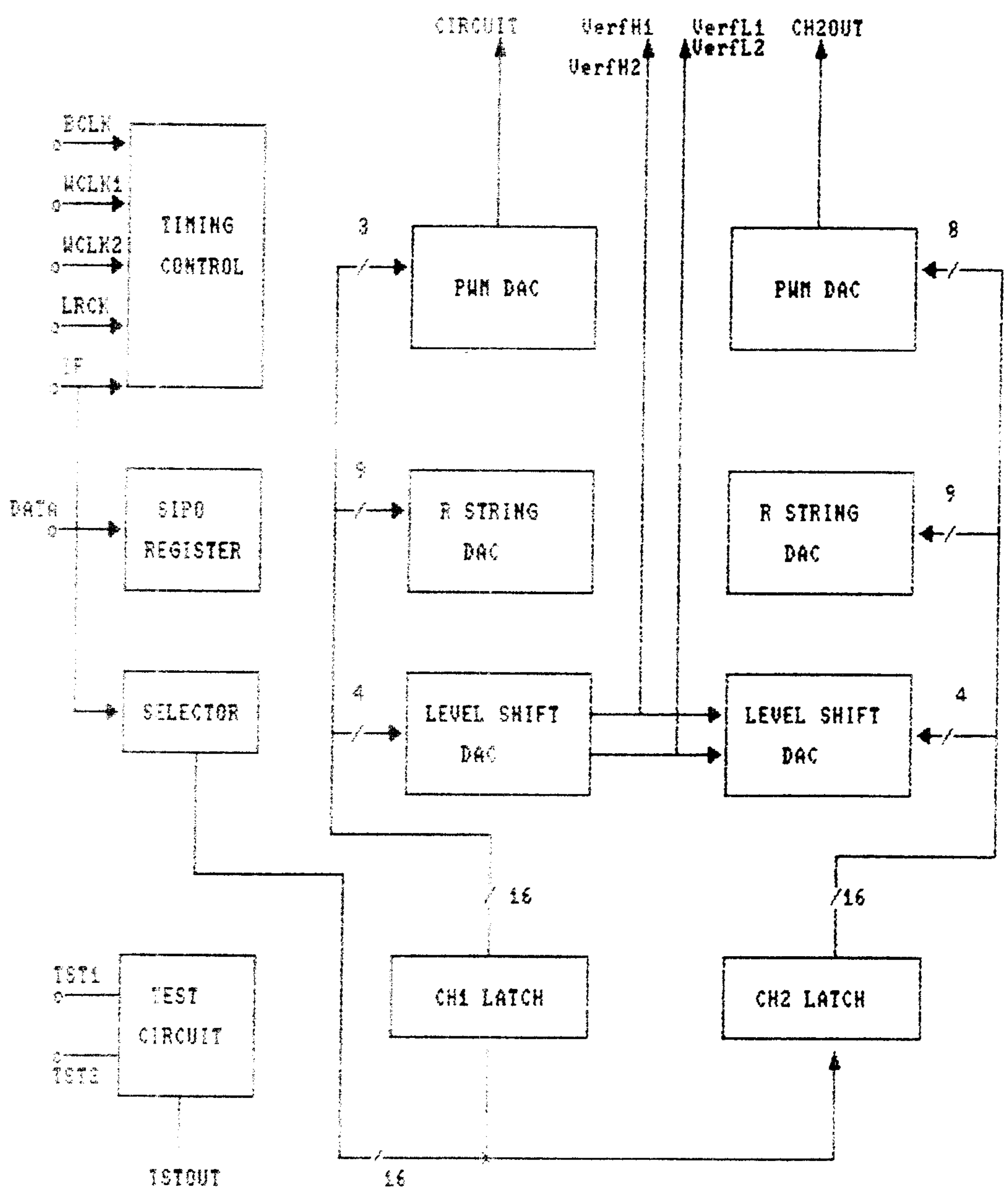
Display Control

This microcontroller is interfaced with the Display system (8 segment LED) for various control indications.

3)Pickup Control

In the search mode, the player's microprocessor takes command to provide faster motion than is possible during normal tracking. Control signals are directed to the pickup for accelerated movement in the forward or reverse direction. Subcode data added to each CD frame is used to determine the pickup's location on the pit spiral.

For forward or reverse jumps to programmed locations on the disc. The tracking signal is disabled, and the microprosser controls the pickup. Just prior to alignment a break. Pulse is generated to compensate for the pickup's inertia. When the correct location is reached, the microprocessor informs the system that proper alignment is imminent, the tracking signal is again activated, the pickup comes to rest on the correct track, and the system resumes auto-tracking.



BLOCK DIAGRAM OF D/A CONVERTER.

POWER DRIVER

This IC is used to drive the actuator motor of the CD player. With a 5v regulator and a general purpose operating amplifier built in.

This power driver is in a very small package, so the set can be made smaller. A thermal shutdown circuit is built in. Gains can be adjusted using an external resistance.

D/A CONVERTER

Here two D/A conversion channels are built in and the O/P to the right and left channel is in-phase making this device ideal for digital audio applications.

This D/A converter uses dynamic level shift, combining resistance strings and PWM with level shifts.

The main features of this D/A converter is it is compatible with 2's complement code, and 176.4 KHz (max conversion frequency (with 4 times oversampling capability).

‘CD’ Working

CIRCUIT WORKING

LIST OF CONTROLS

1. SVR302 : Tracking Balance Adjustment

SEQUENCE OF OPERATION

1. Power ON
2. Microcontroller, Pin 15 (XRST) is Reset.

(Goes "Low" momentarily)

3. Laser device ON through LASER DRIVER (Transister)
(Automatic Laser Power Controlled by ASP)

4. Photo diodes receive reflected signals from the disc at the pin 79 (FIN2 : A + C) and pin 78 (FIN1 : B + D) of (ASP). These are compared in the RF comparator and output at pin 40 (DRF : RF detect) of (ASP and fed to pin 2 (DR) of micro-controller.

5. Disc motor control outputs are fed from pin 10 (CLV+) and pin 11 (CLV -) of DSP to pin 52 (CLV-) of ASP depending on the detected HF signal. The spindle driver output from pin 31 (SPDO) of ASP is fed pin 3 of from pin 1 and 2 of IC401. The spindle motor/disk motor are driven by output from pin one and two of IC TRACKING AND FOCUS SERVO DRIVER.

6. Focusing started when pin 14 (FST) of DSP goes low and lens is lowered pin 13 (FOCS) is high as the lens is gradually raised. At the In-focus position pin 15 (FZD) of DSP becomes low and resets the FOCS signal (LOW) and turns on the Focus servo circuit. Focussing is controlled by the pins 11 and 12 of T&FSD depending on the control output from pin 22 (FDO) of ASP.

7. Tracking photodiode outputs are input to pins 3(F) and 4(E) of ASP. The Amplified error P/O comes from pin 21 (TDO) of ASP and is also used to control sled motor continuosly.

8. The pick up from the photodiodes after receiving the reflected beam, convert into electrical signal. The amplitude signal is taken from pin 72 (RFSM) of ASP. This is fed to pin 8 (EFM IN) of DSP. The peak to peak level of cycle of the HF signal, the inphase signal from pin 7 (EFMO) and out of phase signal from pin 6 (EFMO) of DSP are fed to pin55 (AI) and pin 54 (EFMO) of ASP. The slice level control Amp O/P is fed back from pin 53 (SLCO) of ASP to pin 8 (EFM IN) of DSP.

9. The VCO is generated internally in ASP between pins 2 and 3. The O/P from PM4 (PDO) of DSP is fed to pin 57 (PDO) of ASP. The VCO control Amp O/P (VCOO) from pin 60 of ASP is fed to pin 3 (AI) of DSP. The phase detection O/P from pin 4 (PPO) of DSP is fed back to pin 57 (PDO) of IC301 to increase / decrease the frequency ad required.

10. Tracking hold control command is generated at pin 22 (HLD) of DSP according to track jump command received at pin 21 (TES) and pin 16 (HFL) of DSP. This in turn feed the required kick pulses from pin 19 of ASP feeds the required tracking error signal to the T coil through the TDO O/P of ASP.

11. The EFM signal received is then demodulated within the Dig Sig Processor after sync detection. The demodulated data undergoes error detection and correction and subcode cyclic redendancy check by interfacing with the microprocessor between the pins

50, 51, 52, 53 and 54 of DSP and pins 9, 10, 11, 12 and 13 of microcontroller. The demodulated data is also interfaced with the built in RAM (in DSP). The O/P from pin 35 (DFOUT) of DSP in (serial) digital format.

12. The O/P from pins 33.35 and 36 (of DSP) are fed to the DAC pins 5,6,7 respectively.

13. Pin 29 (EMPH) of DSP switches on the emphasis transistors Q781 and Q881 depending on whether the I/O data was pre-emphasied or not.

14. The audio signal O/P is from Pins 1 and 20 of DAC which are filtered by the RC n/w LPF (R782 and R882, C882).

15. The audio mute signal is fed from pins 14 of microcontroller to the transistors Q782 and 882 during the required period.

