chapter - 5

DATA COMPRESSION

Data compression is the process of converting an input data stream or the source stream or the original raw data into another data stream that has a smaller size. For example: text compression, image compression, audio compression and video compression.

Why do we need to compress?

→ To reduce the volume of data to be stored.

→ To reduce the bandwidth required for transmission.

There are two types of data compression

a. Lossy Compression

b. Lossless Compression

a. Lossy Compression

Lossy compression algorithms is normally not to reproduce an exact copy of the source information after decompression but rather a version of it which is perceived by the recipient as a true copy.

In Lossy compression, some information is lost during the processing, where the image data is stored into important and unimportant data. The system then discards the unimportant data.

It provides much higher compression rates but there will be some loss of information compared to the original source file. The main advantage is that the loss cannot be visible to eye or it is visually lossless.

Visually lossless compression is based on knowledge about color images and human perception.

b. Lossless compression

In this type of compression no information is lost during the compression and the decompression process. Here the reconstructed image is mathematically and visually identical to the original one. It achieves only about a 2:1 compression ratio.

This type of compression technique looks for patterns in strings of bits and then expresses them more concisely. Lossless compression algorithm the aim is to be transmitted in such a way that, when the compressed information is decompressed, therefore is no loss information.

Difference between Lossy and lossless compression

	Lossy Compression	Lossless compression
i.	This technique result in some loss of information.	No amount of information is lost.
ii.	A file does not restore or rebuilt in its original form.	A file can be restored in its original form.
iii.	Data's quality is compromised.	Does not compromise the data's quality

iv.	Lossy compression reduces the size of data.	But lossless compression does not reduce the size of data.
v.	Algorithms used are: Transform coding, Discrete ~~Tran~~ Cosine Transform, Discrete Wavelet transform etc.	Algorithms used are: Run Length Encoding, Huffman coding, Arithmetic coding, etc.
vi.	Used in Images, audio, video.	Used in text, images, sound.
vii.	has more data-holding capacity.	has less data-holding capacity than lossy compression technique.
viii	It is also termed as Prreversible compression.	It is also termed as reversible compression.
ix.	Compressed data are distorted.	They are mostly distortion less.
x.	Comparitively less.	Comparitively expensive.

Coding Requirements

→ Image have higher storage requirements than texts.
→ Audio/video have even more demanding storage requirements.

For example:

→ An uncompressed audio signal of telephone quality is sampled at 8KHz and quantized with 8 bits per sample. This gives $8 \times 8 = 64$ kbits to store 1 second of playback.

→ An uncompressed stereo signal audio CD-quality sampled at 44.1KHz with 16-bits per sampled (quantization) would require 705,600 bits to store one second of playback.

→ For PAL standard, a video defined by 625 lines and 25 frames per second, the luminance (related to brightness) and color difference are encoded seperately.

→ Sampling 13.5MHz is used for luminance (Y) and sampling rate for chrominance (R-Y and B-Y) is 6.75MHz.

→ If the result is a uniform 8-bit coding of each sample, the bandwidth requirement is:
$(13.5\,MHz + 6.75\,MHz + 6.75\,MHz) \times 8\,bit$
$= 216 \times 10^6$ bits/second

→ For HDTV, it is more than double in size & bandwidth

Compression in multimedia system is subjected to:

→ The quality of the coded and decoded data should be as good as possible.

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→ The complexity of the technique used should be minimal to avoid high cost.
→ The processing of compression algorithm must not exceed certain time spans.

In multimedia systems, compression can be categorized as:
a. Entropy Encoding
b. Source Encoding
c. Hybrid Encoding

a. Entropy Encoding
- Data stream is considered to be simple digital sequence without semantics
- Lossless coding, decompression process regenerates the data completely
- Used regardless of the media's specific characteristics
- Example: Run-length encoding, Huffman encoding, Arithmetic encoding

b. Source Encoding
- Semantics of the data are taken into account
- Lossy coding (encoded data are not identical with original data)
- Degree of compression depends on the data contents
- Examples: Content prediction techniques; eg. use of spatial redundancies between still images for data compression.
Discrete Cosine Transformation (DCT) as transformation technique of the spatial domain into the two-dimensional frequency domain.

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c. Hybrid encoding
- Used by most multimedia systems
- Combination of entropy and source encoding
- Examples: JPEG, MPEG, H.261, DVI

Major steps of Data Compression / Image Compression techniques

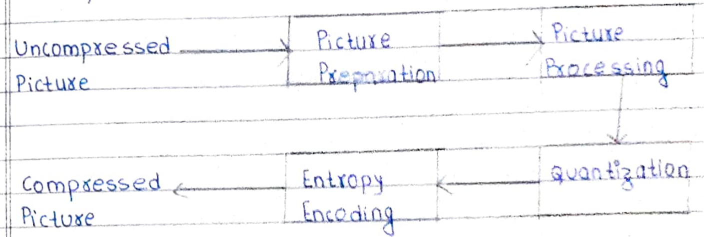

Fig: Major steps of Data Compression

a. Picture Preparation
Preparation includes analog to digital conversion and generating an appropriate digital representation of the information. An image is divided into blocks of 8x8 pixels, and represented by a fixed number of bits per pixel

b. Picture Processing
Processing is actually the first step of the compression process which makes use of sophisticated algorithms. A transformation from the time to the frequency domain

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can be performed using DCT. In the case of motion video compression, interframe coding uses a motion vector for each 8x8 blocks. Motion video computation for digital video.

c. Quantization

Quantization process the results of the previous step. It specifies the granularity of the mapping of real numbers into integers. This process results in an reduction of precision. For example, they could be quantized using a different number of bits per coefficient. For example 12 bits for real values, 8 bits for integer value.

d. Entropy Encoding

Entropy encoding is usually the last step. It compresses a sequential digit data stream without loss. For example, a sequence of zeroes in a data stream can be compressed by specifying the number of occurrences followed by the zero itself.

→ The processing and quantization can be repeated iteratively several times in feedback loop.

→ The term spatial domain (time domain) refer to the image plane itself and approaches in this category are based on discrete manipulation of pixel in an image.

→ After compression, the compressed video stream contains the specification of the image starting point and an identification of the compression technique may be the part of the data stream. The error correction code may also be added to the stream.

