CBCS SCHEME

USN

21EC745

Seventh Semester B.E./B.Tech. Degree Examination, Dec.2024/Jan.2025
Multimedia Communication

Time: 3 hrs. Max. Marks: 100

Note: ***Answer any FIVE full questions, choosing ONE full question from each module.***

Module-1

1 a. Describe with the aid of a diagram, how a PSTN can support range of Multimedia Common Applications. **(10 Marks)**

b. Classify five types of Communication network that are used to provide multimedia services. **(10 Marks)**

OR

2 a. Discuss the term interactive television with the help of a neat diagram. **(10 Marks)**

b. Outline the communication modes available to transfer the information stream. **(10 Marks)**

Module-2

3 a. Describe the function of signal encoder with the associated waveform. **(10 Marks)**

b. Assuming the bandwidth of a speech signal is from 50 Hz through to 10 KHz and that of a music signal is from 15 Hz through to 20 KHz , compute the bit rate that is generated by the digitization procedure in each case assuming the Nyquist sampling rate is used with 12 bits per sample for the speech signal and 16 bits per sample for the music signal. Derive the memory required to store a 10 min passage of stereophonic music. **(10 Marks)**

OR

4 a. Outline Raster – Scan operation associated waveform. **(10 Marks)**

b. Calculate the time to transmit the following digitized images at both 64 Kbps and 1.5 Mbps :

i) A $640 \times 480 \times 8$ VGA – Compatible image.

ii) A $1024 \times 768 \times 24$ SVGA – Compatible image. **(10 Marks)**

Module-3

5 a. A series of messages is to be transferred between Two Computers over a PSTN. The messages comprises just the characters A through H. Analysis has shown that the probability (relative frequency of occurrence) of each character is as follows :

A and B = 0.25 , C and D = 0.14 , E, F, G and H = 0.055.

i) Use Shannon's formula to calculate the minimum average number of bits per character.

ii) Use Huffmann coding to compute a code work set and show that this is the minimum set by constructing the corresponding Huffmann code tree. **(10 Marks)**

b. Describe JPEG Encoder with the aid of diagrams. **(10 Marks)**

OR

6 a. Derive the code for the string "went". Comprising characters with probability of e = 0.3 , n = 0.3 , t = 0.2 , w = 0.1 , • = 0.1, using Arithmetic coding. **(10 Marks)**

b. Discuss GIF and TIFF format. **(10 Marks)**

Module-4

7 a. Describe DPCM encoder and decoder with a neat diagram. **(10 Marks)**

b. A digitized video is to be compressed using the MPEG – I standard. Assuming a frame sequence of 1 B B P B B P B B P B B 1….. and average compression ratio of 10:1(1) , 20 : 1(P) and 50:1(B), calculate the average bit rate that is generated by encoder for both the NTSC and PAL digitization formats. **(10 Marks)**

OR

8 a. Describe linear predictive coding encoder and decoder with neat schematic. **(10 Marks)**

b. Illustrate H-261 Video encoder principles with a necessary diagram. **(10 Marks)**

Module-5

9 a. Discuss the frame format and operational parameters of Ethernet / IEEE 802.3. **(10 Marks)**

b. Describe the physical and MAC sub – layer of LAN protocol. **(10 Marks)**

OR

10 a. Compare the LAN protocols and Protocol framework. **(10 Marks)**

b. Describe in detail with diagrams the token ring configuration , frame formats , frame transmission and reception with priority operation. **(10 Marks)**

* * * * *

Q1) a Solution:

Public switched telephone networks (PSTNs) – initially designed to provide speech services. However, due to the advances in Digital Signal Processing (DSP) hardware and software now can support multimedia applications.

Data networks that initially supported data applications (email and ftp) now support much complex multimedia applications.

Text: Block of characters, each represented by a fixed number of binary digits (bits) known as codeword

Digitized image: Two-dimensional block of picture elements represented by a fixed number of bits

Audio and Video: Type of signal is known as an analogue signal and varies continuously with time (e.g: a telephone conversation can last for several minutes while a movie (audio + video) can last for a number of hours.

Single type of media - basic form of representation of a specific media type used

Mixed media – applications involving text and images or audio and video their basic form is used

Integrated media (text,images,audio,video)- Must convert all the four media into a suitable digital form.

PSTN – Now known as Plain Old Telephone Service (POTs)

The term switched means a subscriber can make a call to any other telephone on the 'total' network.

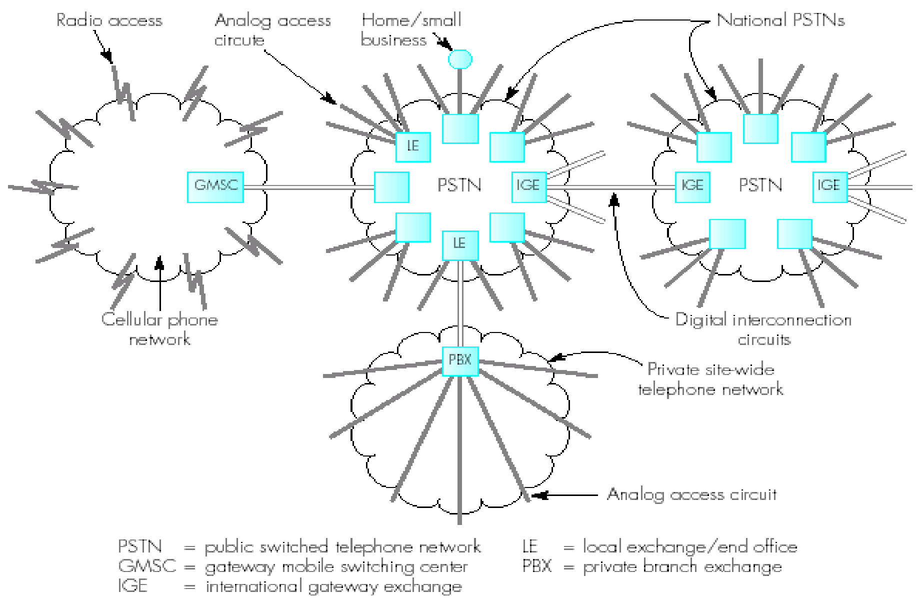


Fig.1 - Telephone Networks

PSTN (public switched telephone network) is the world's collection of interconnected voice-oriented public telephone networks, both commercial and government-owned.

It's the aggregation of circuit-switching telephone networks that has evolved from the days of Alexander Graham Bell.

Today, it is almost entirely digital in technology except for the final link from the central (local) telephone office to the user.

Q1b) Solution :

1)Telephone Networks - Telephony

2)Data Networks – Data Communications

3)Broadcast Television Networks – Broadcast TV

4)Integrated Services Digital Networks (ISDN) – Multi service

5)Broadband Multiservice Networks – Multi service

1)

PSTN (public switched telephone network) is the world's collection of interconnected voice-oriented public telephone networks, both commercial and government-owned.

It's the aggregation of circuit-switching telephone networks that has evolved from the days of Alexander Graham Bell.

Today, it is almost entirely digital in technology except for the final link from the central (local) telephone office to the user.

2)

Designed to provide basic data communication services such as email and general file transfer

Most widely deployed networks: X.25 network (low bit rate data) not suitable for multimedia and the Internet (Interconnected Networks)

Communication protocol: set of rules (defines the sequence and syntax of the messages) that are adhered to by all communicating parties for the exchange of information/data

Packet: Container for a block of data, at its head, is the address of the intended recipient computer which is used to route the packet through the network.

3)

Broadcast television networks support the diffusion of analogue television programs to a wider geographical area via a cable distribution network, a satellite network

A cable modem integrated into the STB (set-top-box) provides both a low bit rate channel (connects the subscriber to the PSTN) and a high bit rate channel (connects to the Internet) from the subscriber back to the cable head-end.

In Satellite and broadcast networks by integrating an H-S modem into the STB a range of interactive services can be supported. This is the origin of the term “interactive television”.

4)

Started to develop in the early 1980s to provide PSTN users the capability to have additional services

Integrated Services Digital Network (ISDN) in concept is the integration of both analogue or voice data together with digital data over the same network.

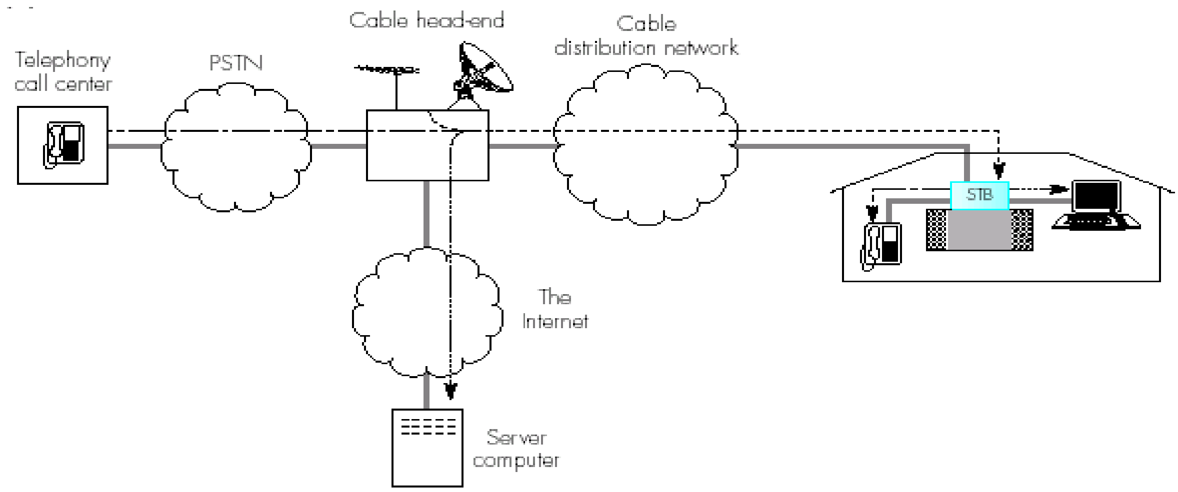
ISDN is a set of ITU standards for digital transmission over ordinary telephone copper wire as well as over other media. Home and business users who install an ISDN adapter (in place of a modem) can see highly-graphic Web pages arriving very quickly (up to 128 Kbps). ISDN requires adapters at both ends of the transmission so your access provider also needs an ISDN adapter. ISDN is generally available from your phone company.

5)

Broadband – Circuits associate with a call could have bit rates in excess of the maximum bit rate of 2Mbps – 30X64 kbps – provided by ISDN

Broadband integrated services digital network (B-ISDN) – All different media types are converted in the source equipment into a digital form, integrated togeather and divided into multiple fixed-sized packets (cells).