16. Pin 37 (SYNC REC) of microcontroller receives signals through the Rec switching transistor Q501, from the main Amp circuit which initiates the PLAY operation during SYNCHRO RECORDING.

17. LED display is driver from pins 17, 18, 24, 29 of microcontroller. Different functions like Play,Stop, Skip and Search are performed by shorting the respective keys inputs.

SMT/SMD

SURFACE MOUNT TECHNOLOGY

In this project we have used surface Mounted components like

Ceramic capacitors

Fixed Registers

SMD IC Packages

Coils

Filters

SURFACE MOUNTED DEVICES

DEFINITION

At the leading edge of development of new requirement of technology are the use of smaller chip components and short pitch IC packages. This new family of electronic components called surface mount devices (SMD) have in many cases a different design and production techniques as compared to conventional leaded components.

SMT/SMD DEVELOPMENT

SMT assembly technology has now matured to an extend that it is firmly rosted in various products requiring high density packaging based on high functionality and requirement for SMALL-ER, THINNER, LIGHTER and LESS EXPENSIVE PRODUCTS.

The use of SMT offers

The use of SMT offers

1. Small size
2. Higher Performance
3. Improved reliability
4. Lower cost for electronic assemblies.

The market demand for small dimensions and light weight products has resulted in very rapid trends towards electronic equipment of smaller volume.

The products like portable CD player personal telephone have decreased both in mass and volume to about one-tenth over the past ten years. This trend in minimization is shown in table.

TECHNOLOGY TRENDS FOR MINIATURISATION

Description	1984	1986	1992	units
Typical Track	300	200	50	micrometer
Component density	4	6	20	pcs/cm^2
Number of board layers	1	2	6	
Board thickness	1	0.8	0.6	mm
Typical IC Pitch	1.27	0.8	0.3	mm

Table II

The development of increase in packaging density is given in Table II

Mounting on PRINTED BOARDS

Year	product	Active Parts	Soldered Joints/cm^2
1955	Radio	Tubes	0.5
1964	Black + white T.V	Transisters	1
1970	Colr T.V	Transiters + ICS	3
1988	CD Player	ICS + Transistors	20

THE MAIN ADVANTAGES:

Reduction in EMI emission

S/N ratio

Assembly Cost

The capacitors for better impulse response faster and more accurate rise time characteristics.

Consumer electronic products like our CD player, which largely use registers, capacitors and other discrete components and have lead pitch of more than 0.5 mm, LSIs with pitch as low as 0.1 mm requires very high precision so for the above reasons we have used surface mounted components for our project.

STATIC ELECTRICITY PRECAUTIONS

1. Use static Electric Prevention Belt.
2. Use Deionizing brush.
3. Use hand strip of 1M OHM.
4. Use electric conductive gloves.
5. Use earth attach soldering iron
6. Use spacer
7. Use electrical conductive part box.
8. Use electric conductive mat.
9. Use electric conductive sheet.
10. Use Deionizer.
11. Conveyor line must be gounded.
12. Measuring equipment must be grounded.
13. Electrical screwdriver must be grounded.
14. machine use on line must be grounded.
15. Dipping machine must be grounded.
16. Use conductive bag.
17. Avoid putting expose equipment which generate magenetic flux.
18. Avoid using magnetic tool.
19. All Tools, Instruments, Etc., used for measuring must be grounded.
20. Wiring wrapper must be grounded.

HANDLING PRECAUTIONS

1. Do not drop the pickup or subject it to impact.

2. Power plug must be remove before attempting to replace any component.

3. Never repair or store the pickup where the temperature or humidity is high.

4. Never expose pickup to dusty environment.

5. Never adjunct any adjustable parts on the pickup.

6. Never user your hand touch the pickup lens.

7. Use air brush to clean dust away.

PCB Design

PCB DESIGN

The starting point in the design of PCB from the schematic diagram of the circuit. This PCB is particular designed for mounting the component in the surface of the board itself.

ie.,) SMD ICs

Fixed Register

tantulam capacitors etc.

This is designed using the software package. ie., Automated Artwork.

LAYOUT

The layout of a PCB has to incorporate all the information on the board before going to the preparation. This layout is designed using the detailed circuit diagram.

ARTWORK

In this system we need very high accuracy so the artwork is produced at 2:1 Scale (layout/artwork).

SUPPLY & GROUNDING

In our system (CD signal processing where analog and digital circuits have to be accommodated on the same PCB the complete separation of the corresponding ground conductors are made. This is to avoid any current spikes from the digital part reaching the analog part where they could get amplified and cause errors, oscillation or instability with the whole analog part.

AUTOMATED ARTWORK DESIGN

The layout sketch is prepared from the designed circuit schematic. Digitizing is a graphic-to-digital conversion of all the relevant information incorporated in the layout sketch such as component location solder pad size, conductor configuration etc. The resolution of this digitizer is in the range of 0.025-0.1 mm.

COMPUTER

The pulse trains from the digitizer are temporarily stored in the storage system of the minicomputer. The data are then processed by the computer in OFF-line mode and the O/P is usually on plotted on paper.

In the photoplotter, the processed digital data are converted back to graphics by photographic means. The head which is a high spot projector changes its relative position towards the drawing medium which is the photographic film.

Testings and Measurements

PURPOSE OF AUDIO MEASUREMENTS

Measurements are made on audio equipment to check performance under specified conditions and access suitability for use in a particular application. They may be used to verify specified system performance or as a way of comparing several pieces of equipment for use in a system.

Measurements may also be used to identify components in need of adjustment or repair. Whatever the application audio measurements are an important part of audio engineering.

Many parameters are important in audio devices and merit attention in the measurement process.

Most measurements in audio are composed of measurements of fundamental parameters. These parameters include

SIGNAL LEVEL

PHASE & FREQUENCY

Most other measurements consists of measuring these fundamental parameters and display these results in combination by using some convenient format.

Ex., Signal to noise ratio (SNR) is a pair of level measurements made under different conditions expressed as a logarithmic (or) decibal dB ratio.

TEST DISCS

Any test needs a standard source. In the case of CD players, test discs are available from Philips (410 055-2/3, 410 055-2/5, 410 055-2/5A and SBC 429), PolyGram (DHF19-410741-2), Sony (CD3-YED-S7 and 48DG3), Denon (38C39-7147 and 33C39-7441), Technics (SH-CD001), Pierre Verany (PV788031/788032), CBS (CD-1) and other manufacturers, as well as organizations such as the Japan Audio Society (CD-1 YDDS-2). Hi-Fi News (HFNO-03) and the Electronic Industries Association of Japan (EIAJ CD-1 YGDS 13). They contain diverse test signals and musical selections.

The CBS test disc CD-1 deserves special mention. It includes all of the tests specified by both the Electronic Industries Association (EIA) and EIAJ, along with several additional tests devised by CBS. It contains tracks of low-level amplitude-swept and amplitude-stepped signals (from about -70 dB to -100 dB) recorded with dither. These permit measurement of linearity and distortion without the effects of quantization noise. The use of dithered test signals is crucial at low amplitudes because it would be otherwise impossible to separate the player's distortion from those in the test signal itself. The -100 dB signal demonstrates the ability of the CD system to accurately reproduce signals below the least significant bit when proper dither has been added. The contents of the CBS CD-1 test disc are shown in table 5.1.

CBS J

CD-1 TEST DISC

FOR MEASURING CD PLAYER PERFORMANCE WITH E.I.A STANDARD SIGNALS

1 Reference, L & R, O dB, 1 kHz

2 Left Separation, O dB, 1K, 125, 4K, 10K, 16K Hz

3 Right Separation, O dB, 1K, 125, 4K, 10K, 16K Hz

4 Signal/Noise, L & R, Infinity Zero w/o emphasis

5 Dynamic Range, L & R, 1kHz, − 60 dB w/o emphasis

6 Frequency Response, L & R, O dB 4, 8, 17, 31 Hz

7 61, 127, 251, [illegible] Hz

8 997, 1999, 4001, 7993 Hz

9 10 007, 12 503, 16 001, 17 989 Hz

10 19 997 Hz (Also used for Pitch Error)

11 Sweep Frequency Response, O dB, [illegible] kHz

12 De-emphasis Error, L & R, 1K, 125, 4K, 10K, 16K Hz

13 Intermodulation Distortion (SMPTE, Two Tones), L & R 60 Hz + [illegible]

14 Wow & Flutter, L & R, O dB, 31[illegible]

15 Access Time, L & R, O dB, 317 Hz

16 Square Wave, L & R, O dB, 1 002.27 Hz

17 Impulse & Polarity Test, [illegible]

18 [illegible] 997 [illegible]

[illegible] − 20, − 30, − 39.96, − 4[illegible], − 59.94, − 70.[illegible] − [illegible], − 90.01 dB

19 Linearity with Dither 997 Hz, L & R − 70.01, − 80.77, − [illegible], − 100 dB

[illegible] Noise with [illegible]

[illegible]

Additional Test Tracks

The EIAJ has issued a test standard in two reports (see the references at the end of this chapter), "Methods of Measurement for CD Players" contains the following thirteen tests: frequency response, signal-to-noise ratio, dynamic range, total harmonic distortion, channel separation de-emphasis error, wow and flutter, intermodulatioo distortion, phase difference between channels, level difference between channels, output voltage, pitch error, and access time. In addition, it specifies a standard method for performing and documenting the tests. "Test Disk for CD Players" describes a standard test disc to be used to perform such tests.