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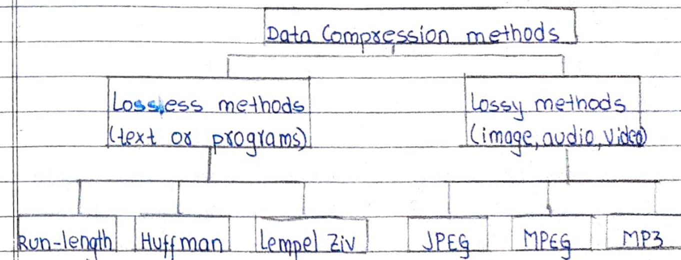
→ Decompression is the inverse process of compression.



Run-Length

Run length encoding is a lossless data compression algorithm. It compresses data by reducing repetitive, and consecutive data called runs. It does so by sorting the number of these runs followed by the data.

The basic idea behind this approach to data compression is this: if a data item occurs n consecutive times in the input stream replace the n occurrences with a single pair. The n consecutive occurrences of a data item are called run length of n and this approach is called run length encoding or RLE.

Example:

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Original data : BBBBBBBBBAAAAAAAAAAAAAAAAN
MMMMMMMMMM

Compressed data : B09 A16 N01 M10 OR
(9, B); (16, A); (1, N); (10, M)

This encoding can have companion as well as negative compression.

Huffman

Huffman coding is a lossless data compression algorithm. The idea is to assign variable-length codes to input characters, lengths of the assigned codes are based on the frequencies of corresponding characters. It is based in the concept that the probability of occurrence of the characters is not same ~~or~~ so different number of bits is assigned for different character.

Basically in variable length coding the characters that occur most frequently are assigned fewer numbers of bits. However in order to used variable length coding the destination must know the set of code-words being used by the source. In Huffman coding the probability of occurrence of the characters are estimated and based on this estimation code-words are assigned to the characters.

Example :
Design Huffman code for probabilities :
0.25, 0.25, 0.125, 0.125, 0.0625, 0.0625

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Symbol	Probability					
S_1	0.25 10	0.25 10	0.25 01	0.25 00	0.5 1	0.5 0
S_2	0.25 11	0.25 11	0.25 10	0.25 01	0.25 00	0.5 1
S_3	0.125 001	0.125 000	0.25 11	0.25 10	0.25 01	
S_4	0.125 010	0.125 001	0.125 000	0.25 11		
S_5	0.125 011	0.125 010	0.125 001			
S_6	0.0625 0000	0.125 011				
S_7	0.0625 0001					

Symbol	Probability	Code	Codelength
S_1	0.25	10	2
S_2	0.25	11	2
S_3	0.125	001	3
S_4	0.125	010	3
S_5	0.125	011	3
S_6	0.0625	0000	4
S_7	0.0625	0001	4

$$L = \sum_{K=1}^{K} P_k L_k$$

$= (0.25*2) + (0.25*2) + (0.125*3) + (0.125*3) + (0.125*3)$
$+ (0.0625*4) + (0.0625*4)$
$= 2.625$ bits per symbol.

[Q] Construct a Huffman coding for the following grey levels.

Grey level	0	1	2	3	4	5	6	7
No: of pixels	30	35	38	10	15	10	38	80

→

Sol^n:

Grey level	No. of pixel	Probability
0	30	$30/256 = 0.11$
1	35	$35/256 = 0.13$
2	38	$38/256 = 0.14$
3	10	$10/256 = 0.039$
4	15	$15/256 = 0.058$
5	10	$10/256 = 0.039$
6	38	$38/256 = 0.14$
7	80	$80/256 = 0.312$
	$\Sigma(x) = 256$	

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LZ77 (sliding Window)

LZ77 and LZ78 are the names for the two lossless data compression algorithms published in papers by Abraham Lempel and Jacob Ziv in 1977 and 1978. They are also known as LZ1 and LZ2 respectively. They are both dictionary coders. LZ77 is the sliding Window compression algorithm.

The main idea of this method is to use part previously seen input stream as the dictionary. The encoder maintains a window to the input stream and shifts the input in that window from right to left as strings of symbols are being encoded. The method is thus based on 'sliding Window'. The window is divided into two parts that is search buffer, which is the current dictionary and lookahed buffer, containing text yet to be encoded.

Arithmetic coding

In this method the input stream is read symbol and appends more to the code each same time a symbol is input and processed. To understand this method it is useful to imagine resulting code as a number in the range [0,1] that is the range of real numbers from 0 to 1 not including one. The first step is to calculate or at least to estimate the frequency of occurrence of each symbol.

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JPEG (Joint Photographic Expert Group)

JPEG is a commonly used method of lossy compression for digital photography (image). The JPEG lossy compression scheme is one of the most popular and versatile compression schemes in widespread use. It's ability to attain considerable size reductions with minimal visual impact with relative light computational requirements and the ability to fine tune the compression level to suit the image at hand has made it the standard for continuous tone still images.

Why JPEG? (Requirements of JPEG standard)

- The JPEG implementation should be independent of image size and should be applicable to any image and pixel aspect ratio.
- JPEG can be applied to color as well as grey scale image.
- It can handle image content of any complexity, with any statistical characteristics.
- The JPEG standard specification should be state of art (or near) regarding the compression factor and achieved image quality.
- The software should run on many available standard processor as possible.
- Sequential decoding and progressive decoding are possible.
- The user can select the quality of the reproduced image the compression processing time and size of complexed image by choosing appropriate parameters.

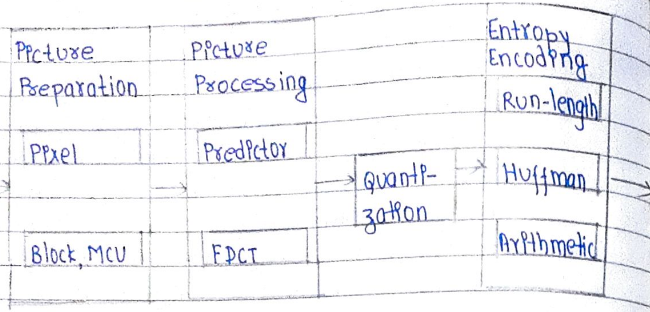


Fig: steps of the JPEG compression Process

a. Image/Picture Preparation
- Analog to digital conversion
- Generation of Appropriate digital representation
- Division of image into 8x8 blocks
- fixing the number of bits/pixel.