Q2)a Solution:

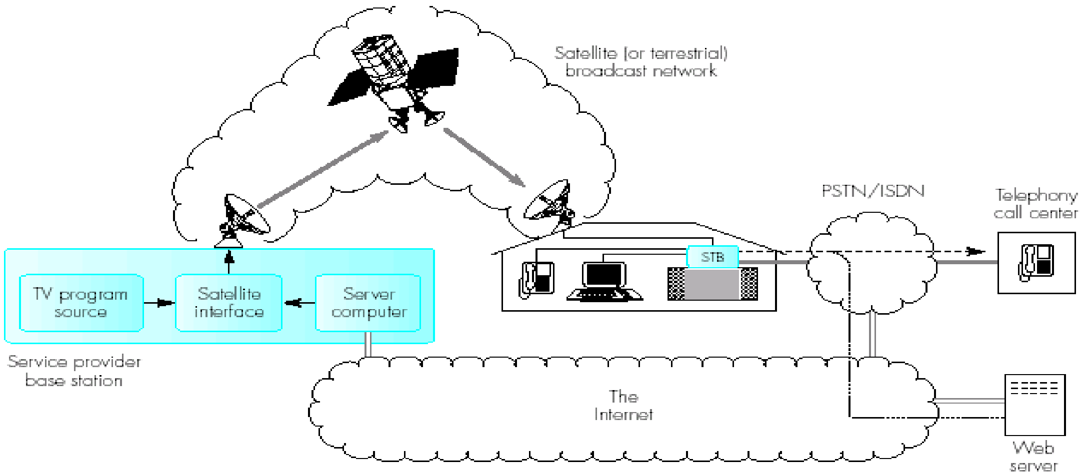


Interactive television (Cable network)

The set-top box (STB) provides both a low bit rate connection to the PSTN and a high bit rate connection to the internet

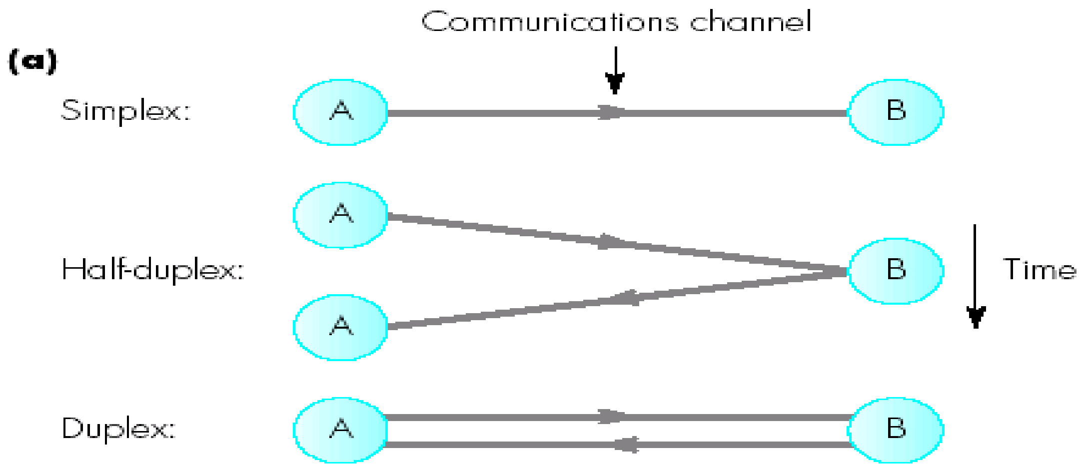
Through the connection to the PSTN, the subscriber is able to actively respond to the information being broadcast.

Interactive television (Satellite/terrestrial broadcast network)



The STB associated requires a high speed modem to provide the connections to the PSTN and the Internet.

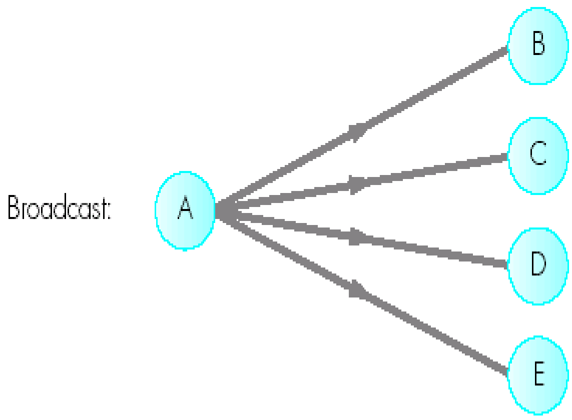
Q2b) Solution :

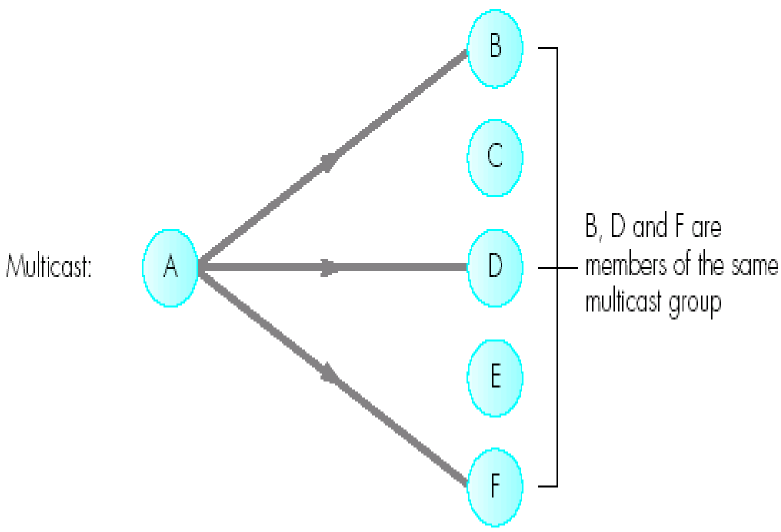


Simplex: The information associated with the application flows in one direction only.

Half-Duplex: Information flows in both directions but alternatively (two-way alternative).

Duplex: Information flows in both directions simultaneously (Two-way simultaneous).





Broadcast: The information output by a single node is received by all the other nodes connected to the same network

Multicast: The information output by the source is received by only a specific subset of the nodes (Latter form known as multicast group).

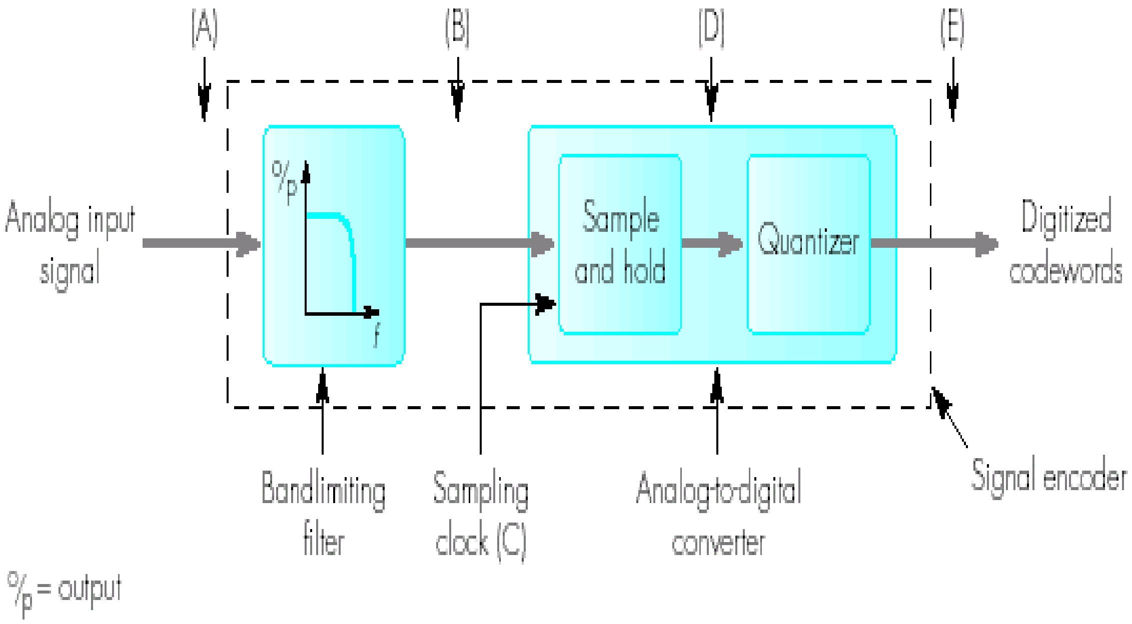
Communication mode Examples

In half-duplex and duplex communications, the bit rate associated with the flow of information in each direction can be equal (symmetric) or different (asymmetric).

Video Telephony – Symmetric duplex communication

Web browsing – Asymmetric half-duplex mode (as different bit rates for downloading and uploading).

Q3a) Solution :



A bandlimiting filter and an analog-to-digital converter(ADC), the latter comprising a sample-and-hold and a quantizer

Fig2.2

Remove selected higher-frequency components from the source signal (A)

(B) is then fed to the sample-and-hold circuit

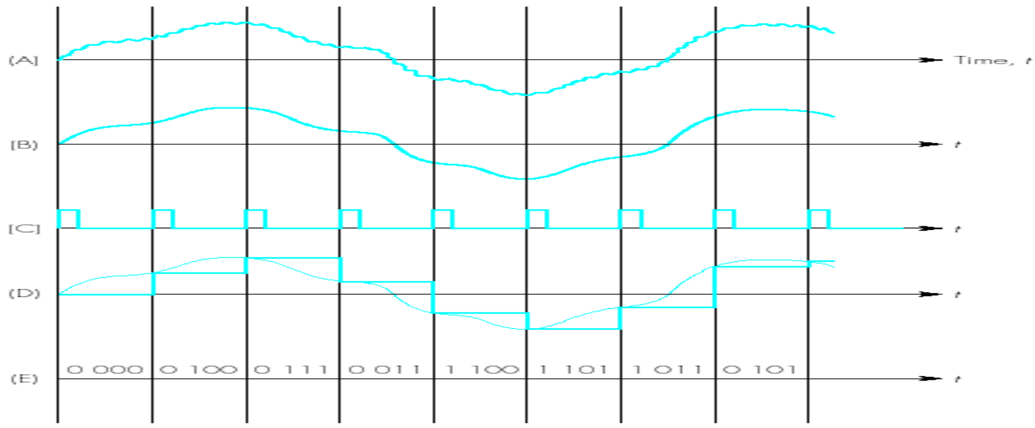
Sample the amplitude of the filtered signal at regular time intervals (C) and hold the sample amplitude constant between samples (D).

Quantizer circuit which converts each sample amplitude into a binary value known as a codeword (E)

The signal to be sampled at a rate which is higher than the maximum rate of change of the signal amplitude

The number of different quantization levels used to be as large as possible

Nyquist sampling theorem states that: in order to obtain an accurate representation of a time-varying analog signal, its amplitude must be sampled at a minimum rate that is equal to or greater than twice the highest sinusoidal frequency component that is present in the signal.



Bandlimiting filter: Removes the selected higher frequency components from the source signal

Sample and hold Circuit: Samples amplitude of the filtered signal at regular intervals and holds the sampled amplitudes between samples

Quantizer: Converts the samples into their corresponding binary form.

The most significant bit of the codeword represents the sign of the sample

A binary 0 indicates a positive value and a binary 1 indicates a negative value

The signal must be sampled at a much higher rate than the maximum rate of change of the signal amplitude

The number of quantization levels should be as large as possible to represent the signal accurately.

Q3b) Solution :

Assuming the bandwidth of a speech signal is from 50 Hz through to 10 kHz and that of a music signal is from 15 Hz through to 20 kHz, derive the bit rate that is generated by the digitization procedure in each case assuming the Nyquist sampling rate is used with 12 bits per sample for the speech signal and 16 bits per sample for the music signal. Derive the memory required to store a 10 minute passage of stereophonic music.

Answer:

(i) Bit rates: Nyquist sampling rate = $2 f_{max}$

Speech: Nyquist rate = 2×10 kHz = 20 kHz or 20 ksps
Hence with 12 bits per sample, bit rate generated = $20\text{k} \times 12 = 240$ kbps

Music: Nyquist rate = 2×20 kHz = 40 kHz or 40 ksps
Hence bit rate generated = $40\text{k} \times 16 = 640$ kbps (mono)
or $2 \times 640\text{k} = 1280$ kbps (stereo)

(ii) Memory required: Memory required = bit rate (bps) × time (s)/8 bytes
Hence at 1280 kbps and 600 s,

$$\text{Memory required} = \frac{1280 \times 10^3 \times 600}{8} = 96 \text{ Mbytes}$$

Q4a) Solution :

Color principles

A whole spectrum of colors— known as a color gamut —can be produced by using different proportions of red(R), green(G), and blue (B)

Fig 2.12

Additive color mixing producing a color image on a black surface

Subtractive color mixing for producing a color image on a white surface

Fig 2.13

Figure 2.12 Color derivation principles: (a) additive color mixing; (b) subtractive color mixing.