The signals on test discs are computer - generater to guaranted accuracy. Standard specifications can be measured with thier menus of signals. Of course, test gear is needed to decipher the results; a good test bench has an oscilloscope, a distortion and level meter, and a harmonic analyser. (Indeed a good test bench should also have a CD player and test disc. They provide a highly accurate set of reference signals usefull in testing other audio equipment, room acoustics, etc) As an alternative to separate pieces of measuring equipment, integrated, computer - based test sets offer high accuracy and fast operation.Equipped with a test disc, measuring gear, and the player under test, we can run some numbers.

FREQUENCY RESPONSE

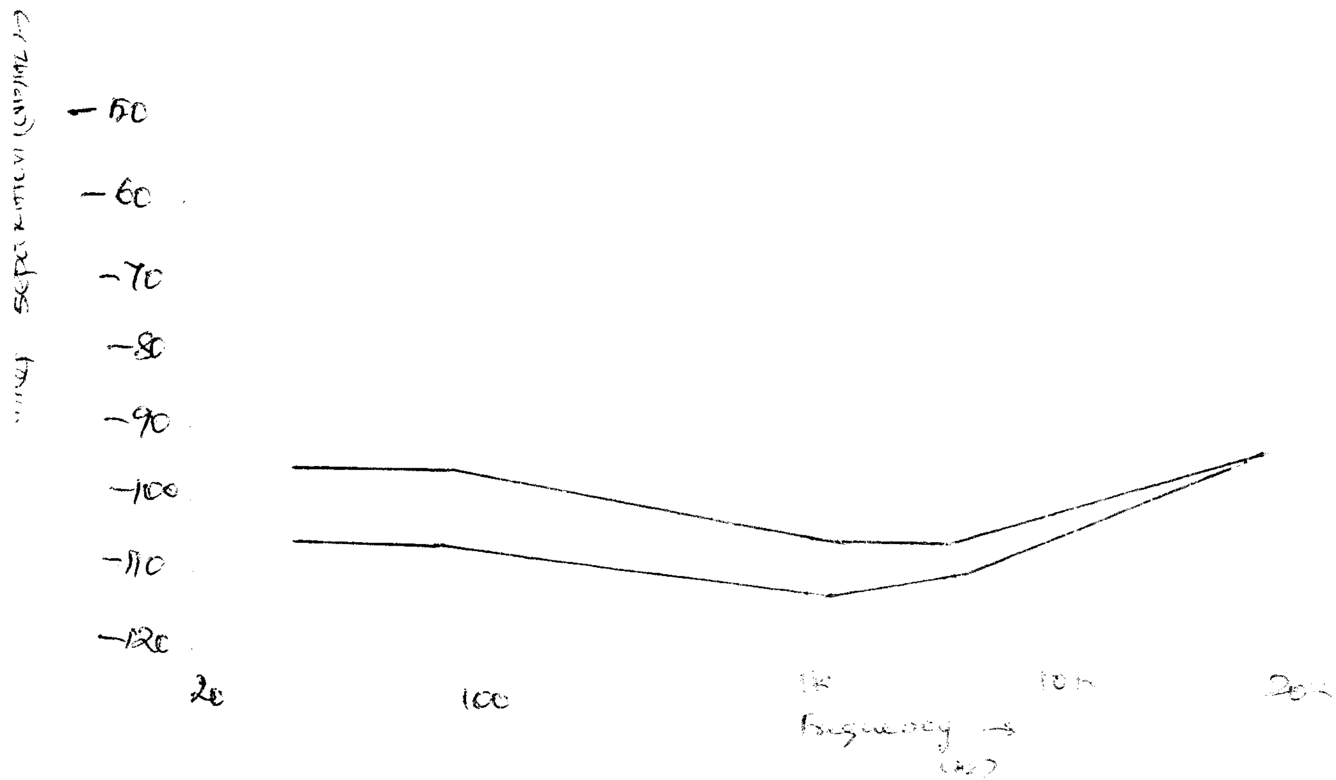
Frequency response measures a signal's amplitude deviation over the audio bandwidth of 0 to 20 KHz ;any amplitude deviation, either positive or negative, is measured in decibels (dB) relative to a 0 dB (maximum) amplitude at 1Khz. The test can be performed by using either independent tones or a swept tone recorded at 0 dB. The two channels are tested separately. By definition, the digital signal should have a perfectly flat response, but the output analog circuitry usually interferes. A single channel measurement is shown in fig.,

Frequency response measures one of the most audible aspects of any audio system. A slight difference in frequency response can amount for diverse qualitative adjustments including image, depth, air, clarity and warmth. Fortunately the frequency response of digital audio equipment in one of its fortes.A good CD player will have an essentially flat response with a negative deviation of perhaps a few tenths of a decibel at the high frequenc es

CHANNEL SEPARATION

Channel separation measures the ability of the player to keep the stereo signals distinct from each other: again the problem lies mainly in the analog output circuitry and usually grows more severe at high frequencies. For this test, 0 dB reference tones of various frequencies are played through one channel and the crosstalk is measured on the otherwise quiet channel as shown in fig . When a broadband measurements is

FREQUENCY RESPONSE MEASUREMENT



CHANNEL SEPARATION MEASUREMENT

taken, crosstalk of both the fundamental and its harmonic components are measured. Many manufactures prefer to measure only crosstalk from the fundamental frequency under test. Thus a filter (a 22 KHz low-pass or a selected band-pass) is used to remove wideband signals which could decrease the measured value of separation.

Whatever method is employed the crosstalk level is measured as an amplitude in dB. A fig of -100 dB at 1Khz for example means that 1KHz information on one channel will also appear on the other channel but down 100 dB. A channel separation frequency response may thus be plotted. Curiously, channel separation is usually not reciprocal between two channels. A good player should have channel separation of more than 90 dB across the audio band.

TOTAL HARMONIC DISTORTION PLUS NOISE

Total distortion plus noise (THD + n) measures a device's ability to reproduce a signal without adding components of its own. Specifically distortion results if the device adds new harmonic information. For example if a 50 Hz tone was applied 50 Hz would be reproduced as well as harmonics: 100 Hz second harmonic, 150 Hz third, 200 Hz fourth etc., Devices such as D/A converters are often guilty of contributing harmonic distortion.

The THD + N of a device is measured by inputting a sine wave (a signal with no harmonics) then filtering out the sine wave at the output and measuring the remaining harmonic signal as a percentage of the original signal. A THD + N analyzer accom-