Let us suppose an image having the pixels as:

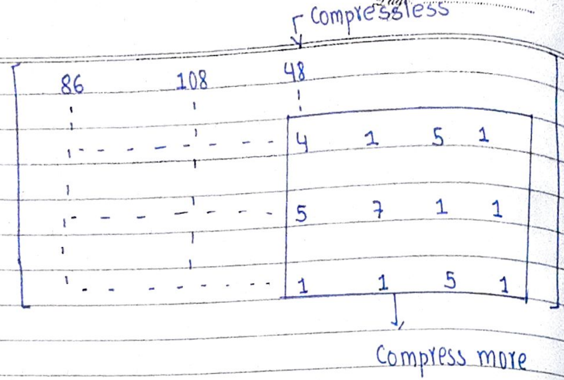
$$\begin{bmatrix} 101 & \cdots & 130 & \cdots & 133 \\ \vdots & & \vdots & & \vdots \\ 78 & \cdots & 149 & \cdots & 106 \\ \vdots & & \vdots & & \vdots \\ 99 & \cdots & 40 & \cdots & 129 \end{bmatrix}$$

Typical image are represented as the matrix shown above where;
larger pixel value = brighter pixel
Smallest pixel value = darker pixel

→ The first step is the color-space conversion.
→ Instead of representing the image with its RGB color components, they are converted into color space where one channel represents the light intensity (Y) and other two channel represents the chroma components i.e. C_b and C_r
where, C_b = color Blue
C_r = Color Red
The conversion seperates the luminance from the chrominance.
→ Since our visual system is much more sensitive to the changes in brightness than changes in color, we can safely down size the chroma components to save some space. This process is called chroma sub-sampling. In other words, the low contrast high frequency components are barely visible to our eye.

b. Image/Picture Processing
- JPEG does the compression by dividing the image into 8x8 blocks and quantizing them into its frequency domain.
- This is done by comparing each one of those blocks into 64 frequency patterns.
- This process decompasses the image into its frequency components.
- Converting the 8x8 blocks where each element represents the brightness level into other 8x8 block where each element represents the presence of particular frequency.
- This method is called DCT (Discrete Cosine Transform)

Example:



c. Quantization

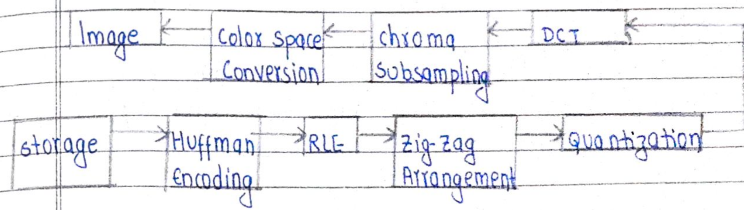
- It simply means rounding the result into its nearest integer. This results in higher compression ratio but lower image quality.
- After quantization we end up with a lot of zeros in higher frequency region.
- Then we apply the zigzag run to store the information more efficiently.

Example:

$$\begin{bmatrix} 5 & 1 & 2 & 1 & 1 & 0 & 0 & 0 \\ 3 & 1 & 2 & 5 & 0 & 1 & 0 & 0 \\ 2 & 1 & 2 & 0 & 0 & 1 & 0 & 0 \\ 3 & 1 & 0 & 0 & 0 & 0 & 0 & 0 \\ 1 & 1 & 0 & 1 & 0 & 0 & 0 & 0 \\ 1 & 1 & 0 & 0 & 0 & 0 & 0 & 0 \\ 1 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \end{bmatrix}$$

- If we rearrange these coefficients in zigzag order from left to right, we can group these zeros together
i.e. example [...... 0000000 ....]
repeat [0, 7]

- This technique is called RLE where no. of repetition of certain values are coded.
- Finally we can compress even more by encoding the more frequent values with fewer bits and less frequent values with higher bit. This is called Huffman Encoding.
when its time to decode the image, all the steps are reversed as:



Lossy Sequential DCT-based mode of JPEG

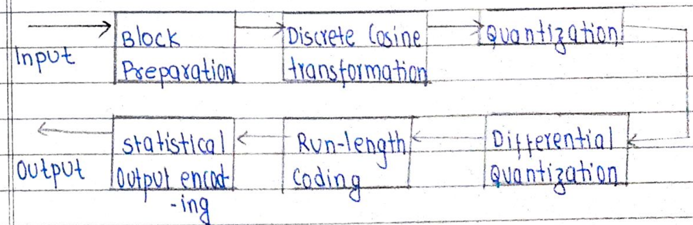


Fig: The operation of JPEG in lossy-sequential mode.

a. Image Processing

It basically involves the block preparation where the image samples are grouped into 8*8 pixels and passed to the encoder.

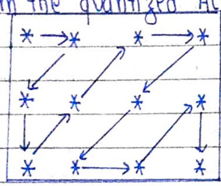
Then Discrete Cosine Transformation is applied to the blocks where the pixel values are shifted into the range [-128, 127] with zero as the center.
Each of these values is then transformed using Forward DCT (FDCT). DCT is similar to Discrete Fourier Transformation as it maps the values from the time to the frequency domain.

b. Quantization

The JPEG application provides a table of 64 entries. Each entry will be used for the quantization of one of the 64 DCT- coefficients.

c. Entropy Encoding

During the initial step of entropy encoding, the quantized DC- coefficients are treated seperately from the quantized AC- coefficients.



- The DC- coefficient determines the basic color of the data units.
- The DCT processing order of the AC coefficients involves the zigzag sequence to concentrate the number of zeros.

JPEG specifies Huffman and arithmetic encoding as entropy encoding methods. However, as this is lossy sequential DCT-based mode, only Huffman encoding is allowed. In lossy sequential mode the framework of the whole picture is not formed but parts of it are drawn i.e. sequentially done.

Expanded Lossy DCT-based mode

It differs from the sequential mode in terms of number of bits per sample. Here 12 bits along with 8 bits per sample can be used

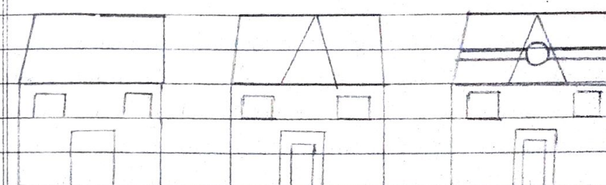


Fig: Progressive Picture Presentation

For the expanded lossy DCT- based mode, JPEG specifies progressive encoding in additional to sequential encoding.
At first, a very rough representation of the image appears which is progressively refined utill the whole image is formed. This progressive coding is achieved by layered coding.