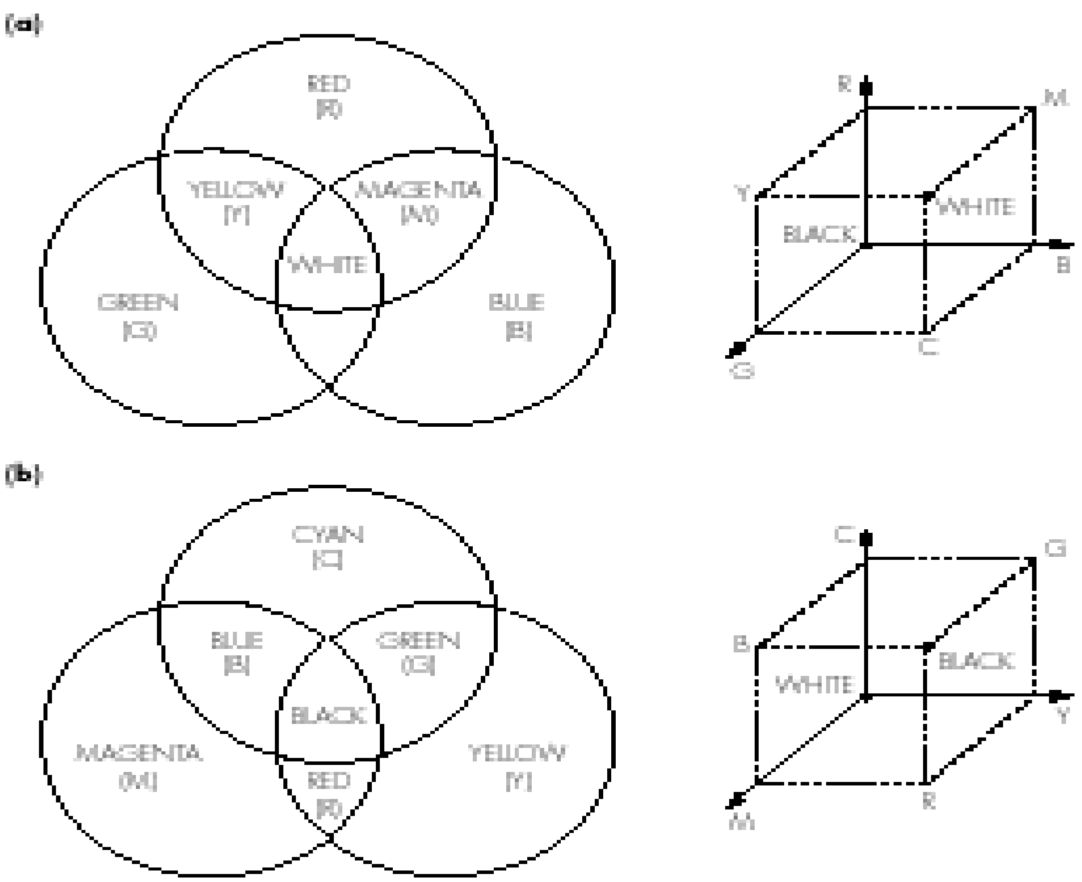
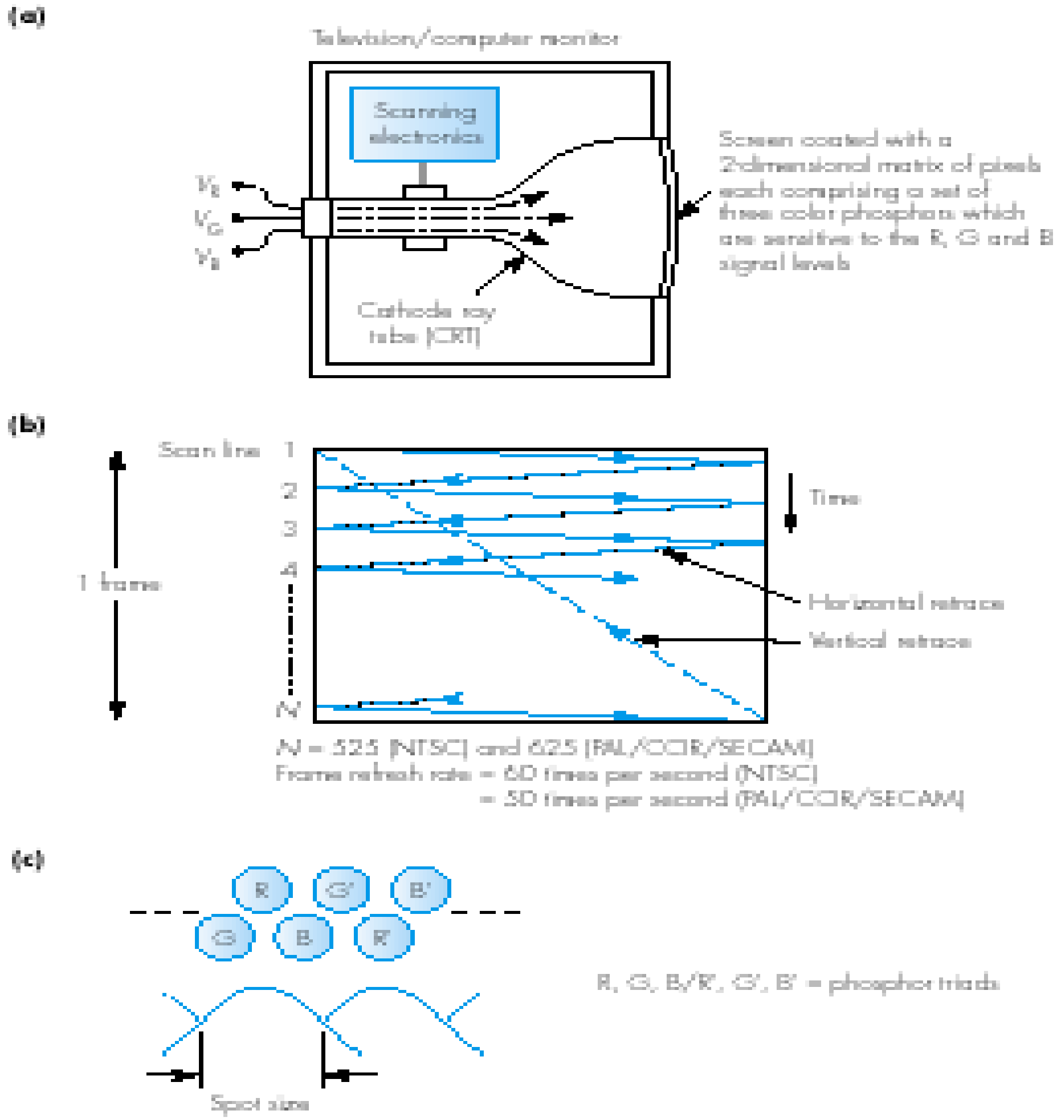


Figure 2.13 Television/computer monitor principles: (a) schematic; (b) raster-scan principles; (c) pixel format on each scan line.



Raster-scan principles

Progressive scanning

Each complete set of horizontal scan is called a frame

The number of bits per pixel is known as the pixel depth and determines the range of different colors.

Aspect ratio

Both the number of pixels per scanned line and the number of lines per frame

The ratio of the screen width to the screen height

National Television Standards Committee (NTSC), PAL(UK), CCIR(Germany), SECAM (France)

Table 2.1

Table 2.1 Example display resolutions and memory requirements.

Standard	Resolution	Number of colors	Memory required per frame (bytes)
VGA	640 × 480 × 8	256	307.2 kB
XGA	640 × 480 × 16	64K	614.4 kB
	1024 × 768 × 8	256	786.432 kB
SVGA	800 × 600 × 16	64k	960 kB
	1024 × 768 × 8	256	786.432 kB
	1024 × 768 × 24	16M	2359.296 kB

Q4b) Soluton :

Derive the time to transmit the following digitized images at both 64 kbps and 1.5 Mbps:

- a $640 \times 480 \times 8$ VGA-compatible image,
- a $1024 \times 768 \times 24$ SVGA-compatible image.

Answer:

The size of each image in bits is:

$$\text{VGA} = 640 \times 480 \times 8 = 2.457600\,\text{Mbits}$$
$$\text{SVGA} = 1024 \times 768 \times 24 = 18.874368\,\text{Mbits}$$

Hence the time to transmit each image is:

$$\text{At 64 kbps: VGA} = \frac{2.4576 \times 10^6}{64 \times 10^3} = 38.4\,\text{s}$$

$$\text{SVGA} = \frac{18.874368 \times 10^6}{64 \times 10^3} = 294.912\,\text{s}$$

$$\text{At 1.5 Mbps: VGA} = \frac{2.4576 \times 10^6}{1.5 \times 10^6} = 1.6384\,\text{s}$$

$$\text{SVGA} = \frac{18.874368 \times 10^6}{1.5 \times 10^6} = 12.5829\,\text{s}$$

As we can see, the times to transmit a signal image at 64 kbps are such that interactive access would not be feasible, nor at 1.5 Mbps with the higher-resolution SVGA image.

Q5a) Solution :

A series of messages is to be transferred between two computers over a PSTN. The messages comprise just the characters A through H. Analysis has shown that the probability (relative frequency of occurrence) of each character is as follows:

A and B = 0.25, C and D = 0.14, E, F, G, and H = 0.055

(a) Use Shannon's formula to derive the minimum average number of bits per character.

(b) Use Huffman coding to derive a codeword set and prove this is the minimum set by constructing the corresponding Huffman code tree.

(c) Derive the average number of bits per character for your codeword set and compare this with:
 (i) the entropy of the messages (Shannon's value),
 (ii) fixed-length binary codewords,
 (iii) 7-bit ASCII codewords.

Answer:

(a) Shannon's formula states:

$$\text{Entropy, } H = -\sum_{i=1}^{8} P_i \log_2 P_i \text{ bits per codeword}$$

Therefore:

$$H = -(2(0.25 \log_2 0.25) + 2(0.14 \log_2 0.14) + 4(0.055 \log_2 0.055))$$
$$= 1 + 0.794 + 0.921 = 2.175 \text{ bits per codeword}$$

(b) The derivation of the codeword set using Huffman coding is shown in Figure 3.4(a). The characters are first listed in weight order and the two characters at the bottom of the list are assigned to the (1) and (0) branches. Note that in this case, however, when the two nodes are combined, the weight of the resulting branch node (0.11) is greater than the weight of the two characters E and F (0.055). Hence the branch node is inserted into the second list higher than both of these. The same procedure then repeats until there are only two entries in the list remaining.

The Huffman code tree corresponding to the derived set of codewords is given in Figure 3.4(b) and, as we can see, this is the optimum tree since all leaf and branch nodes increment in numerical order.

(c) Average number of bits per codeword using Huffman coding is:

$2\,(2 \times 0.25) + 2(3 \times 0.14) + 4(4 \times 0.055) = 2.72$ bits per codeword

which is 99.8% of the Shannon value.