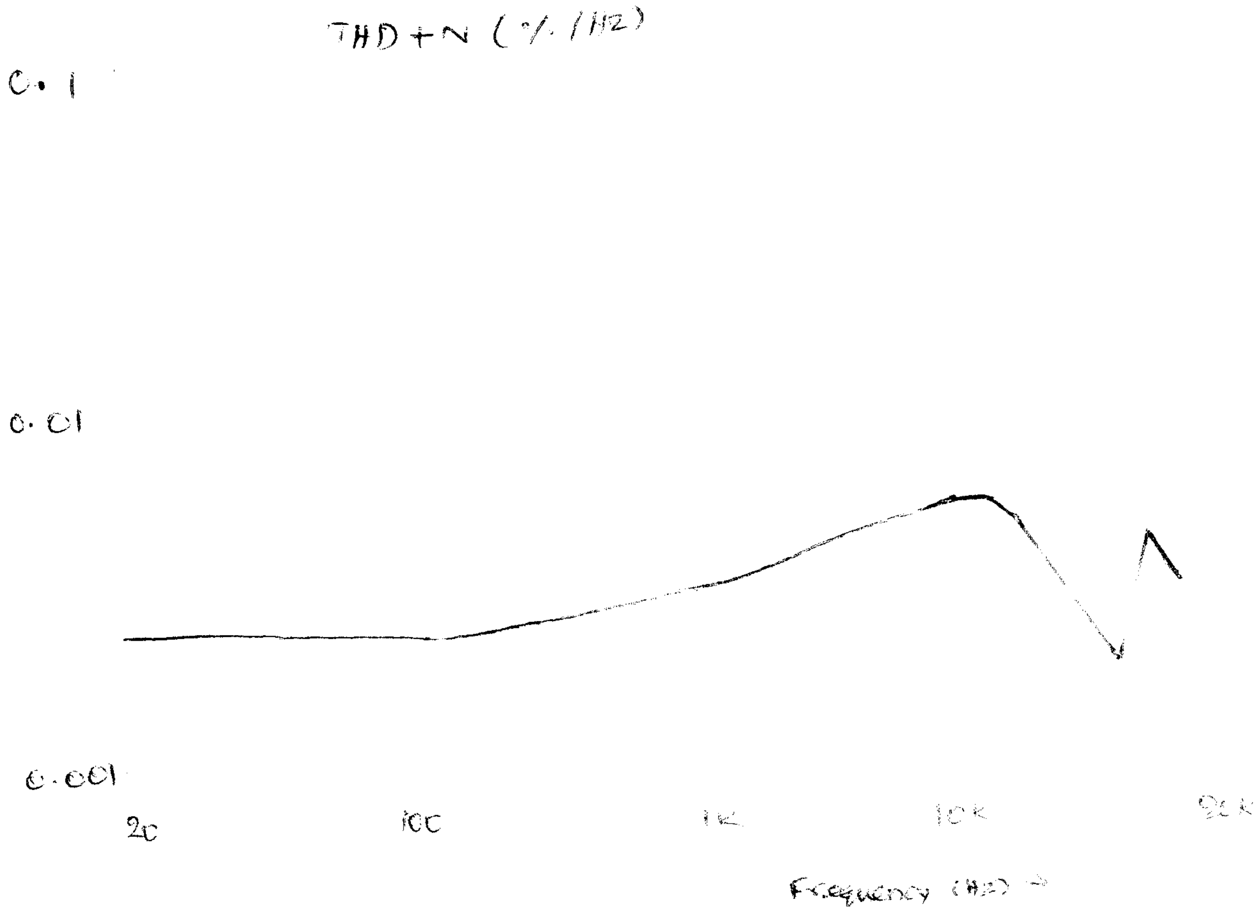
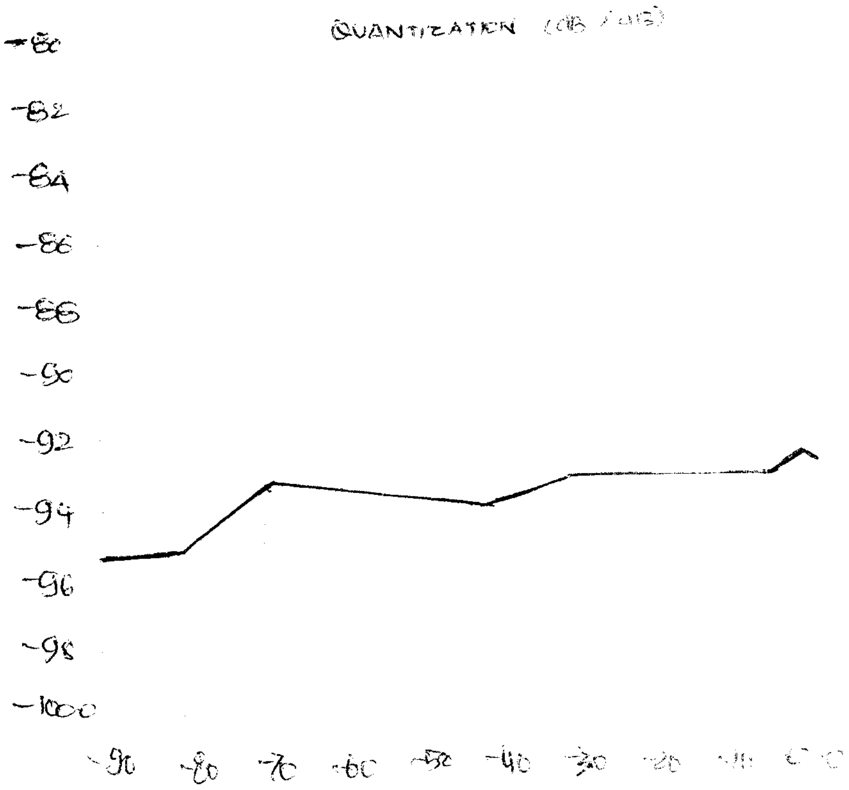


Fig: TOTAL HARMONIC DISTORTION PLUS NOISE (THD+N) MEASUREMENT.



(THD VERSUS LEVEL) MEASUREMENT.

plishes the process directly as seen in figure . THD + N varies with respect to both amplitude and frequency. It must therefore be measured over a wide frequency range and at different levels. Unlike analog system in which THD + N increases with amplitude, a digital system's THD + N decreases with amplitude CD player manufactures wage war over THD + N measurements often achieving levels below 0.002 percent.

The THD + N can be measured at frequencies recorded at 0 dB and lower amplitudes such as -24 or -30 dB. Some players show a rise in distortion at high signal amplitudes due to distortion in the analog output stages as well as bit weighting errors in the D/A converter which increase quantization error. Steps in the measured distortion over amplitude indicate the latter. A rise in THD + N at high frequencies is often observed particularly in players with poor low-pass filters. This is usually not due to harmonic distortion but is caused by the beat frequency between the CD player's clock and the test signal will measure 14.1 KHz.

The EIAJ and most manufactures insert a 22 KHz low-pass filter between the player's output and the distortion meter. Otherwise spurious beat frequency components related to the player's sampling frequency add to the distortion measurement whereas in fact they are not harmonic surely it limits the use of the measurement for example a 24 KHz low-pass filter removes all legitimate harmonic components for frequencies under test greater than 12 KHz (24 KHz is the second harmonic of 12 KHz). Thus THD +

N cannot be accurately measured for frequencies greater than 12 KHz. Thorough reviewers are careful to examine the unfiltered output signal with a spectrum analyzer to detect any components, harmonic of otherwise which might degrade performance. Because of the low distortion levels to be measured care must be taken in grounding and interconnecting the player to the test equipment.

QUANTIZATION NOISE

Quantization noise can be evaluated by measuring distortion at very low signal levels. To measure quantization noise, a dithered test signal at various low-level amplitudes is used. THD + N is measured thus removing the fundamental component of the test signal. In addition low-and moderate order harmonics may be removed with a high-pass filter (400 Hz) to prevent distortion products from obscuring the quantization noise measurement across the audio spectrum above 400 Hz. Fig, shows quantization noise in dB (y axis) against signal amplitude from maximum down to -90 dB (x axis).

INTERMODULATION DISTORTION

Intermodulation (IM) distortion measures the creation of new frequencies from a combination of any legitimate input frequencies. Some circuits generates a series of sum and difference frequencies related to two input frequencies. For example if 50 Hz and 1,000 Hz are input , 950 Hz, 1,050 Hz, 1,100 Hz and an entire series of frequencies might be added in Since these frequencies are not harmonically related to the originals they tend to be audible disagreeable and might contribute to listener fatigue. To measure IM, twin sine waves most often recorded at 0 dB are input to the device and then filtered at the output; any remaining intermodulation signals are measured and expressed as a percentage of the original signal. Several standard frequency pairs can be used. a SMPTE IM distortion CCIF IM distortion standard specifies 11 and 12 KHz in a 1:1 ratio. In either case the percentage of the beat frequency at the player's output constitutes the measurement. A low-pass filter is usually inserted as part of the measurement. A good player's SMPTE IM distortion should be 0.004 percent or less. A spectral analysis of the IM products may yield insight into their cause.