Progressiveness is achieved in two different ways:

→ By using a spectral selection in the first run only, the quantized DCT coefficients of low frequencies of each data unit are passed in the entropy encoding. In successive runs, the coefficients of higher frequencies are processed.

→ Successive approximation transfers all of the quantized coefficients in each run, but single bits are differentiated according to their significance. The most-significant bits are encoded first, then the less-significant bits.

MPEG (Motion/Moving Pictures Expert Group)

Motion Pictures Expert Group (MPEG) standards are digital video encoding processes that coordinate the transmission of multiple forms of media (multimedia). MPEG is a method for video compression, which involves the compression of digital images and sound, as well as synchronization of the two. There currently are several MPEG standards. MPEG-1 is intended for intermediate data rates, on the order of 1.5Mbit/s. MPEG-2 is intended for high data rates of at least 10Mbit/s. MPEG-3 was intended for HDTV compression but was found to be redundant and was merged with MPEG-2. MPEG-4 is intended for very low data rates of less than 64kbit/s.

Types of Image Frames in MPEG

There are two basic types of compressed frame: those

that are encoded independently and those that are predicted. The first are as intracoded frames or I-frames. In practice, there are two types of predicted frames: predictive or P-frames and bidirectional or B-frames and because of the way they are derived, the latter are also known as intercoded or interpolation frames.

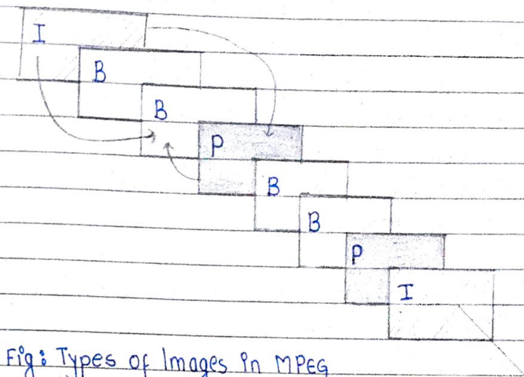


Fig: Types of Images in MPEG

a. I-frames (Intra coded Images)

I-frames are encoded without reference to any other frames.

Each frame is treated as a separate (digitized) picture and the Y, C, and C matrices are encoded independently using the JPEG algorithm.

b r The level of compression obtained with I-frames is relatively small. I-frames must be present in the output stream at regular intervals in order to allow for the possibility of the contents of an encoded I-frame being corrupted during transmission. The number of frames/pictures between successive I-frame is known as a group of pictures (GOP).

b. P-frames (Predictive-coded frames)

The encoding of a P-frame is relative to the contents of either a preceding I-frame or a preceding P frame.

As indicated, P-frames are encoded using a combination of motion estimation and motion compensation and hence significantly higher levels of compression can be obtained. In practice, the number of P-frames between each successive pair of I-frames is limited since any errors present in the first P-frame will be propagated to the next. The number of frames between a P-frame and the immediately preceding I- or P-frame is called the prediction span. It is given the symbol M and typical values range from 1 through 3.

c. B-frames (Bi-directionally predictive-coded frames)

Motion estimation involves comparing small segments of two consecutive frames for differences and, should a difference be detected, a search is carried out to determine to which neighboring segment the original segments has moved. In order to minimize the time for each search, the search region is limited to just a few neighbouring segments. Some applications may involve very fast moving objects; however, it is possible for a segment to have moved outside the search region. To allow for this possibility, in applications such as movies, in addition to P-frames, second types of frames are used called B-frames.

Audio Encoding

→ MPEG audio coding uses the same sampling frequency as CD i.e. 44.1 kHz.

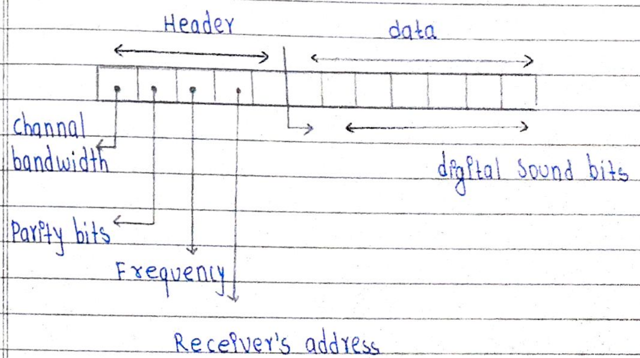
→ Similar to video the different compressed audio file mostly used are MP3, AAC, etc.

→ They are used to decrease the file size for audio significantly. e.g.: A normal uncompressed audio having file size 40 MB can be compressed to mp3 format having file size of 3-4 MB.

→ Just like video compression the transformation into frequency domain is achieved using DCT.

→ The digital sound spectrum is divided into 32-sub bands where each band contains the header section and the actual data.

i.e.



→ For each sub-band the amplitude is calculated.

→ For higher noise a rough quantization is applied and for a lower noise a fine quantization is applied.

→ Finally the compression technique is applied. i.e. Huffman Encoding.

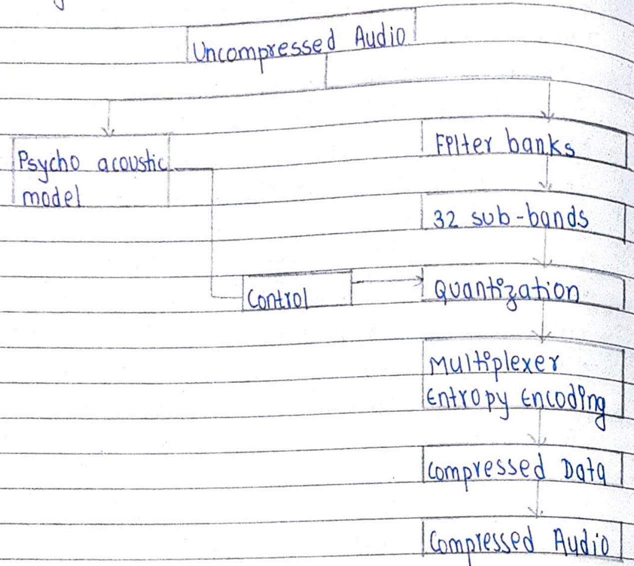


Fig: MPEG basic steps for audio Encoding

The time-varying audio input signal is first sampled and quantized using PCM, the sampling rate and number of bits per sample being determined by the specific application. The bandwidth that is available for transmission is divided into a number of frequency sub bands using a bank of analysis filters which, because of their role, are also known as critical-band filters.