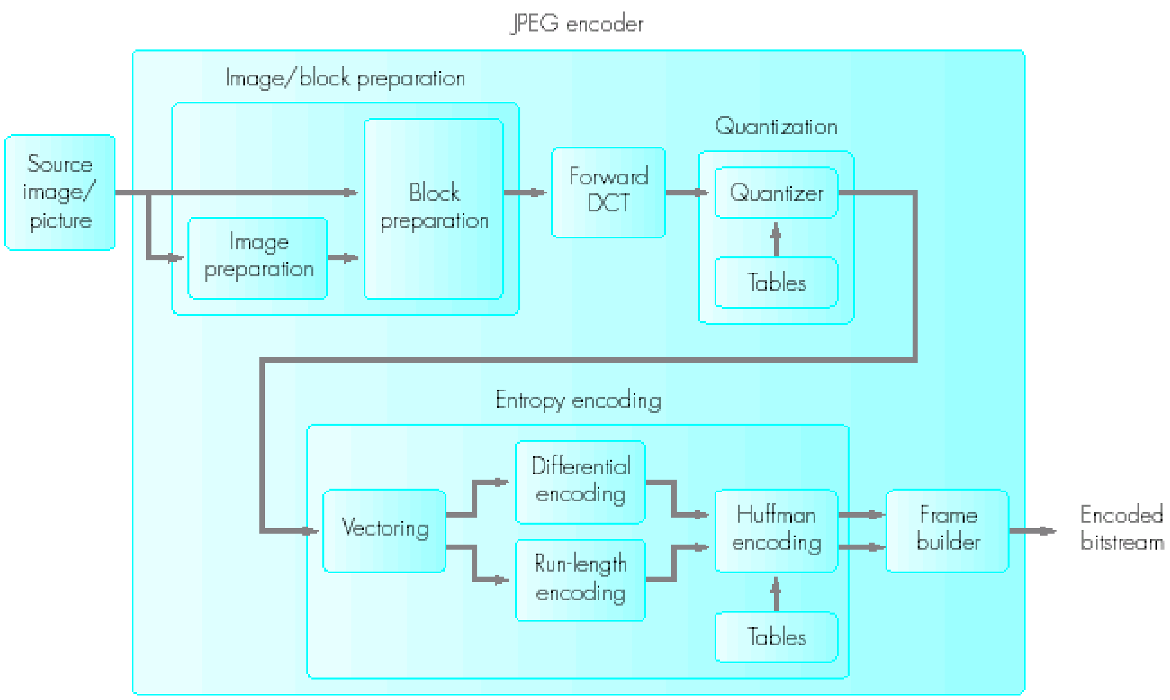
Using fixed-length binary codewords:

There are 8 characters – A through H – and hence 3 bits per codeword is sufficient which is 90.7% of the Huffman value.

Using 7-bit ASCII codewords:

7 bits per codeword
which is 38.86% of the Huffman value.

Q5b) Solutions :



3.4.5 JPEG

As we can deduce from the name, the JPEG standard was developed by a team of experts, each of whom had an in-depth knowledge of the compression of digitized pictures. They were working on behalf of the ISO, the ITU, and the IEC and JPEG is defined in the international standard **IS 10918**. In practice, the standard defines a range of different compression modes, each of which is intended for use in a particular application domain. We shall

restrict our discussion here to the **lossy sequential mode** – also known as the **baseline mode** – since it is this which is intended for the compression of both monochromatic and color digitized pictures/images as used in multimedia communication applications. There are five main stages associated with this mode: image/block preparation, forward DCT, quantization, entropy encoding, and frame building. These are shown in Figure 3.14 and we shall discuss the role of each separately.

Image/block preparation

As we described in Section 2.4.3, in its pixel form, the source image/picture is made up of one or more 2-D matrices of values. In the case of a continuous-tone monochrome image, just a single 2-D matrix is required to store the set of 8-bit gray-level values that represent the image. Similarly, for a color image, if a CLUT is used just a single matrix of values is required.

Alternatively, if the image is represented in an *R, G, B* format three matrices are required, one each for the *R, G,* and *B* quantized values. Also, as we saw in Section 2.6.1 when we discussed the representation of a video signal, for color images the alternative form of representation known as Y, C_b, C_r can optionally be used. This is done to exploit the fact that the two chrominance signals, C_b and C_r, require half the bandwidth of the luminance signal, Y. This in turn allows the two matrices that contain the digitized chrominance components to be smaller in size than the Y matrix so producing a reduced form of representation over the equivalent *R, G, B* form of representation. For example, in the 4:2:0 format, groups of four neighboring chrominance values are averaged to produce a single value in the reduced matrix so reducing the size of the C_b and C_r matrices by a factor of four. The four alternative forms of representation are shown in Figure 3.15(a).

Once the source image format has been selected and prepared, the set of values in each matrix are compressed separately using the DCT. Before performing the DCT on each matrix, however, a second step known as **block preparation** is carried out. This is necessary since to compute the transformed value for each position in a matrix requires the values in all the locations of the matrix to be processed. It would be too time consuming to compute the DCT of the total matrix in a single step so each matrix is first divided into a set of smaller 8 × 8 submatrices. Each is known as a **block** and, as we can see in part (b) of the figure, these are then fed sequentially to the DCT which transforms each block separately.

Q6b) Solution :

3.4.1 Graphics interchange format

The **graphics interchange format** (**GIF**) is used extensively with the Internet for the representation and compression of graphical images. Although color images comprising 24-bit pixels are supported – 8 bits each for R, G, and B – GIF reduces the number of possible colors that are present by choosing the 256 colors from the original set of 2^{24} colors that match most closely those used in the original image. The resulting table of colors therefore consists of 256 entries, each of which contains a 24-bit color value. Hence instead of sending each pixel as a 24-bit value, only the 8-bit index to the table entry that contains the closest match color to the original is sent. This results in a compression ratio of 3:1. The table of colors can relate either to the whole image – in which case it is referred to as the **global color table** – or to a portion of the image, when it is referred to as a **local color table**. The contents of the table are sent across the network – together with the compressed image data and other information such as the screen size and aspect ratio – in a standardized format. The principles of the scheme are shown in Figure 3.9(a).

As we show in Figure 3.9(b), the LZW coding algorithm can be used to obtain further levels of compression. We described this earlier in Section 3.3.5 when we discussed text compression and, in the case of image compression, this works by extending the basic color table dynamically as the compressed image data is being encoded and decoded. As with text compression, the occurrence of common strings of pixel values – such as long strings of the same color – are detected and these are entered into the color table after the 256 selected colors. However in this application, since each entry in the color table comprises 24 bits, in order to save memory, to represent each string of pixel values just the corresponding string of 8-bit indices to the basic color table are used. If we limit each entry in the table to 24 bits, then this will allow common strings comprising three pixel values to be stored in each location of the extended table. Normally, since the basic table contains 256 entries, an initial table size of 512 entries is selected which allows for up to 256 common strings to be stored. As with text compression, however, should more strings be found, then the number of entries in the table is allowed to increase incrementally by extending the length of the index by 1 bit.

GIF also allows an image to be stored and subsequently transferred over the network in an interlaced mode. This can be useful when transferring images over either low bit rate channels or the Internet which provides a variable transmission rate. With this mode, the compressed image data is organized so that the decompressed image is built up in a progressive way as the data arrives. To achieve this, the compressed data is divided into four groups as shown in Figure 3.10 and, as we can see, the first contains 1/8 of the total compressed image data, the second a further 1/8, the third a further 1/4, and the last the remaining 1/2.

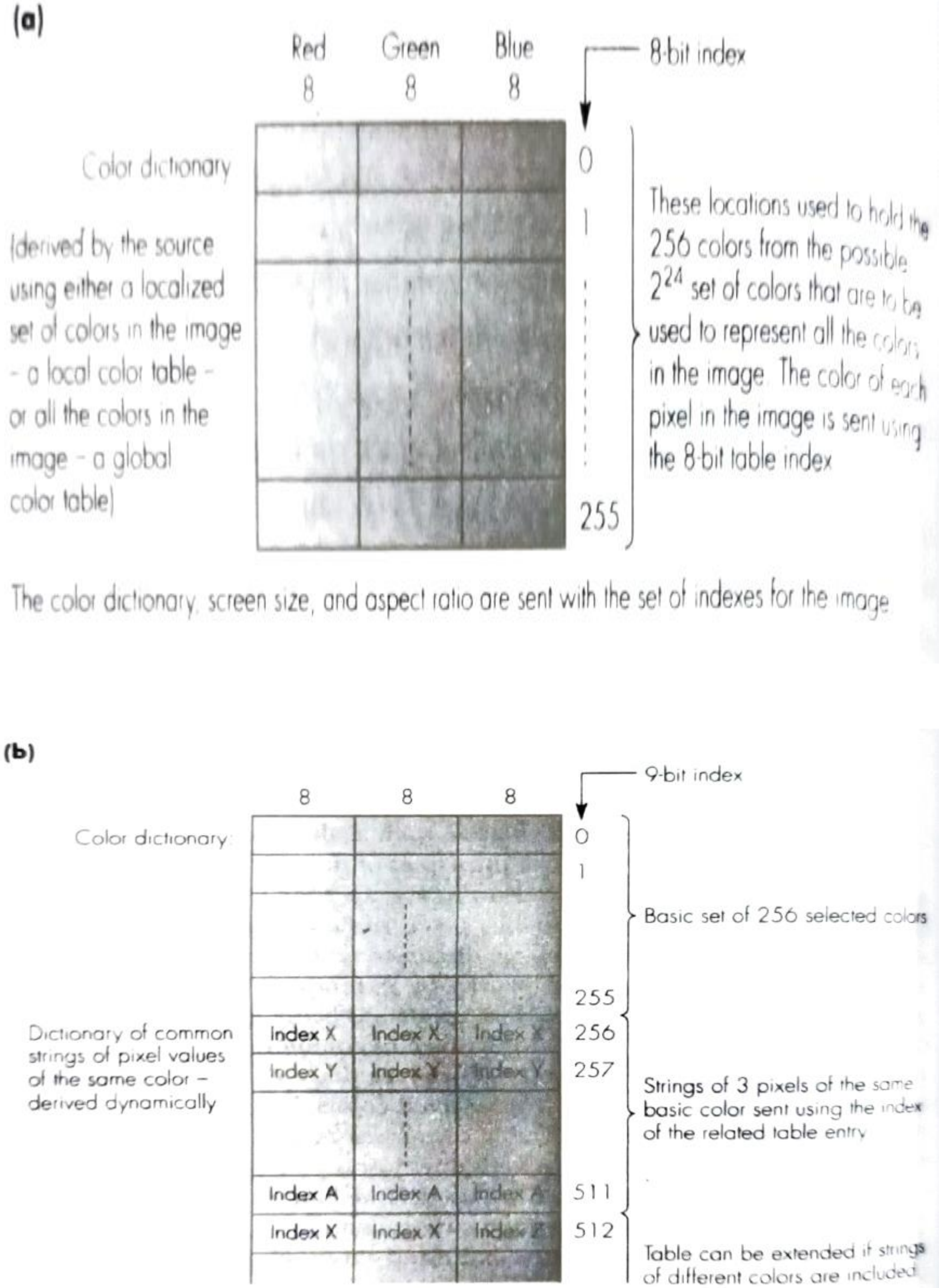
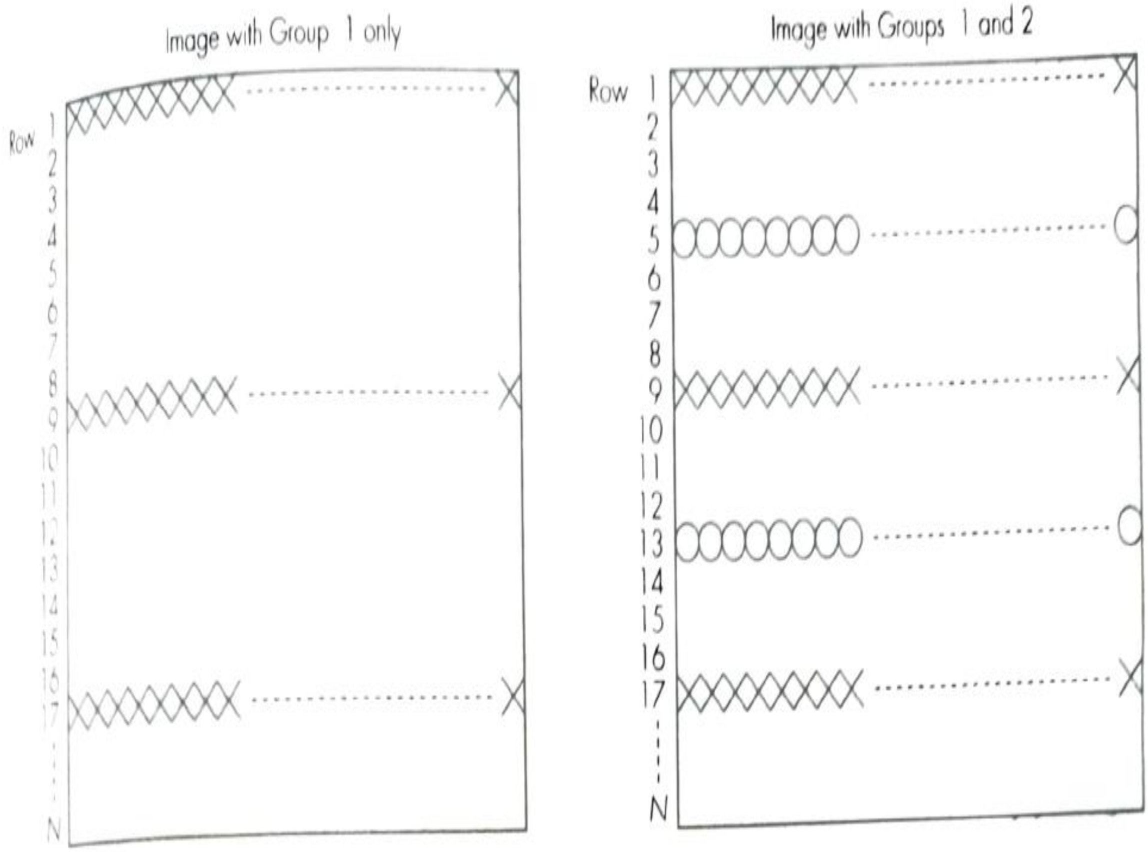


Figure 3.9 GIF compression principles: (a) basic operational mode; (b) dynamic mode using LZW coding.