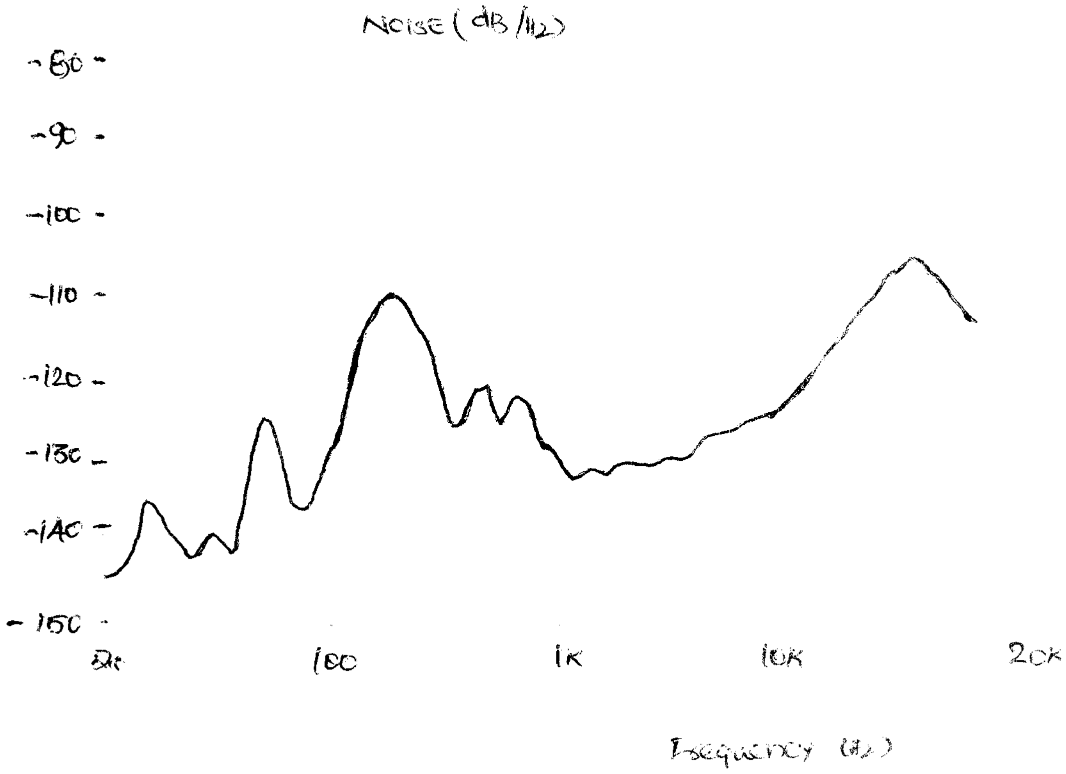
DISTORTION TEST FREQUENCIES

Distortion measurements in CD players point up a unique problem: Certain test frequencies must be avoided because they yield unrepresentative (and higher) distortion readings. Specifically frequencies which are integral submultiples of the 44.1 KHz sampling frequency-they are thus synchronous with the clock frequency- will utilize a fewer number of quantization intervals than noninteger submultiple,resulting in higher distortion. The synchronous relationship does not test linearity of every state of the D/A converter because the points where the test waveform are sampled will repeat after a relatively small number of samples. A non-synchronous relationship is preferred because the waveform passed through many timing relationship causing every converter state to be exercised. For example a test frequency of 315 Hz (1/140 of 44.1 KHz) recorded at -20 dB would use 71 different quantization intervals in a period of 100 milliseconds in a 16 bit converters whereas a 317 Hz signal of the same amplitude and over the same time peroid uses 3.282 intervals.

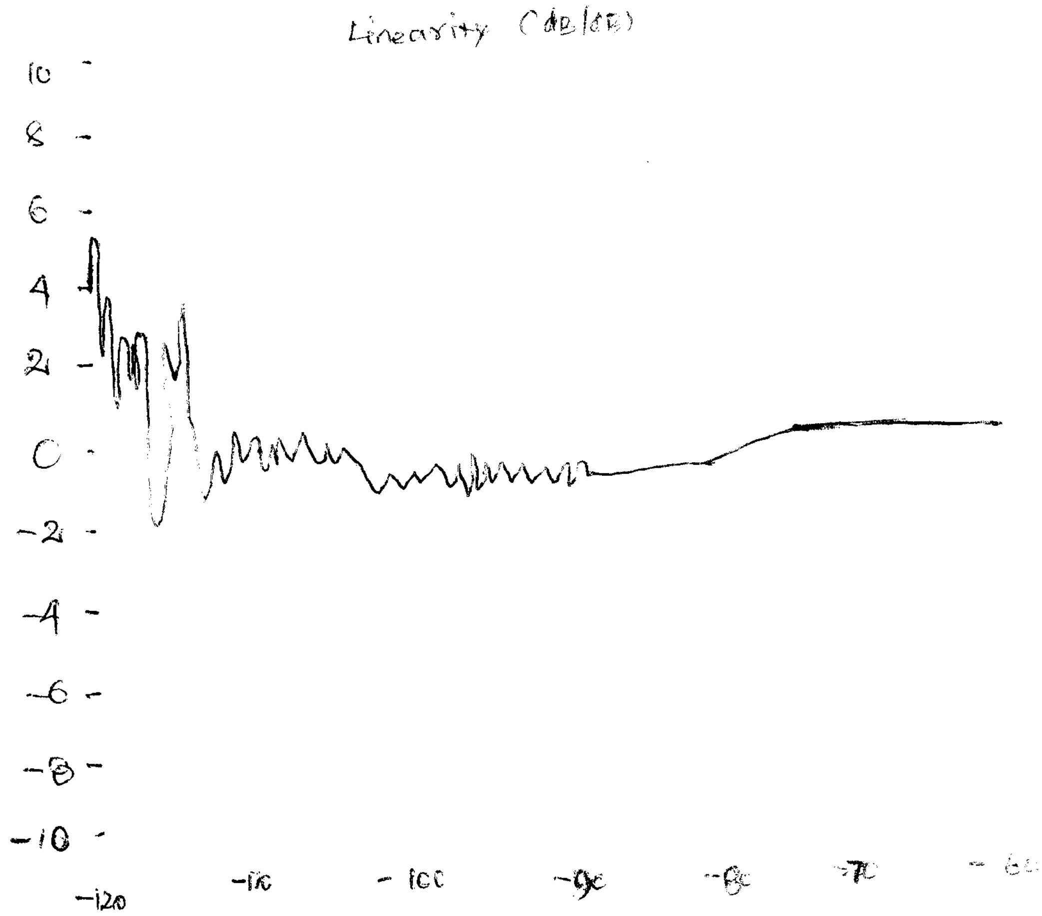
Thus the EIAJ and manufactures recommend avoiding test frequencies that are integer submultiples. Hence new test frequencies have been standardized by the EIAJ. These are presented in table 5.2 . Measurements of criteria other than distortion such as frequency response are not affected by the anomaly.

SIGNAL-TO NOISE RATIO

Ideally an device have no output signal when there is no input signal. In practice the S/N (signal-to-Noise) ratio measures the noise floor of the player's output relative to the player's rated maximum output level (0 dB). The number of bits in a digital audio system determines the maximum signal-to-noise ratio possible. but the analog output circuitry is often the limiting factor. Any noise or hum in the output stages will determine the floor over which the signal may range. To make the test, a test disc track with 16 zeros per sample (digital silence) is played and the noise which is measured is referenced to a maximum level of 0 dB. An S/N measurement with a digital salience track does not provide information on the player's digital circuitry since the D/A converter is not being exercised (and its



S/N RATIO (UNWEIGHTED) MEASUREMENT.

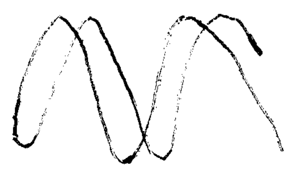


output is muted in some designs). The digital salience track is thus useful for separation analog noise from digital noise sources. Although the D/A converter is quiescent, and thus contributes no noise due to conversion low-frequency noise sources. such as the power supply, the analog circuitry and poor grounding as well as high-frequency noise sources such as clock leakage will limit the noise floor of the player. A spectral analysis of the noise signal is often revealing.

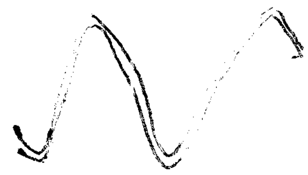
Once again there are different measuring methods. Some manufactures measure a broadband S/N ratio, while others employ a low-pass filter to eliminate out-of-band components. (The EIAJ calls for a filter with cutoff at 4 Hz and 20 KHz and with an attenuation of atleast 60 dB at 24.1 KHz). An "A" weighted filter may be used to attenuate low-frequency components of the noise signal. A good player should have an S/N of greater than 110 dB. Unweighed at 1 kHz as shown in fig . Careful attention to grounding and interconnections is required to measure such low levels.

DYNAMIC RANGE

Dynamic range is often confused with signal-to-noise ratio but there is a difference. The EIAJ defines dynamic range as the level of total harmonic distortion plus 60 dB when a CD reproduces a 1 KHz signal recorded at -60 dB below maximum level. For example a THD + N reading of -25 dB plus 60 dB yields a dynamic range measurements of 85 dB. The measurement is thus not

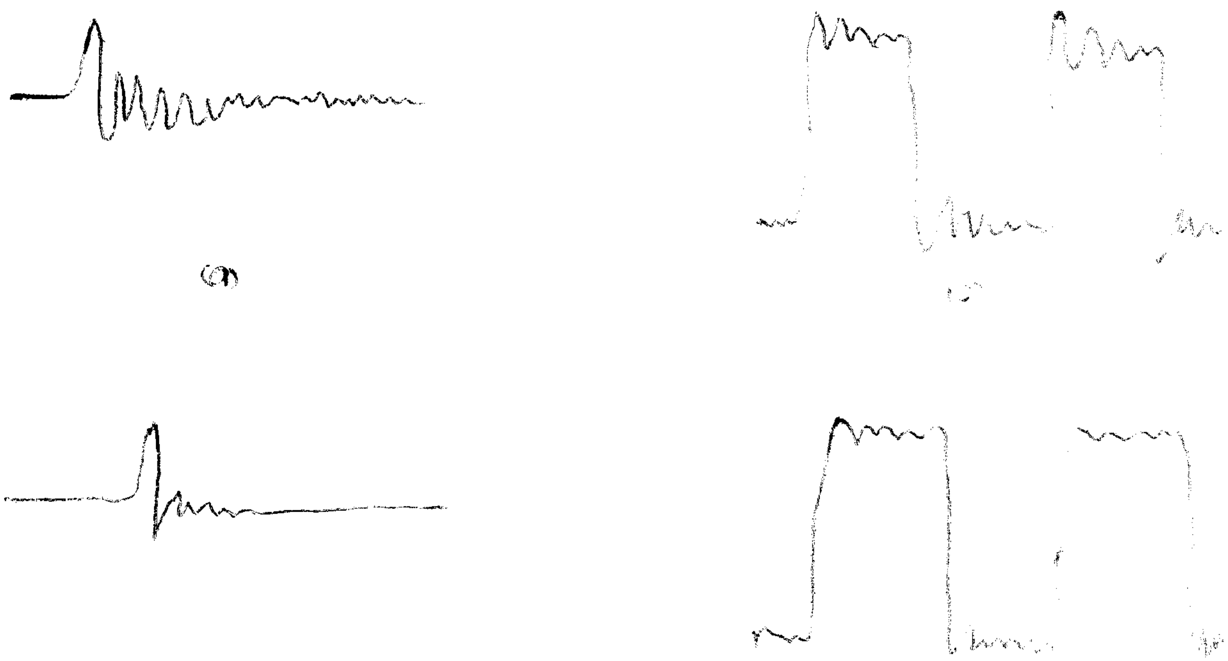


PHASE ERROR



PHASE LINEARITY

FIG: INTERCHANNEL PHASE DIFFERENCE CLOCKED AT 20 KHZ.