Each frequency subbands is of equal width and, essentially, the bank of filters maps each set of 32 PCM samples into an equivalent frequency samples, one per subband. Hence each is known as a sub band sample and indicates the magnitude of each of the 32 frequency components that are present in a segment of the audio input signals of a time duration equal to 32 PCM samples.

H.261 (p x 64)
↳ times (*), p range from 1 to 30

→ H.261 is a video coding standard published by the ITU-T in 1990.
→ It uses ISDN standard
→ It has a bit rate of 64 kbit/second.
→ H.261 is usually used in conjunction with other control and framing standards.
→ The primary ISDN applications were videotelephone and video conferencing. so, they must be carried at in real time.
→ The maximum delay for both compression and decompression must not exceed 150 ms.

Image preperation:
→ It uses a very precise format with a refresh rate of 29.97 frames per second and aspect ratio of 4:3.
→ Often the amount of information in the difference between two frames is lot less.
→ As video is the sequence of image, so the main idea is to send only the difference between two frames but not the frame itself.

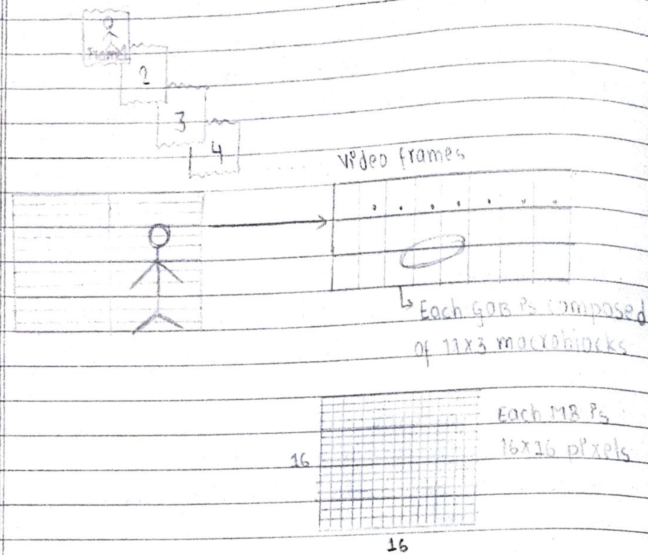


Fig : H.261 structure

- Common Intermediate Format (CIF) defines a luminance component of 288 lines each with 352 pixels.
- chrominance components have resolution with a rate of 144 lines and 176 pixels per line of fulfill 2:1:1 ratio
- QCIF (Quater CIF) has exactly half the CIF resolution
- Macroblock Y the result of combining four-blocks of Y-matrix each with one block of Cb and Cr components.
- The Images are encoded as luminance (Y) and chrominance Cb and Cr according to subsampling technique.

Video Encoding

For motion video encoding, the DVI (Digital Video Interface) distinguishes two technique.

i. Presentation level Video (PLV)

- It is characterized for its better quality.
- It consumes more time and performed by specialized compression techniques.
- It is mostly available for applications distributed on CD-ROM's.

ii. Real Time Video (RTV)

- It is a symmetric compression technique that works with hardware and software and can be performed in Real time.
- It has reduced image quality.
- RTV is mostly used for interactive communication in the same manner as px64.

Advantages of data compression

- It helps to occupy less disk space or storage space. This is due to the fact that when compressed, quatity of bits used to store the information is reduced.
- Compressed data is read/written faster than original data.
- It enables faster file transfer on the internet due to reduction in file size. Moreover file compression can zip

up several small files into a single file for more convenient email transmission.
→ Data compression is byte order independent.
→ Data compression has variable dynamic range which depends on algorithm used during compression.

Disadvantages:

→ Compression is mathematically intense process. Hence it may be time consuming when we have large number of files to be compressed. Moreover level of compression vary as per algorithm used, as a result compression time also vary.

Chapter - 6

Optical Storage Media | Device

Current magnetic data storage take the form of floppy disk or hard disks and used as secondary storage device. They have low average access time.

Optical storage media offers a higher storage density at a lower cost.

Data is recorded by making marks in a pattern that can be read back with the aid of light, usually a beam of laser light precisely focused on a spinning optical disc.

In optical-storage technology, a laser beam encodes digital data onto an optical, or laser, disk in the form of tiny pits arranged in concentric tracks on the disk's surface.

The audio compact disk is a commercial successful product in the multimedia industry.
e.g: CD-WORM (Compact Disk Write Once Read Memory)
CD-ROM
CD-ROM/XA (Extended Architecture)
CD-MO (Compact Disk Magnetic Optical)
...... etc.

CD-DA (Compact Disk Digital Audio)

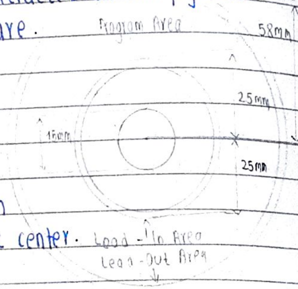
→ Compact Disc Digital Audio (CDDA or CD-DA) is the standard format for audio compact discs.

→ An audio CD is a music CD like that you buy in a music store.

→ It can be played on any standard CD player (such as a CD deck, or your car CD player, or a portable CD player).

→ Music is stored on Audio CDs as uncompressed digital data, no data is lost and quality is very high, exactly as in WAV (Waveform Audio File Format, WAVE) digitally encoded files.

→ When you put an Audio CD into your personal computer CD player and play it, audio is extracted on the fly and played by your PC sound hardware.