3.4.2 Tagged image file format

The **tagged image file format (TIFF)** is also used extensively. It supports pixel resolutions of up to 48 bits – 16 bits each for R, G, and B – and is intended for the transfer of both images and digitized documents. The image data, therefore, can be stored – and hence transferred over the network – in a number of different formats. The particular format being used is indicated by a code number and these range from the uncompressed format (code number 1) through to LZW-compressed which is code number 5. Code numbers 2, 3, and 4 are intended for use with digitized documents. These use the same



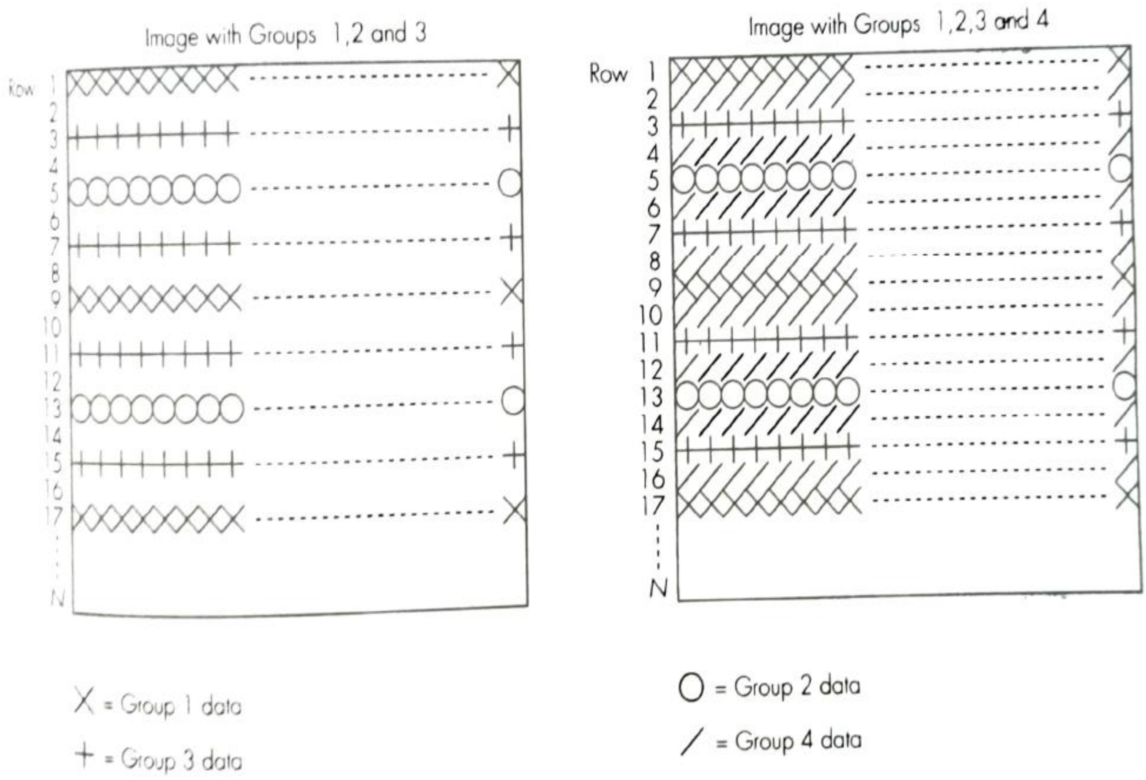


Figure 3.10 GIF interlaced mode.

compression algorithms that are used in facsimile machines which we discuss in the next section.

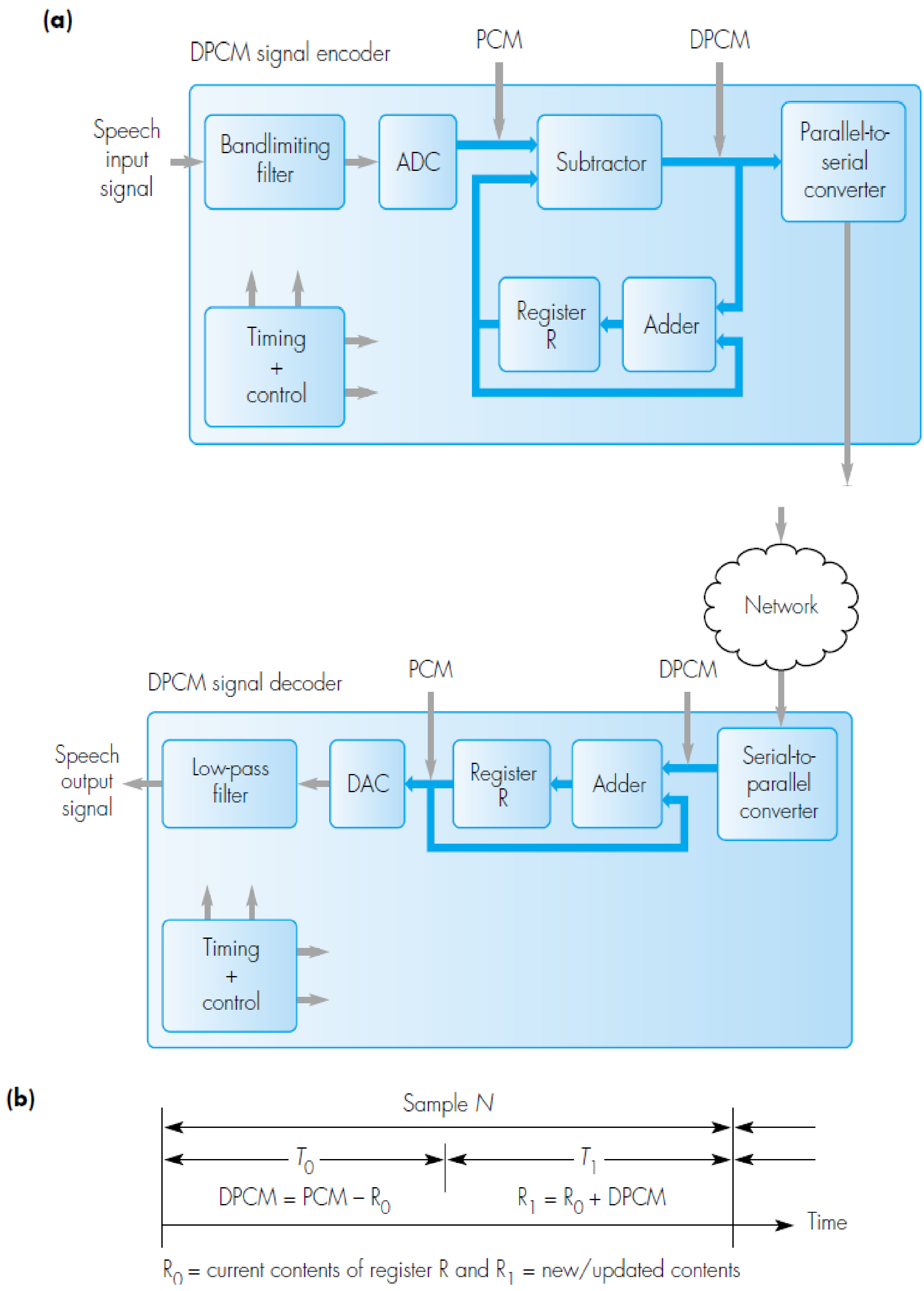
The LZW compression algorithm that is used is the same as that used with GIF. It starts with a basic color table containing 256 colors and the table can be extended to contain up to 4096 entries containing common strings of pixels in the image being transferred. Again, a standard format is used for the transfer of both the color table and the compressed image data.

Q7a) Solution :

DPCM is a derivative of standard PCM

for most audio signals, the range of the differences in amplitude between successive samples of the audio waveform is less than the range of the actual sample amplitudes.

Figure4.1



The previous digitized sample value is held in reg R

Difference signal is by subtracting (Ro) from the digitized sample of ADC

Reg R is updated with the difference signal

The decoder adds the DPCM with previously computed signal in the reg

The o/p of ADC is also known as residual

There are schemes to predict the more accurate previous signal

The proportions used are determined by predictor co-efficients

Q7b) Solution :

A digitized video is to be compressed using the MPEG-1 standard. Assuming a frame sequence of:

IBBPBBPBBPBBI...

and average compression ratios of 10:1 (I), 20:1 (P) and 50:1 (B), derive the average bit rate that is generated by the encoder for both the NTSC and PAL digitization formats.

Answer:

Frame sequence = IBBPBBPBBPBBI...

Hence: 1/12 of frames are I-frames, 3/12 are P-frames, and 8/12 are B-frames.

and Average compression ratio $= (1 \times 0.1 + 3 \times 0.05 + 8 \times 0.02)/12$
$= 0.0342$ or 29.24:1

NTSC frame size:

$$\text{Without compression} = 352 \times 240 \times 8 + 2\ (176 \times 120 \times 8)$$
$$= 1.013760\ \text{Mbits per frame}$$

$$\text{With compression} = 1.01376 \times 1/29.24$$
$$= 34.670\ \text{kbits per frame}$$

$$\text{Hence bit rate generated at 30 fps} = 1.040\ \text{Mbps}$$

Q8a) Solution :

All algorithms – sampling, digitization and quantization using DPCM / ADPCM

DSP crcuits help in analyzing the signal based on the required features (perceptual) and then quantized

Origin of sound is also important – vocal tract excitation parameters

Voiced sounds-generated through vocal chords

Unvoiced sounds – vocal chords are open

These are used with proper model of vocal tract to produce synthesized speech

After analyzing the audio waveform , These are then quantized and sent and the destination uses them,together with a sound synthesizer,to regenerate a sound that is perceptually comparable with the source audio signal.this is LPC technique.