(a) (c) (b) (d)

a) ANALOG FILTERS IMPULSE RESPONSE
b) DIGITAL FILTERS IMPULSE RESPONSE

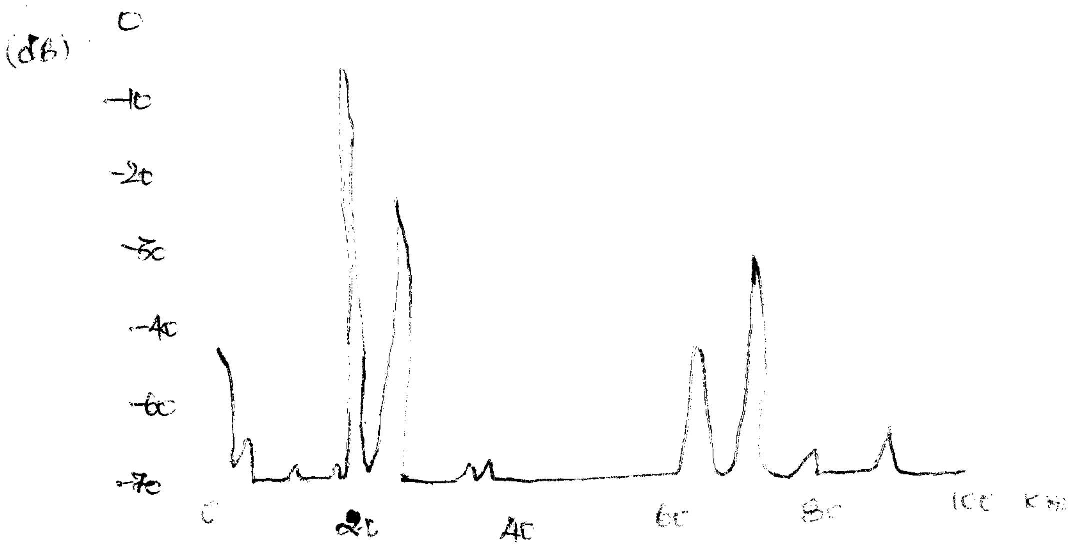
c) ANALOG FILTER 1KHZ SQUARE
d) DIGITAL FILTER 1KHZ SQUARE

FIG: O/P FILTERS CAN BE EXAMINED THROUGH SINGLE PULSE + SQUARE WAVE.

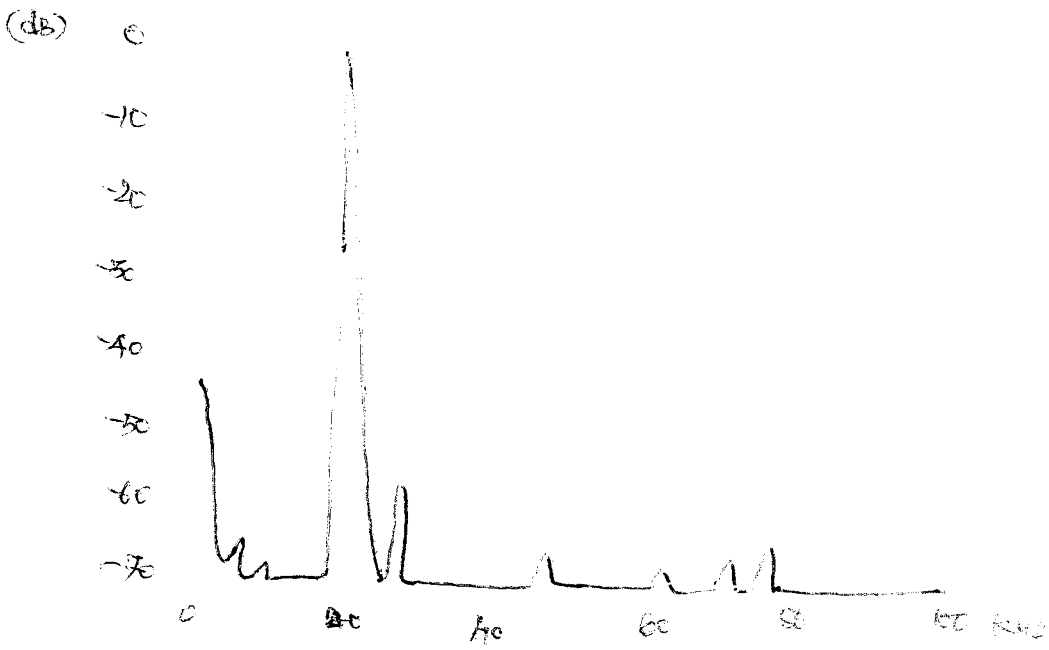
equivalent to S/N ratio. The EIAJ specifies that an "A"-weighted filter and 22 KHz low-pass filter be used to measure THD + N. This produces a noise measurement as opposed to a distortion measurement and the results can be compared to "A" -weighted noise measurements. However it may be misleading to measure only harmonic components for example. D/A non linearities may produce alias frequencies. A more rigorous test is to measure the total power of all harmonic, nonharmonic , and noise components relative to the fundamental. The EIA has proposed means of measuring dynamic range using 500 Hz dithered test signals recorded at -60 and -120 dB. The player's output is passed through a one-third octave filter tuned to 500Hz and the output is measured in dB. The noise floor is determined by adding 3 dB to the output signal level from the -120 dB recorded level. This signal level is assumed to be the lowest signal that can be reliably distingiushed from noise. Maximum signal level is thus 60dB above the output signal level of -60 dB. Dynamic range is ascertained by subtracting the noise floor (-120 dB recorded +3 dB) from the signal level (-60 dB recorded + 60 dB).

D/A LINEARITY MEASUREMENTS

As we observe in chapter four D/A counter linearity is a crucial concern in a digital audio system. No matter what word length is converted the accuracy of the conversion and hence the fidelity of the audio signal hinge on the linearity of the converter. A linearity test measures the D/A converter's ability to reproduce various signals, particularly low amplitude signals



a) UN FILTERED SPECTRA



b) FILTERED SPECTRA.

FIG: EFFECTS OF OUTPUT FILTERS USED IN MAKING MEASUREMENTS.

apparent as a deviation from a nominal amplitude and will result in distortion of the audio signal. Multibit converters exhibit differential non-linearity due to bit weighing errors; thermal or physical stress, aging and temperature variation are also factors. Low-bit converters are more robust in this regard.

A linearity test measures the reproduced analog signals relative to their recorded amplitudes. The test is often performed with 500 Hz test tones, with descending test disc amplitudes of -1, -3, -6, -10, -20, -30, -40, -50, -60, -70, -80, -90 dB and -100 dB with the greatest scrutiny in which the tone fades approximately -60 dB to -120 dB where the signal disappears into system noise. It is important to note the difference between dithered and undithered test tones when making this test. For example, because of averaging an undithered test tone of -90 dB will be reproduced at -89 dB through a perfect D/A. When a dithered test is used, the fundamental level as obtained from a spectrum analyzer should fall preciously at -90 dB. Careful attention to grounding and interconnection is required to measure such low levels. In some cases, D/A converter nonlinearities can be uncovered by ear. This is done by playing a dithered fade-to-zero signal. Effects of nonlinearity are audible as harmonic distortion.

Using tests for D/A linearity, the results of poorly performing D/A converters can be evaluated. For example, a -90 dB signal reproduced at -85 dB would result in expansion of low-amplitude information. A -80 dB signal reproduced as -76 dB would result in compression of low level information. A D/A converter with extreme nonlinearity problems might compress the signal through -80 dB, then expand the signal at -90 dB, reproducing low-level audio waveforms in a highly nonlinear fashion. Ambient information, for example, could be audibly affected. Figure 5.10 shows the output of a CD player in which the D/A output is quite linear through -90 dB as measured with a dithered fade-to-zero linearity test. It would be safe to say that this nonlinearity would not be audible under most listening conditions.