Physical characteristics:

- Diameter: 120 mm
- Constant linear velocity (CLV), i.e. number of rotations/s depends on position of head relative to disc center.
- Track shape: One spiral with appr. 20000 turns (LP: 850 turns)

Audio data rate:

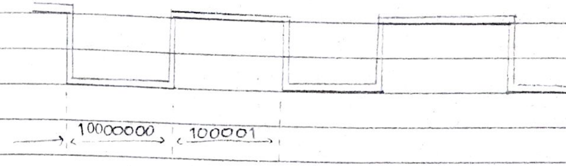
- Sampling frequency: 44100 Hz
- 16 bit quantization (Each audio sample is a signed 16-bit two's complement integer, with sample values ranging from -32768 to +32767)
- Pulse code modulation (PCM)
- Audio data rate = 1411200 bit/s = 176.4 kbytes

Quality:

- Signal to noise ratio (S/N): ~ 6dB/bit, 16 bit quantization → S/N exactly 98dB
- LP, tape: S/N 50-60 dB

Capacity:

- Playback time: maximal 74 min
- Capacity = 74 min * 1411200 bit/s = 6265728000 bit ≅ 747 MByte



CD-DA: Eight to fourteen Modulation

→ Eight-to-Fourteen Modulation, or EFM as it is abbreviated, is an encoding technique used by CDs and provides a way of countering errors by encoding a byte into 2 bytes.

→ Using EFM the data is broken into 8-bit blocks (bytes). Each 8-bit block is translated into a corresponding 14-bit codeword using a predefined lookup table.

→ The 14-bit codeword are chosen so that binary ones are always separated by a minimum of two and a maximum of ten binary zeroes.

→ EFM maximizes the number of transitions possible with an arbitrary pit and land length which is determined by the wavelength of the laser light used to read the data.

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Restricted laser resolution:

- Minimal distance between transitions (pit to land, land to pit)
- At least two "0" between two "1"

Generation of clock signal:

- Maximal distance between transitions (pit to land, land to pit)
- Not more than 10 consecutive "0"

Eight-to-fourteen modulation:

- 8 bit value is encoded using 14 bits
- 267 combinations possible
- 256 are used (criterion: efficient implementation with small number of gates)

CD-DA: Error Handling

Typical Errors:

- scratches, dust fingerprints can be characterized as "Burst errors"
- To be detected and corrected.

Two-level Reed-Solomon code with frame interleaving:

- First level: byte level, EDC and ECC

→ two groups: each with 4 correction bytes for 24 data bytes

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→ first group: correction of single byte errors

→ second group: correction of double byte errors, detection of further errors.

- second level: Frame interleaving

→ frame: 588 channel bits = 24 audio data bytes

→ distribution of consecutive data bytes and corresponding ECC bytes over adjacent frames.

CD-WO (Compact Disc Write Once)

→ The compack Disk write once (CD-WO), like WORM (Write Once Read Many), allows the user to write once to a CD and afterwards to read it many times.

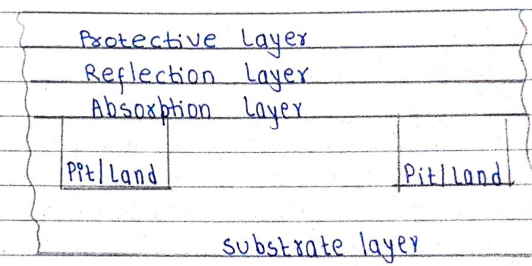

Fig: Cross-section of a CD-WO disk

→ Figure shows a cross-section of a CD-WO, vertical to the disk surface and data track.

→ In the case of read-only CDs, the substrate (a polycarbonate) lies directly next to the reflection layer.

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→ In the case of a CD-WO, an absorption layer exists between the substrate and the reflection layer. This layer can be irreversibly (permanently) modified through strong thermal influence, which changes the reflection properties of the laser beams.
→ In its original state, a CD-WO player recognizes a track consisting of lands.
→ The absorption layer in the pre-grooved track is heated to above 250°c with a laser three to four times the intensity of a reading player. Hence, the absorption layer changes such that the reflection of the laser light now corresponds to a pit.
→ This method determines the most remarkable property of the CD-WO: its data can be played by any devices which are meant only for read-only CD's.

CD-MO (Compact Disk Magneto Optical)

→ It has a high storage capacity and allows one to write multiple times to the CD.
→ The magnetic-optical method is based on the polarization of the magnetic field where the polarization is caused by a heat.
→ To be written, the block (sector) is heated to above 150°c. Simultaneously, a magnetic field approximately 10 times the strength of the earth's magnetic field is created.
→ The individual dipoles in the material are then polarized according to this magnetic field.

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→ Hereby, a pit corresponds to a low value of the magnetic field. A land is coded through a high value of the magnetic field.
→ After the CD is irradiated (illuminated) with a laser beam, the polarization of the light changes corresponding to the existing magnetization. Using this process, the read operation is executed.
→ For a delete activity, a constant magnetic field is created in the area of a block and the sector is simultaneously heated.

CD-ROM (Compact Disk - Read Only Memory

→ CD-ROM was designed for the storage format for general computer data.
→ A CD-ROM may contain audio and data: In such mixed form, the data tracks are usually at the begining followed by audio tracks. Hence, they are called Mixed Mode Disk.

Blocks:

A CD-DA has an error rate less than 10^{-8}. The CD-ROM requires much more error correction. For this purpose the CD data unit is divided into logical blocks.

Modes:

The CD-ROM is used to hold uncompressed Audio and computer data using two modes.

a. Mode-1

b. Mode-2

a. Mode-1

It serves for storage of computer datas. The block contains 2048 bytes for information storage out of the available 2352 bytes.

The 2352 bytes are split into the following groups:

- 12 bytes for synchronization i.e. for the detection of the block beginning.
- 4 bytes for the header, which carries an unambiguous specification of the block.
- 2048 bytes for the user data.
- 4 byte for error detection
- 8 unused bytes
- 276 bytes for error correction.

Sync	Header	User Data	EDC	blanks	ECC
12	4	2048	4	8	276

← 2352 byte →

Fig : CD-ROM mode-1 block layout

A CD-ROM contain 333,000 blocks to be played in 74 minutes. The capacity of a CD-ROM with all blocks in mode-1 can be computed as follows

$$\text{Capacity}_{\text{CD-ROM, mode 1}} = 333{,}000 \text{ blocks} \times 2048 \text{ bytes/blocks}$$

$$= 681{,}984000 \text{ bytes}$$

$$= 681{,}984{,}000 \times \frac{1}{2024 \text{ bytes/kbyte}} \times \frac{1}{1024 \text{ kbytes/Mbyte}}$$

$$\cong 650 \text{ Mbytes}$$

The data rate in mode-1 is:

$$\text{Rate}_{\text{CD-ROM, Mode-1}} = 2048 \text{ bytes/Block} \times 75 \text{ Blocks/s}$$

$$= 153.6 \text{ kbytes/s}$$

$$\cong 150 \text{ kbytes/s}$$

b. Mode-2

It holds data for any media. The synchronization are processed in the same way as in mode-1. Each block offers, 2336 bytes for information storage.