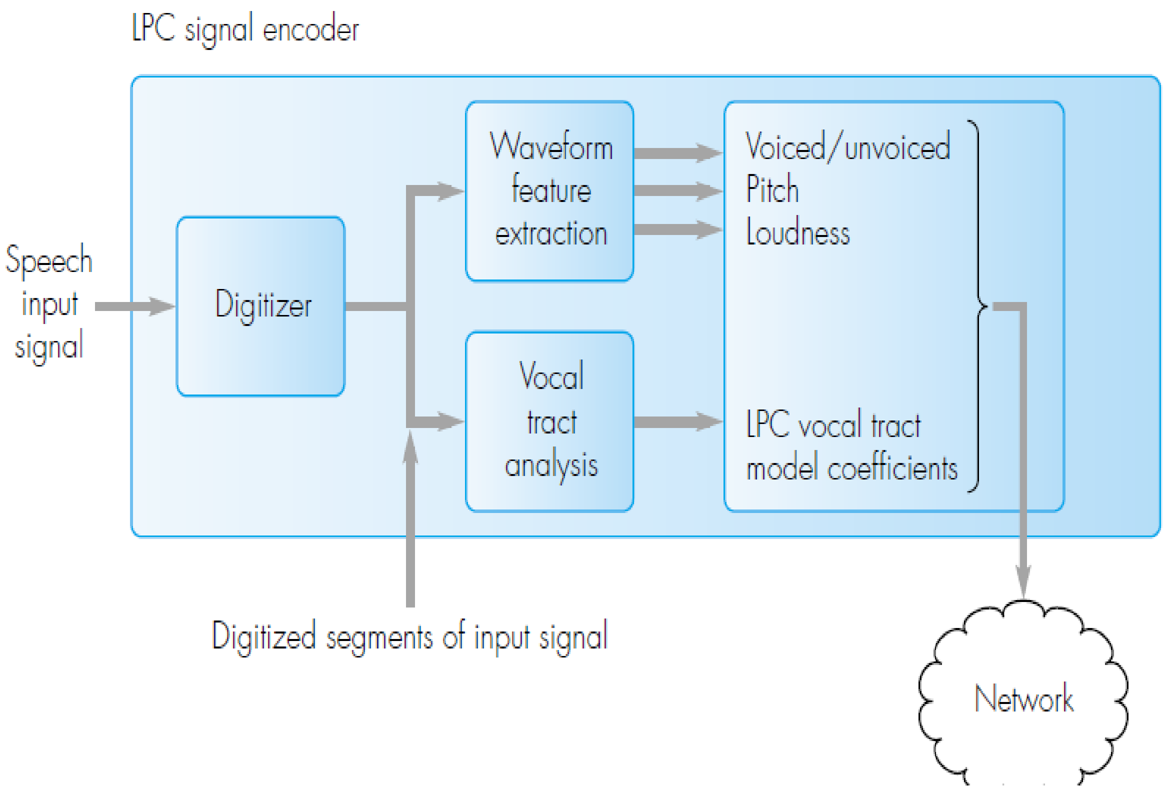
Three feature which determine the perception of a signal by the ear are its:

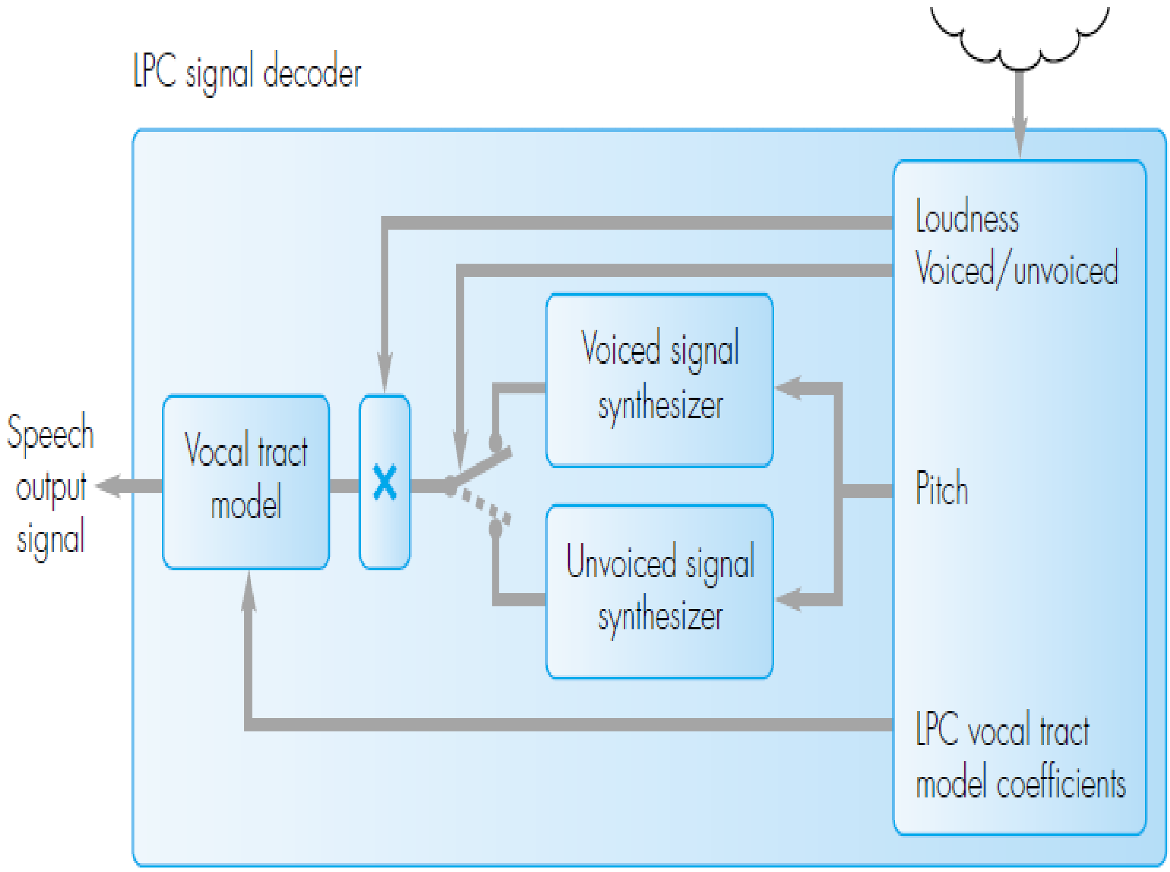
Pitch

Period

Loudness

Basic feature of an LPC encoder/decoder: figure 4.4





The i/p waveform is first sampled and quantized at a defined rate

Segment- block of sampled signals are analyzed to define perceptual parameters of speech

The speech signal generated by the vocal tract model in the decoder is the present o/p signal of speech synthesizers and linear combination of previous set of model coefficients

Hence the vocal tract model is adaptive

Encoder determines and sends a new set of coefficients for each quantized segment

The output of encoder is a set of frames ,each frame consists of fields for pitch and loudness

Bit rates as low as 2.4 or 1.2 kbps. Generated sound at these rates is very synthetic and LPC encoders are used in military applications, where bandwidth is important.

Q8b) Solution :

For the provision of video telephony and videoconferencing services over an ISDN

Transmission channels multiples of 64kbps

Digitization format used is either the common intermediate format(CIF) or the quarter CIF(QCIF)

Progressive scanning used with frame refresh rate of 30fps for CIF and 15or 7.5fps for QCIF

CIF:Y=352X288, Cb=Cr=176X144

QCIF:Y=176X144, Cb=Cr=88X72

H.261 encoding format show figure 4.15

I Frame and pframes are used with 3 p frames between each pair of I frames

Each macroblock has an address for identification

Type field indicates the macroblock is intracoded or intercoded

Quantization value is threshold value and mv is the encoded vector

Coded block pattern defines which of six 8x8 pixel block make up macroblock and the JPEG encoded DCT COEFFICIENTS are given in each block

Picture start code- Start of each video frame

Temporal ref field- time stamp to synchronize video block with the associated audio block of the same time stamp

Picture type field- type of frame (I or P frame)

GOB- GROUP OF MACROBLOCKS (size is chosen such that CIF and QCIF has integral number of GOBs)

EACH GOB – Unique start code – resynchronization marker

Each GOB also has group no.

For bandwidth optimization variable bit rate of encoder is converted into const bit rate

By passing through FIFO buffer

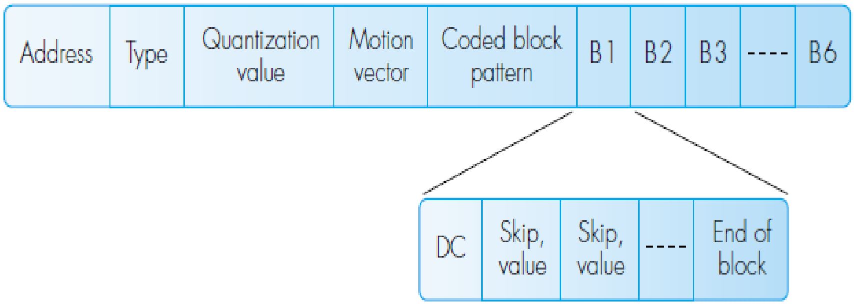
Feedback is provided to quantizer

o/p of the buffer is defined by the transmission bit rate, two threshold values are defined low and high

If contents of buffer is below the low threshold ,quantization threshold is reduced and the o/p rate is increased, if it is above high threshold then the threshold is increased and the o/p rate is reduced

Control proceedure is implemented for GOB

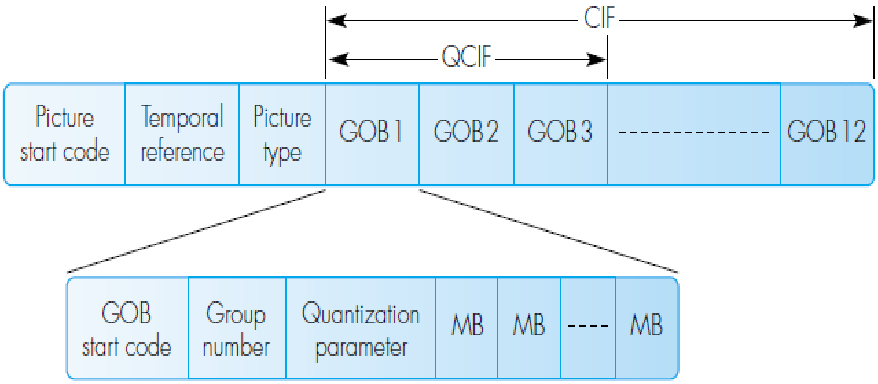
(a)