To help equipment manufacturers use D/A converters to their best advantage, many multibit D/A chips provide a means of calibration. Some D/A chips offer calibration of the MSB, while others may offer calibration of the 4 most significant bits. As a practical tip, remember to let the D/A converter warm up for a half-hour or more before attempting any calibration; its linearity changes as the device reaches a stable operating temperature.

PHASE MEASUREMENTS

Interchannel phase difference measures the shift between identical frequencies recorded on two channels. The stereo signal must be recovered from the single data stream and reconstructed with the original simultaneous phase between the two channels. If a single D/A converter is used and if compensation isn't applied, the channels will be out of phase with respect to each other as the converter delays one channel relative to the other. For example, a 44.1 kHz clocked player with a single D/A may show an 82 degree phase shift at 20 kHz (11.34 microseconds at the 44.1 kHz clock rate). Similarly, an 88.1 kHz clocked player may show a 41 degree shift and a 176.4 kHz clocked player would show about a 20 degree phase shift. Time delay compensation used in single D/A players may improve this phase shift. Even dual D/As can contribute a slight error.

To measure any interchannel phase error, 20 kHz is output to each channel; any difference will be apparent from the misaligned waveforms such as the ones in figure A good player should have negligible interchannel phase error. This is illustrated in Figure . Alternatively, time delay or phase error between channels can be measured through a Lissajous pattern. The output of one channel is connected to the horizontal (X) input of an oscilloscope, and the output of the other channel is connected to the vertical (Y) input. A trace angled at 45 degrees (from lower left to upper right) indicates the channels are in phase.

Phase linearity measures any phase shift between different frequencies. Digital filters and carefully compensated analog filters can reduce phase error to a few degrees or less. Although this is not a rigorous test, some test discs contain twin tones of 2 kHz and 20 kHz, each on a single channel output. The tones are encoded with simultaneous positive-going axis crossings. Any relative phase discrepancy can thus be observed on an oscilloscope. Simultaneous zero-crossing of the positive-going waveforms shows there is no phase discrepancy. An offset of the waveforms at that point would indicate a discrepancy. A CD player with digital filtering might show phase linearity of less than 0.1 degrees at 20 kHz.

DE-EMPHASIS

To reduce noise, some compact discs are recorded with pre emphasis, a boosting of high frequencies as described in Chapter Three. During playback, a high frequency de-emphasis circuit should automatically restore a flat response. De-emphasis error measures any deviation occurring when the de-emphasis curve is not the correct reciprocal of the pre-emphasis curve. To accomplish the test, selected frequencies are output, and any de-emphasis deviation is measured in dB. The EIAJ proposes use of 125 Hz, 1 kHz, 4 kHz, 10 kHz and 16 kHz.

Other Evaluations

Any level imbalance between the two output channels is measured by comparing two 1 kHz signals recorded at 0 dB. The difference is expressed in dB. In addition, voltage readings of the maximum output values are often specified.

Although the wow and flutter in CD players should be negligible, some test discs supply a 3.15 kHz tone, the frequency usually used to measure speed fluctuation in analog equipment to accomplish the measurement for CD players. The EIAJ specifies that the test be made for at least 5 but no more than 30 seconds. Typically, the weighted peak value of CD wow and flutter is below the ± 0.001 percent measurement threshold, and most manufacturers simply state, correctly, that it is below measurable limits.

Although an unlikely occurrence, any pitch error introduced by a CD player may be specified by measuring a 20 kHz output signal with a frequency counter and expressing the error as a percentage : $(F_1 - F_0)/(F_0)x100$ where F_0=20 kHz.

The nature of the output filter can be examined through single pulses and square waves. A single pulse or a 1 kHz square wave will show the damped pattern of analog filtering or the symmetrical pattern of digital filtering as shown in figure 5.12. However, it is important not to misinterpret the ripple present on the waveform. A perfectly square wave could only exist in the case of infinite bandwidth - that is, only if an infinite number of sine wave harmonics were present to completely compose the square wave. In the case of a CD player, we know that our band-width is limited to about 20 kHz. Thus the square wave will show ripples. This is normal operation and is described as the Gibbs phenomenon.

The single pulse response is also useful to uncover incorrect absolute polarity. For example, if a positive-going pulse recorded on the test disc results in an oscilloscope trace showing a negative signal, the player's absolute polarity is reversed.

The EIAJ also specifies a test to measure access time, the time it takes for a pick up to move to a location on the disc. Short and long access time s are specified. Short access time measures the time for the pickup to move the next adjacent track and begin playing. Long access time measures the time for the pickup to move from the innermost track to the outermost track and begin playing. Both measurements are specified in seconds.

Tests discs can be used to exercise the error correction/tracking ability of the player. The philips 5A test disc contains three types of intentional defects on specified tracks : simulated fingerprint, calibrated black dots on the readout side, and calibrated interruptions on the information layer. The dimensions of the black dots are 300, 500, 600 and 800 micrometers, and the interruptions on the information layer are 400, 500, 600, 700, 800 and 900 micrometers. These obstacles help determine a player's limitations in error correction, concealment and tracking. All CD players should be able to play through the fingerprint and the smallest of the other defects. In theory a player with fully implemented error correction strategy (not all have this) should correct any defect up to 2400 micrometers. In practice most players handle all the defects on the Philips test-disc.

The more strenuous defects in the Verany 788032 disc may be used to determine the player's limits. This disc contains seven test conditions designed to evaluate the mechanical digital, and processing performance of a CD player. The first series of tracks is recorded with linear velocity ranging from 1.2 to 1.4 meters/second. The second series of tracks combines maximum and minimum cutting velocities with maximum and minimum track pitches. One track in the test is recorded with a continuous velocity variation. The third series of tracks contains variations in track pitch ranging from 1.5 to 1.7 micrometers. The fourth series of tests can be used to measure the detection level of the RF signal from a pickup by simulating variations in cutting intensity. Asymmetry is created by varying the ratio of pit/land amplitude. The fifth series of tests evaluates a pickup's dropout performance. Non data areas ranging from 0.05 millimeters to 4.0 millimeters are traced across the disc surface. The CD standard specifies correction of 0.2 millimeter dropouts, but the CIRC is theoretically capable of correcting dropouts of up to 2.47 millimeters. Interpolation of 8.5 millimeters dropouts is theoretically possible. The sixth series of tests combines dropouts with minimum track pitch. Dropouts ranging from 1 to 2.4 millimeters are recorded at a pitch of 1.5 micrometers. The seventh test can be used to evaluate performance in the face of successive dropouts. Some players may be able to correct a single [illegible]

Some players may be able to correct a single dropout but fail to correct two dropouts. The test contains two successive dropouts, each ranging from 0.1 to 3 millimeters. Because many of these test conditions exceed the expectations of the CD standard, error correction and tracking performance may be critically evaluated.

CAVEATS

The use of low-pass filters to limit measurements to in-band information has generated confusion. Some players output only in-band information with no spurious high-frequency components. Specifications such as THD + N can be measured directly. However, other players exhibit out-of-band frequency components. The rationale for bandleader measurements is that supersonic components cannot be heard and should not therefore affect the measurement.

The difference between unfiltered and filtered measurements is illustrated in figure 5.13. The first oscilloscope trace of the unfiltered player under test shows a 20 kHz signal and an image frequency at 24.1 kHz (44.1 kHz-20 kHz) as well as images of those two frequencies at 64.1 kHz and 68.2 kHz. The second trace shows what happens when a 20 kHz seventh order low-pass filter is put on the player's output; the troublesome images are largely attenuated, leading to much better measurements.

We should note that some critics are not so sure of the validity of filtered measurements. Since many players measure satisfactorily without a special brick wall filter, is it fair to use special measuring conditions for the benefit of other players ? Moreover, it is possible that some listeners can hear out-of-band components or that the components might affect inland signals. If a player has them, measurements should show the fact. Finally, unless the manufacturer wishes to supply the consumer with a special low-pass filter, it doesn't seem right to measure the player with one.

The bottom line is that measurements are valid only if they accurately portray performance. If some players have higher THD + N, then that should be made clear. If THD + N measurements are made with a brickwall filter, then a spectrum analysis should also be supplied with the player to show the supersonic components. Honesty in specification is important, especially in a new technology such as digital audio. Only then can we expect evolution toward better performance.

A specification is meaningful only if we understand the conditions of measuremnt. Greater consumer awareness will lead the way to new specifications designed to clarify the performance of digital audio devices and to help supplement the pair of precision measuring devices you's naturally equipped with.

How to Enjoy our 'CD' System

GUIDELINES FOR INSTALLATION AND MAINTENANCE

This unit has been manufactured to assure your personal safety. But improper use can result in potential electric shock or fire hazard. In order not to defeat the safeguards. please observe the following.