Sync	header	User data
12	4	2336

← 2352 byte →

Fig : CD-ROM mode-2 block layout

The capacity and data rate of a CD-ROM with all blocks in mode-2 be computed as follows:

$$\text{Capacity}_{\text{CD-ROM, mode-2}} = 333{,}000 \text{ blocks} \times 2336 \text{ bytes/block}$$

$$\cong 777{,}888{,}000 \text{ bytes}$$

$$\text{Rate}_{\text{CD-ROM, Mode-2}} = 2336 \text{ bytes/Blocks} \times 75 \text{ Blocks/s}$$

$$= 175.2 \text{ kbytes/s}$$

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CD-ROM / XA (Extended Architecture)

→ The compact disk read only memory / extended architecture (CD-ROM/XA) standard was established by N.V phellips and the sony and Microsoft Corporations and is based on the CD-ROM specification.
→ The main motive for this additional development is the concurrent output of several media.
→ The CD-ROM/XA uses CD-ROM mode-2, a sub header which describes a particular blocks.
→ The CD-ROM/XA differentiate blocks with form1 and form2 format that are similar to CD-ROM modes.

a. Form-1
→ CD-ROM mode-2 XA provides improved error detection and correction.
→ Unlike the eight bytes of mode-1 that are unused they are used as subheaders.

Sync	header	sub-header	User data	EDC	ECC
12	4	8	2048	4	276

2352 byte

Fig: CD-ROM/XA block layout Form-1

b. Form-2
→ This allow 13% more storage capacity for user data which means 2324 bytes for user data from 2352 bytes.
→ This is gained at the loss of error handling capability.

Sync	header	Sub-header	User data	EDC
12	4	8	2324	4

2352 byte

Fig: CD-ROM/XA block layout Form-2

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chapter- 7

COMPUTER TECHNOLOGY AND MULTIMEDIA OPERATING SYSTEMS (MOS)

The integration of multimedia hardware components with network to support multimedia communication. system is essential without necessary hardware and data storage capacity, the multimedia implementation would not be possible.

Communication Architecture

→ Local multimedia systems frequently use a network interface card (NIC). e.g: ethernet card for communication.

→ Until now the continuous and discrete media have been considered independently against each other. e.g: Analog telephone system carries the data through copper wires to different offices through intermediate switching systems.

→ Digital system uses network adaptors such as switches and routers.

1. Hybrid systems

→ By the use of today's technologies, integration and interaction between analog and digital environments can be implemented which is called hybrid approach.

→ The main advantages of this approach is the high quality audio and video output, input, storage and transmission.

→ To meet the goal of full digital integration, we consider the following approaches.

a. Integrated Device control

→ It mentions the control of analog input/output audio-video components in the digital environments.

→ The connections between source and destinations or switching of audio-video signals could be controlled digitally.

b. Integrated transmission control

→ second possibility to integrate analog and digital is to provide a common transmission control.

→ It implies that the analog audio-video source and destinations are connected to the computer for control process digitally.

c. Integrated transmission

→ Other possibility is to integrate is to provide a common transmission network.

→ External devices are connected to computer using A/D, D/A converters.

2. Digital systems

→ Digital systems are designed to store, process, and communicate information in digital form.

→ They are found in a wide range of applications, including process control, communication systems, digital instruments, and consumer products.

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a. connection to workstation
→ Here, the audio-video device can be connected directly to the computers and digitized audio-video are transmitted over networks.

b. Connection to switches
→ They are the device that are similar to bridge.
→ The switches extracts the destination header from the frame header and looks up in the table to see where to send the frame.

Multimedia Workstation

Workstation, a high-performance computer system that is basically designed for a single user and has advanced graphics capabilities, large storage capacity, and a powerful central prosessing unit.

Components of multimedia workstation:

a. Processor: High core count CPU's capable of handling video editing software, encoding, and rendering duty.

b. Capture and TV tuner cards: Capture cards and TV tuners are used to pull in video from external sources. You may or may not need one of these depending on your use case.

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c. Internal storage / NAS: Multimedia production typically requires A LOT of storage. If you have a large tower it can be housed internally, however NAS (Network Attached Storage) systems are becoming more and more popular due to their increased flexibility.

d. RAM: Multimedia applications require large amounts of RAM. These systems typically start at 32GB and can go up over 512GB.

e. Graphics card: Workstation GPU's are typically the choice here. Nvidia Quardo, and AMD Firepo cards offer advance drivers that deliver the accuracy needed for professionals.

f. Studio Monitors: Sound is an important part of any multimedia production computer. Good studio monitor speakers or headphones offer neutral, and uncolored audio suitable for production.

Multimedia Operating System
→ The operating system provides a comfortable environment for the execution of programs and ensures the effective utilization of computer hardware. It offers various services related to the neutral resources of a computer. i.e. CPU, main memory, storage, I/O devices.

→ The major aspect in this context is the real-time processing of the continuous media. Process management must take into account the timing requirements. The operating system should have proper scheduling algorithms.

→ Memory management has to provide access to datas with a guaranteed timing delay and efficient data manipulation functions.

→ It should have proper database management system. The incorporation of CD-ROM/XA file system is an integral part of a multimedia file system.

Real-Time Multimedia OS

A real-time process is a process which delivers the results of the processing in a given time-span.

Real-time systems:

"A system in which the correctness of a computation depends not only on obtaining the right result, but also upon providing the result on time."