Address
Type
Quantization value
Motion vector
Coded block pattern
B 1
B 2
B 3

B 6
DC
Skip, value
Skip, value

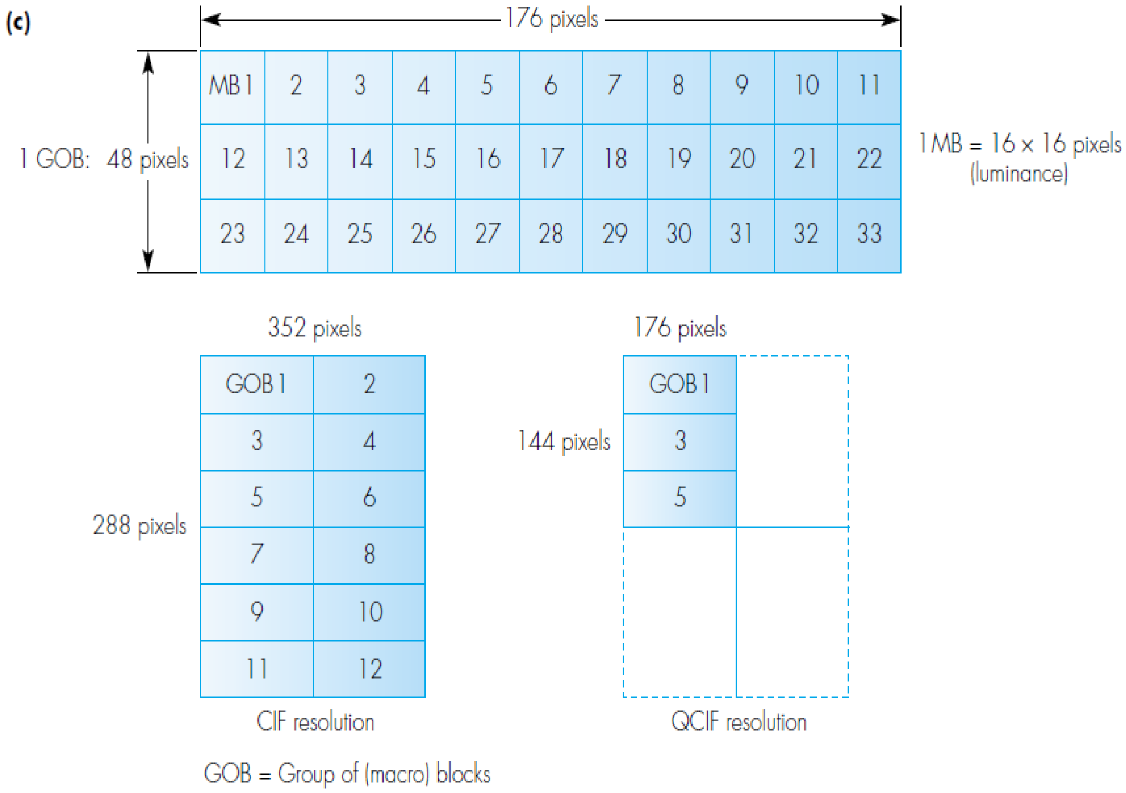
End of block

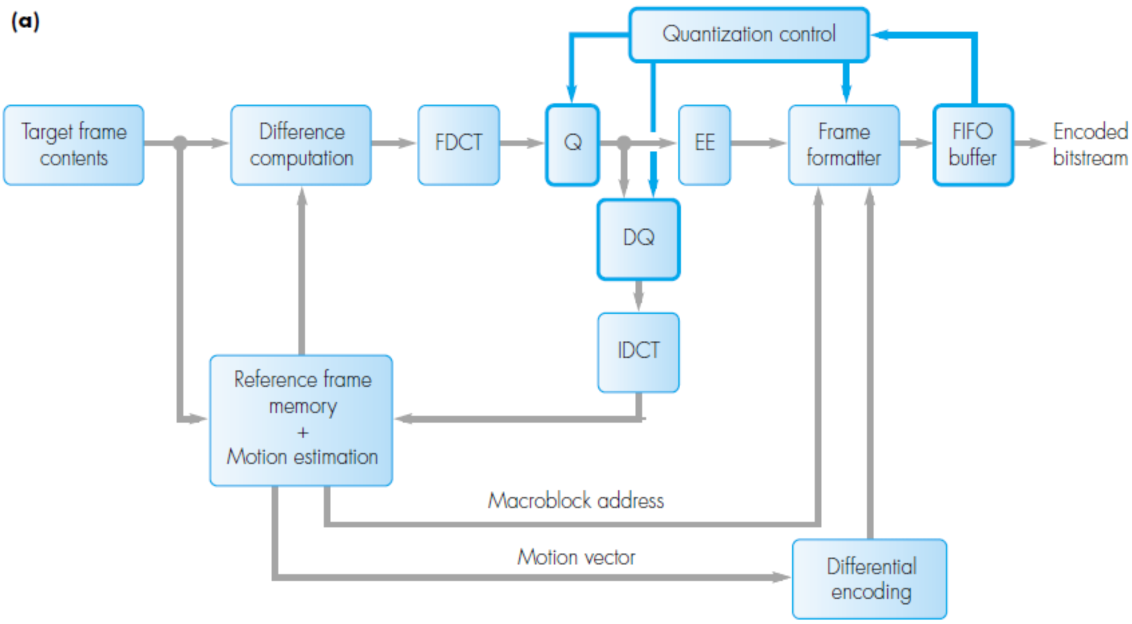
(b)


CIF
QCIF
Picture start code
Temporal reference
Picture type
GOB 1
GOB 2
GOB 3

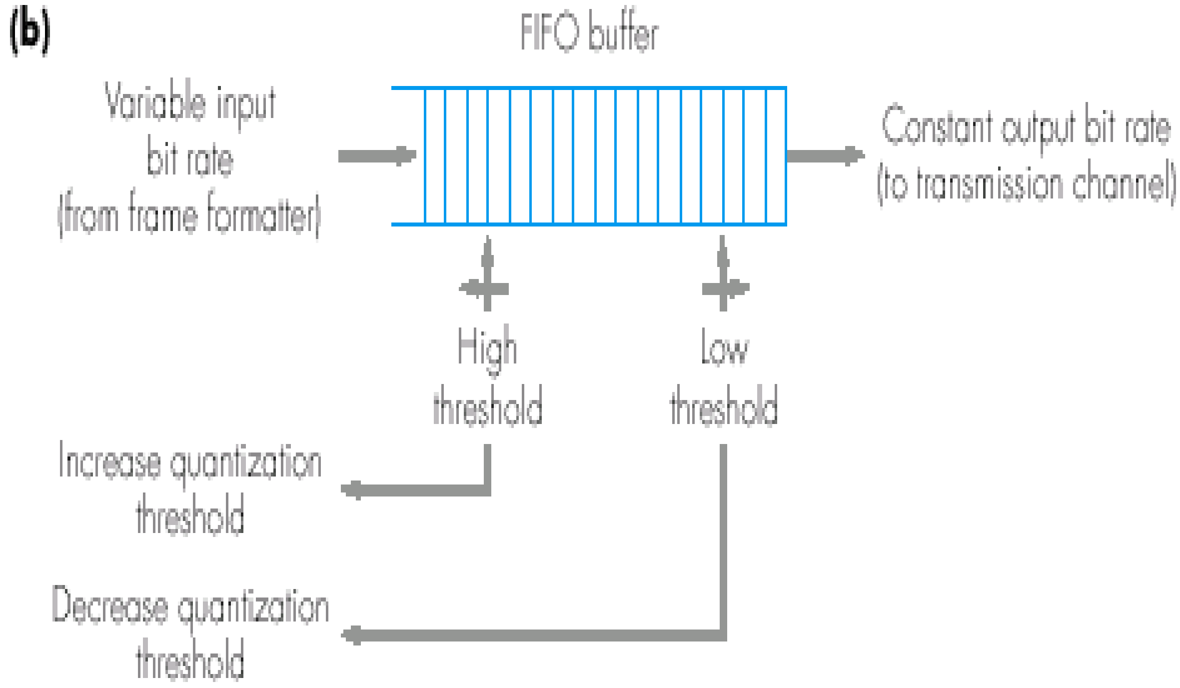
GOB 12
GOB start code
Group number
Quantization parameter
MB
MB

MB





Video encoder principles



Q9a) Solution :

Ethernet networks – and the more recent derivative IEEE802.3 – are used extensively in technical and office environment

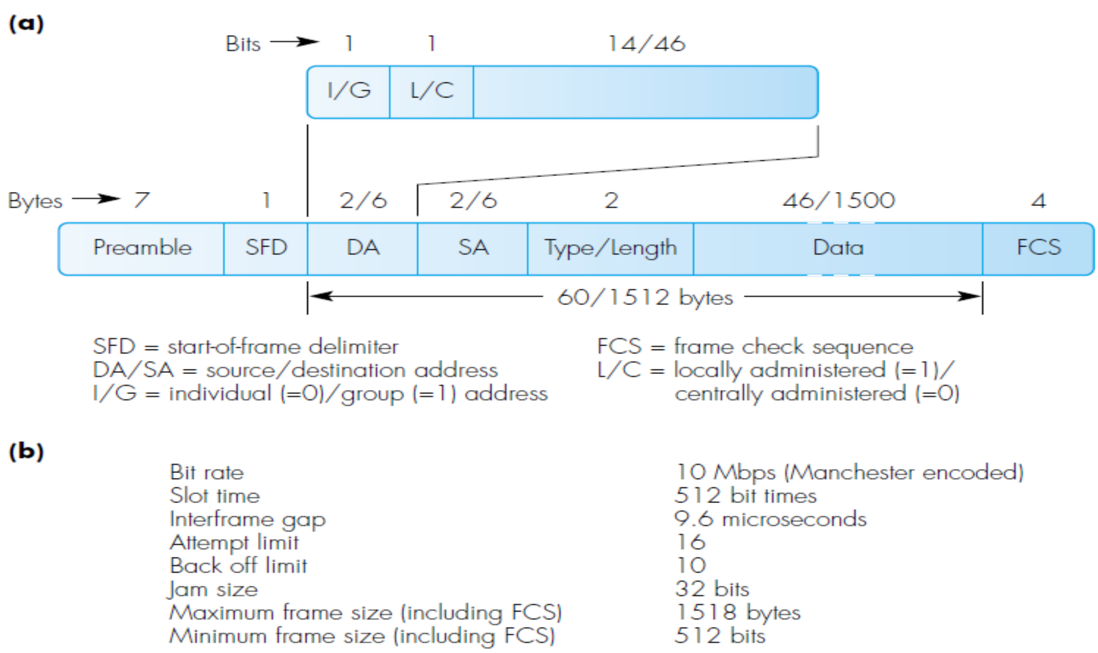
CSMA/CD

All the stations are attached directly to the same cable/bus ,it is said to operate in a multiple access mode

The bus operates in the broadcast mode which means that every frames transmitted is received by all the other stations that are attached to the bus

Because of the broadcast mode ,this will result in the contents of the two frames being corrupted and a collision is said to have occurred.

Figure 8.3 Ethernet/IEEE802.3 characteristics: (a) frame format; (b) operational parameters.



Frame format :

Preamble field- sent at the head of all frames, for synchronization of bits

Start of frame delimiter after preamble, single byte and informs the valid frame start.

Destination and source address – MAC address as used by MAC layer

First bit in the destination address specifies the address is individual or group address

Type of grouping is specified in second bit and can be locally or centrally administered

Group address is used for multicasting

Two byte type field indicates network layer protocol

Length field indicates number of bytes in the data field

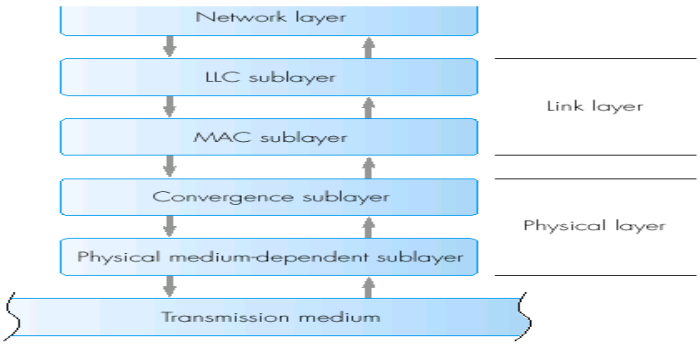
Maximum size of data field – MTU (Maximum Transmission Unit)

FCS – Frame Check Sequence used for error detection.

Q9b) Solution :

Figure 8.31 LAN protocols: (a) protocol framework; (b) examples.