* Do not place this equipment where the power cord will be damaged by persons walking on or tripping over it.
* Do not overload wall outlets and extension boards ad this can result in fire of electric shock.
* Before cleaning, unplug the equipment from the mains outlet. Use only damp cloth for cleaning. Do not apply liquid or aerosol cleaners directly on it.
* Remove power plug from mains socket when the set is not on use for a long time.
* Do not touch mains cable with wet hands. While unplugging always pull the plug and not the cord.
* Keep away from heat and moisture.
* Removal of the cabinet cover may expose you to hazardous voltages or radiations.
* Sudden fall in surrounding temperature may cause dew to form on the optical lens inside the set. In this condition the laser cannot pick up signal and the unit will not operate properly. If this should happen, do not operate the player until it has adjusted to the surrounding temperature.

CONTROLS ([redacted])

1.Power switch (POWER)

2.Disc tray

3.Track display (TRACK)

4.Play indicator (PLAY)

5.Repeat indicator (REPEAT)

6.Programmer indicator (PROGRAM)

7.Play/Pause switch (PALM/PAUSE)

8.Stop switch (STOP)

9.Skip /Search switches (SKIP/SEARCH)

10.Repeat switch (REPEAT)

11.Memory switch (MEMORY)

12.Open and close switch for the disc tray (OPEN/CLOSE)

13.System remote socket (SYSTEM REMOTE)

14.Output sockets (OUTPUT)

15.Mains lead

CONNECTIONS

-Before making connections or disconnections be sure to turn off the power to each component.

-Connect one end of the Phono plug terminated leads (Supplied) to the OUTPUT sockets of the unit and the other end to the input sockets designated as CD (or AUX, TAPE PLAY, T.V, TUNER etc.) of any other stereo system or amplifier. Do not connect to PHONO sockets.

-For remote control function (only when the unit is used with the system) connect one end of the connection lead (supplied with the stereo system) to the SYSTEM REMOTE socket of the unit and the other end of the CD PLAYER socket of the stereo system.

-After all connections are completed, connect the mains lead to an AC wall outlet.

-If the CD-Player is placed too close to the tuner, some interference in AM of FM reception may occur. In such a case, we recommend to simply switch off the CD-Player during tuner operation.

PREPARATION FOR OPERATION

Both 12cm and 8 cm compact discs can be played on this player.

1. Press the POWER switch to ON. The display indicates "CD".

2. Press the OPEN/CLOSE switch. The disc tray will come out and the display indicates "- -".

3. Without touching the disc surface, place the disc in the tray with the label facing upward (Fig.).

4. Press the OPEN/CLOSE switch once again. The disc tray will close. The display indicates the total number of tracks.

Note:

-When the disc is inserted upside down (label downwards) or no disc is inserted, the player will not operate.

-Only discs with the symbol shown below can be played on this unit.

WHEN HANDLING COMPACT DISCS

In order to enjoy your music under the best conditions, always keep your discs clean and note the following.

-Do not touch the playing surface (shiny, rainbow-colourated surface).

-When you will not be playing a disc of high temperature of high humidity where they may become warped.

-When you will not be playing a disc for long periods, remove it from the player and return it to its case.

-Do not put tape, etc, on the label surface.

-Before playing a disc, wipe any dust or fingerprints from the playing surface with a soft cloth.

-Do not use cleaners or antistatic sprays used on conventional analogue phonograph records.

OPERATION

NORMAL PLAY

1. Press the PLAY/PAUSE switch. The PLAY indicator illuminates and the first selection (TRACK No.1) will begin playing.

2. Press the STOP switch to stop at anytime.

3. When the last track has played, the player will go to stop mode.

4. When play is finished, open the disc tray, remove the disc, close the disc tray and press the POWER switch once again to turn off the power.

the disc tray will come out.

PAUSE

When the PLAY/PAUSE switch is pressed during play, the player will go to pause mode. (The PLAY indicator will flash). in order to cancel pause, press the PLAY/PAUSE switch once again and play will begin from where it had stopped.

TRACK SEARCH

When the ⏭/⏩ or ⏪/⏮ SKIP / SEARCH switches is pressed during play, the player will quickly advance or reverse while the switch is being pressed. When the switch is released, play will begin again at that point.

-If track search is performed during pause, the player will return to pause when that switch is released.

-During track search operation, the display includes the elapsed playing time of the track which is searching. The left-hand digit indicates the last place of minutes and the right-hand digit indicates ten's place of seconds.
eg., 15:30 = 5.3, 1:32 = 1.3

TRACK SKIP

By using the skip mode during play, the beginning of desired tracks can easily be found. The number of the track to be searched is displayed on the display.

Press the ⏭/⏩ or ⏪/⏮ SKIP/SEARCH switch with one touch (or if required, repeatedly) until the desired track No is displayed. The player will start playing from the desired track.

-The track No. will change every time by pressing the SKIP/SEARCH switch.

-If in pause before pressing the SKIP/SEARCH switch, the player will go to pause (standby) after beginning of the track is located.

PROGRAMMED PLAY

1.Press the STOP switch to prepare for the programming.

2. Press the ⏭/⏩ or ⏪/⏪ SKIP/SEARCH switch to select the first program (to be played first). The selected track No.

3. Press the MEMORY switch to store the information. The stored track No. will display and PROGRAM indicator will illuminate.

4. Press the ⏭/⏩ or ⏪/⏮ switch to store the second program .

5. Press the remaining tracks available on the disc in the same way.

Note:

Upto 16 selections can be programmed.

7. Press the MEMORY switch to confirm programming.

8. Press the PLAY/PAUSE switch. Play will begin from the programmed track.

9. When the last programmed track has played. The player will go to stop mode.

The programmed memories will remain.

Note:

-If your want to add more programmer numbers to the next free memory position after confirmation (Step 7), selected the track number and press the MEMORY switch.

It is impossible to change the programmed entry or to add a program between the previous programmer.

-In order to cancel (clear) an entire program press the STOP switch twice or press the OPEN/CLOSE switch.

When the unit is turned off, the entire program is canceled.

REPEATED PLAY

When the REPEAT switch is pressed during play the REPEAT indicator will illuminate and the player will repeat the tracks continuously.

To cancel repeated play, press the REPEAT switch once again. (The REPEAT indicator will go out). When the STOP or OPEN/CANCEL switch is pressed, the repeated play will be canceled.

SPECIFICATIONS

Audio

Channels	2 channels
Frequency response	5-20,000 Hz
S/N RATIO	98 dB
Channel separation	90 dB (1 KHz)
Distortion	0.03% (1 KHz)
Wow and flutter	Undetectable
Outputs	1.8 V/1K ohms

Signal format

Sampling frequency	44.1 KHz
Quantization	16 bits linear/channel
Transmission rate	4.32 Mbit/sec.

Pickup

System	Light pickup (3-beam)
Light source	Semiconductor laser
Wave length	790 nm

General

Power supply	AC: 210-240v. 50/60 Hz
Dimensions (approx)	360 (W) x 89 (H) x 280 (D) mm

Further Enhancements

FURTHER ENCHANCEMENTS :

CDs are an optical disc system dedicated to reproducing music for enjoyment. Since a vastly larger amount of data than that which could be recorded on the media used upto now can be recorded on the disc, it is possible to consider it for use as a medium for recording other kinds of information such as video images and data.

The project we have completed be further enhanced by implementing the following latest features in the field of digital compact disc systems.

(i) Read / Write CDs

Research in this field has made possible the design of R/W CD. CD RAM's are mow possible and very attractive products can appear in the near future.

(ii) CD Video

Another attractive feature of CD's is CD-Video. CD-Viedo is a hybrid of compact disc and video disc in function as well as in appearance.

CD-V has inherited the best of both technologies. CD's digital sound along with video disc picture.

(iii) MULTI DISC PLAYERS

Due to advancement is CD technology multidisc compact disc players, five to ten or more this could be loaded at a time and programmed for playing as required.

(iv) Palm Size CD's

New technology has miniaturised CD audio machines for the home, car and portable listening.

The above features are the latest new developments in the fast emerging field of CD's.

Conclusion

CONCLUSION

Many existing systems employ only one D/A/ converter the multiplexed data of the two audio channels is handled by one D/A. Only after the D/A has performed its conversion are the two analog channels separated into left and right channels.

(i) This single D/A Design creates several anomalies in the signal. The interleaved segments of analog voltages produced from single D/A Design are not so easily separated. A switch must be used to route these pieces of waveform to the correct output channel. The switch might cause distortion as it flip back and forth.

(ii) The single D/A design introduces a slight time difference (11.34 micro seconds) between the two audio channels. So the interchannel difference can occurs stereo imaging.

These drawbacks are overcome in Dual D/A converter. In a dual D/A converters design, the demultiplexed data appears at the two converters simultaneously, exactly as it appeared at the A/D converters of the digital recorder. The O/P analog channels are thus exactly in phase that is exactly matched in time. It is possible to compensate for the offset introduced by a single D/A However, this is difficult to achieve uniformely over the entire frequency range.

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