Deadline:

A deadline represents the latest acceptable time for the presentation of a processing result. It marks the border between normal (correct) and anomalous (failing behavior. A real-time system has both hard and soft deadlines.

a. Soft deadlines:

The term soft deadline is often used for a deadline which cannot be exactly determined and which failing to meet does not produce and unacceptable result. If violated does not result in unacceptable results.

b. Hard deadline:

Hard deadlines should never be violated. A hard deadline violation is a system failure. Hard deadlines are determined by the physical characteristics of real-time processes.

characteristics of Real-time systems:

→ Predictably fast response to time-critical events and accurate timing information. For example in the control system of a nuclear power plant, the response to a malfunction must occur within a well-defined period to avoid a potential disaster.

→ High degree of schedulability. schedulability refers to the degree of resource utilization at which, the deadline of each time-critical task can be taken into account.

→ stability under transient overload. Under system overload, the processing of critical tasks must be ensured. These critical tasks are vital to the basic functionality provided by the system.

→ Some traditional real-time applications include manufacturing and monitoring systems. New applications include multimedia system, surveillance, etc.

characteristics of Multimedia OS:

→ More fault-tolerant – compare to real-time system for nuclear power plant control, the system in a video playback product will cause less damage if some errors occur.

→ Deadlines tend to be soft – for example, small errors in video playback timing is not noticeable.

→ Schedyability consideration is much easier because the media streams tend to be periodic (result of sampling and consistent.

→ Bandwidth requirement is not always stringent – more compression or lower resolution can always be used to achieve lower bit rates.

Resource Management

→ Multimedia systems with integrated audio and video processing are at the limit of their capacity.

→ Even with data compression, the computers do not allow processing of data compression, the computers do not allow processing of data according to their deadline without any resource reservation and real-time process management.

→ In real world scenario, we may see several multimedia application which uses should resources concorrently.

→ The shortage of resource requires careful allocations. Here, the system management must employ adequate scheduling algorithms.

→ Resource management in distributed multimedia system covers several computers involved during communication.

Resources:

A resource is a system entity required by tasks for manipulating data. Each resource has a set of distinguishing characteristics classified ~~dur~~ using the following scheme:

→ Active resources and passive resources

Active resources: An active resource is the CPU or a network adapter for protocol processing; it provide service.

Passive resources: A passive resource is the main memory, communication bandwidth or a file systems; it denotes some system capability required by active resources.

→ Resources can be used exclusively by one process at a time or shared between various processes.

Active resources: Active resources are often exclusive.

Passive resources: Passive resources can usually be shared among processes.

→ A resource that exists only once in the system is known as a single, otherwise it is a multiple resource. In a transputer-based multiprocessor system, the individual CPU is a multiple resource.

Phase of Resource Reservation and Management Process:

i. Resource manager provides components for the different phases of the allocation and management process. They are:

a. Schedulability Test
The RM checks with QOS parameter and determines if there is enough remaining resources and capacity to handle the request.

b. Quality of service calculation
After schedulability test, the RM calculates, the best possible performance for the request.

c. Resource reservation
The RM allocates the required capacity to meet the QOS guarantee for each request.

d. Resource scheduling
Incoming message from connections are scheduled according to the given QoS.

The reservation of resource can be made either in a pessimistic approach or an optimistic approach.

ii. The pessimistic approach avoids resource conflicts by making reservation for worst case i.e. resource bandwidth for longest processing time which leads to the under-utilization of resource.

iii. In optimistic approach, resource are reserved according to an average workload only and are highly utilized.

Traditional Real Time Scheduling
Two algorithm are analyzed for solving real time scheduling problems

a. EDF (Earliest Deadline First)
b. RMA (Rate Monotonic Algorithm)

→ A task can be preemptive task which means it can be interrupted by the request at any task with higher priority.
→ A non-preemptive task cannot be interrupted untill it releases the processor use; Here, the higher priority task may be subjected to priority inversion.

a. EDF (Earliest Deadline First)
The scheduler selects the tasks with the earliest deadline. If a new task arrives with a higher priority then the new task is processed immediately. The interrupted task can be scheduled after the task finishes.

characteristics of EDF:
i. It is optimal and dynamic algorithm where the processor utilization is 100%.
ii. It has a extension known as (TDS - Time Division Scheduling) which can handle the overloaded data situation during which

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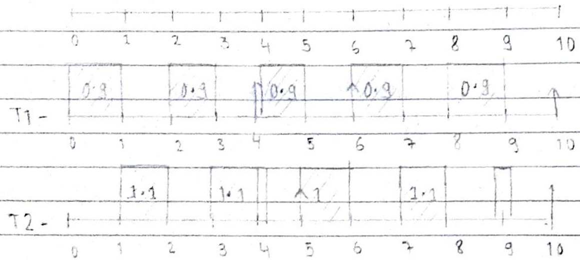
abort the task which cannot meet their deadline anymore

E.g -

Period	Execution	Deadline
Task 1 (T1): 2	0.9	2
Task 2 (T2): 5	2.3	5

soln:
Priority order, T1 → T2
Hyper period LCM (2,5) = 10

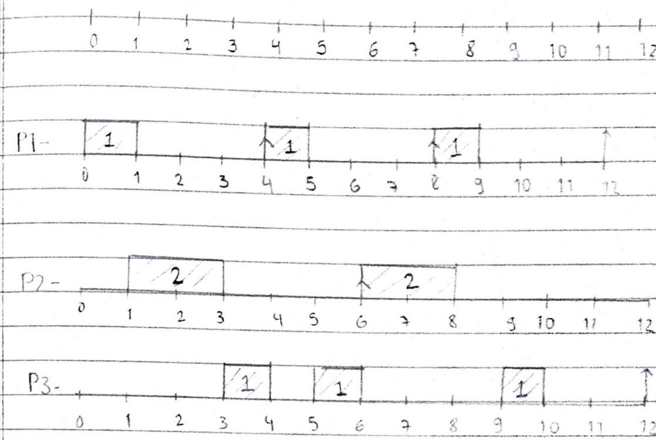


b. RMA (Rate Monotonic Algorithm)
- The algorithm is optimal, static, priority driven.
- There are no other static algorithm that is able to schedule a task set which cannot be scheduled by the RMA
- It schedules the task on the basis of the priority that is calculated in the beginning.
- Tasks with higher request rate will have higher priority.
- Tasks with shorter period will get high priority then task with longer period.

Eg:

Process	Execution time	Period
P1	1	4
P2	2	6
P3	3	12

soln:
Priority order: P1, P2, P3
Hyper period LCM = (4, 6, 12) = 12



P1 is preemptive by P3