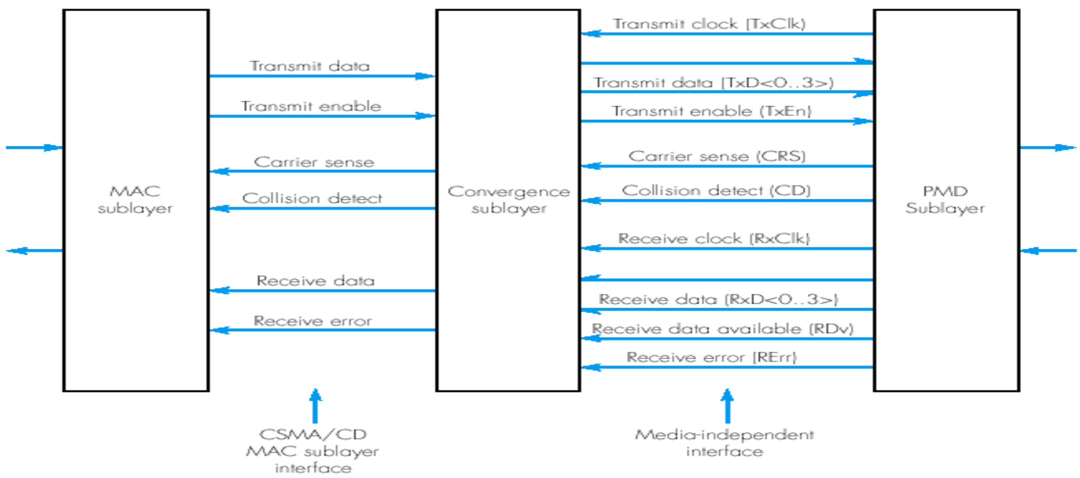
(a)



(b)

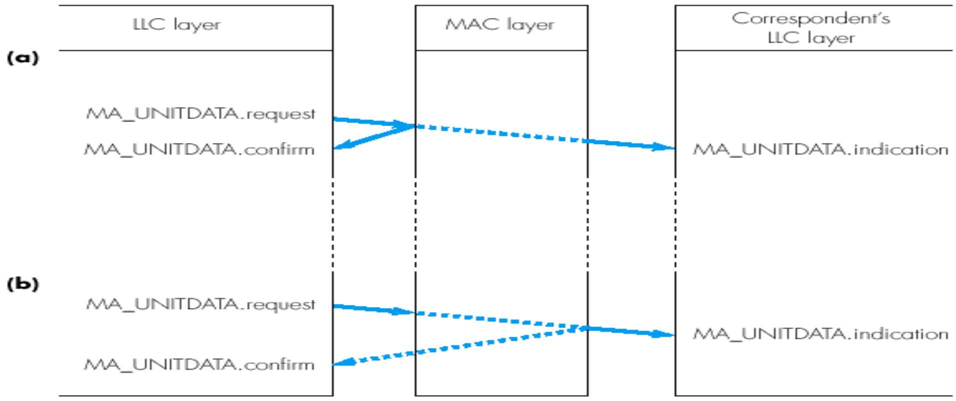
IEEE	802.1	Station management
	802.1d	Transparent bridges
	802.2	Logical link control (LLC)
	802.3	CSMA/CD (Ethernet) bus
	802.3u	Fast Ethernet
	802.3x	Hop-by-hop switch flow control
	802.3z	Gigabit Ethernet
	802.5	Token ring

Figure 8.32 Fast Ethernet media-independent interface.



Physical layer has been divided in to 2 sub layers. Physical medium dependent PMD and convergence sub layer CS. MMI media independent interface use between these two layers. CS role is to make the use of different media types transparent to the MAC sub layer.

Figure 8.33 MAC user service primitives: (a) CSMA/CD; (b) token ring.



Standard se of user service primitives are

MA_UNITDATA.request - includes required destination address, service data unit and the required class of service

MA_UNITDATA.indication

MA_UNITDATA.confirm - includes a parameter that specifies a success or failure of data primitive.

Q10b) Solution :

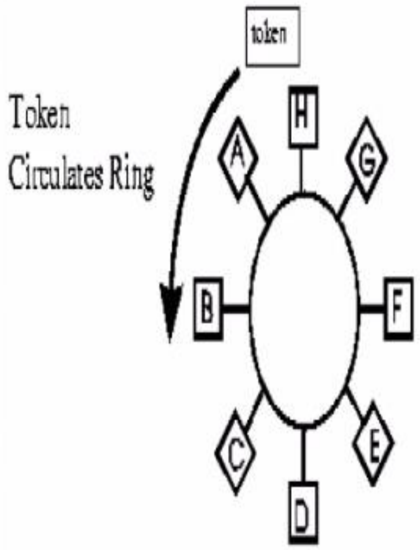
IEEE 802.5 TOKEN RING

Proposed in 1969 and initially referred to as a Newhall ring.

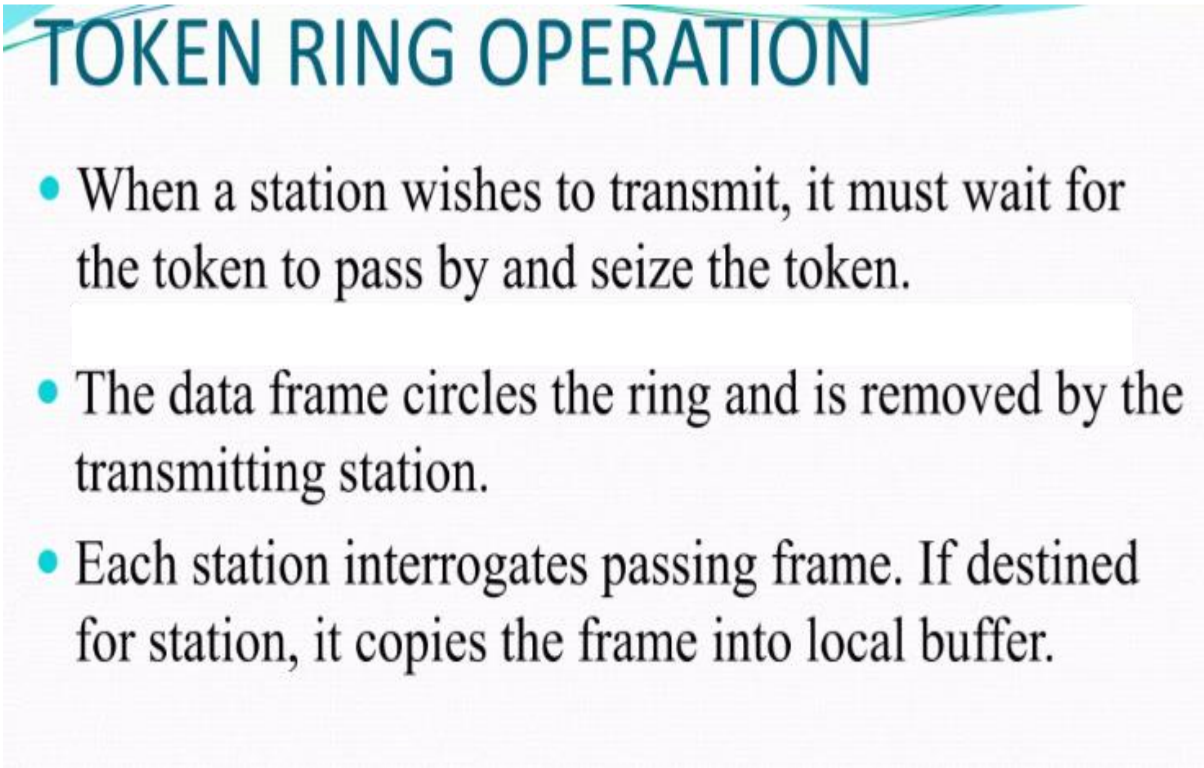
Token ring :: a number of stations connected by transmission links in a ring topology. Information flows in one direction along the ring from source to destination and back to source. Can both be implemented using star as well as ring topologies but basically it uses ring topology logically and star topology physically.

Medium access control :: is provided by a small frame, the token, that circulates around the ring when all stations are idle. Only the station possessing the token is allowed to transmit at any given time.

IEEE 802.5 TOKEN RING



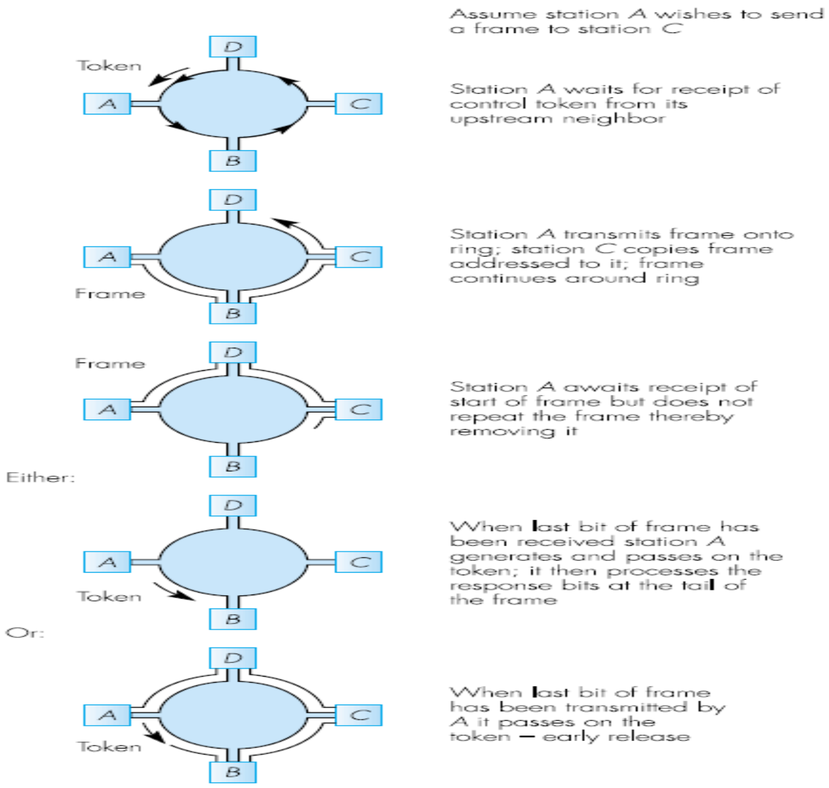
- There is a point to point link between stations that form a ring.
- Physical Layer Topology: *Ring*
  - Stations connected in a loop
  - Signals go in only one direction, station-to-station
- In a token ring a special bit format called a token circulated around all the stations.



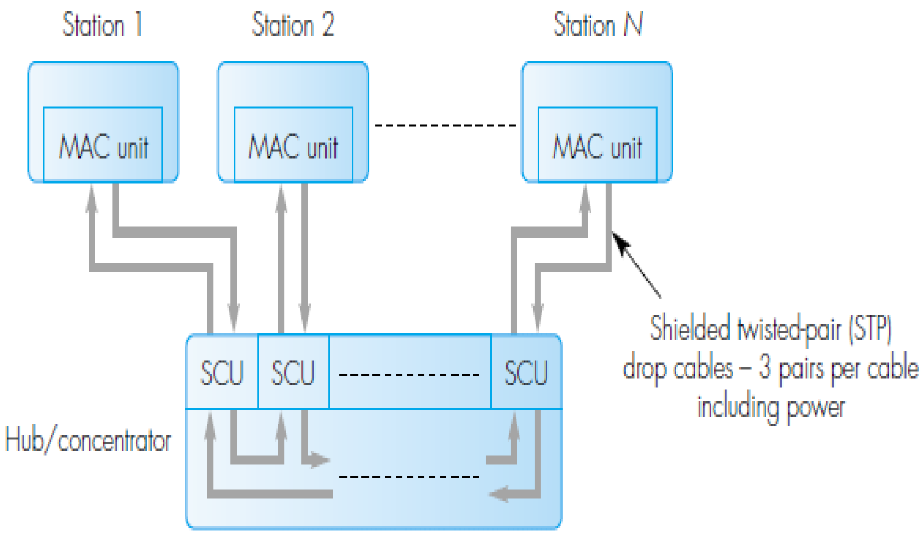
All the stations are connected together by a set of unidirectional links in the form of a ring and all frame transmissions between any of the stations take place over it by circulating the frame around the ring

Only one frame transfer can be in progress over the ring at a time

Fig 8.5


Assume station A wishes to send a frame to station C
Token
D
A
C
B
Station A waits for receipt of control token from its upstream neighbor
Station A transmits frame onto ring; station C copies frame addressed to it; frame continues around ring
Frame
Frame
Station A awaits receipt of start of frame but does not repeat the frame thereby removing it
Either:
When last bit of frame has been received station A generates and passes on the token; it then processes the response bits at the tail of the frame
Token
Or:
When last bit of frame has been transmitted by A it passes on the token — early release
Token

(a)


Station 1
Station 2
Station N
MAC unit
MAC unit
MAC unit
Shielded twisted-pair (STP) drop cables – 3 pairs per cable including power
SCU
SCU
SCU
Hub/concentrator

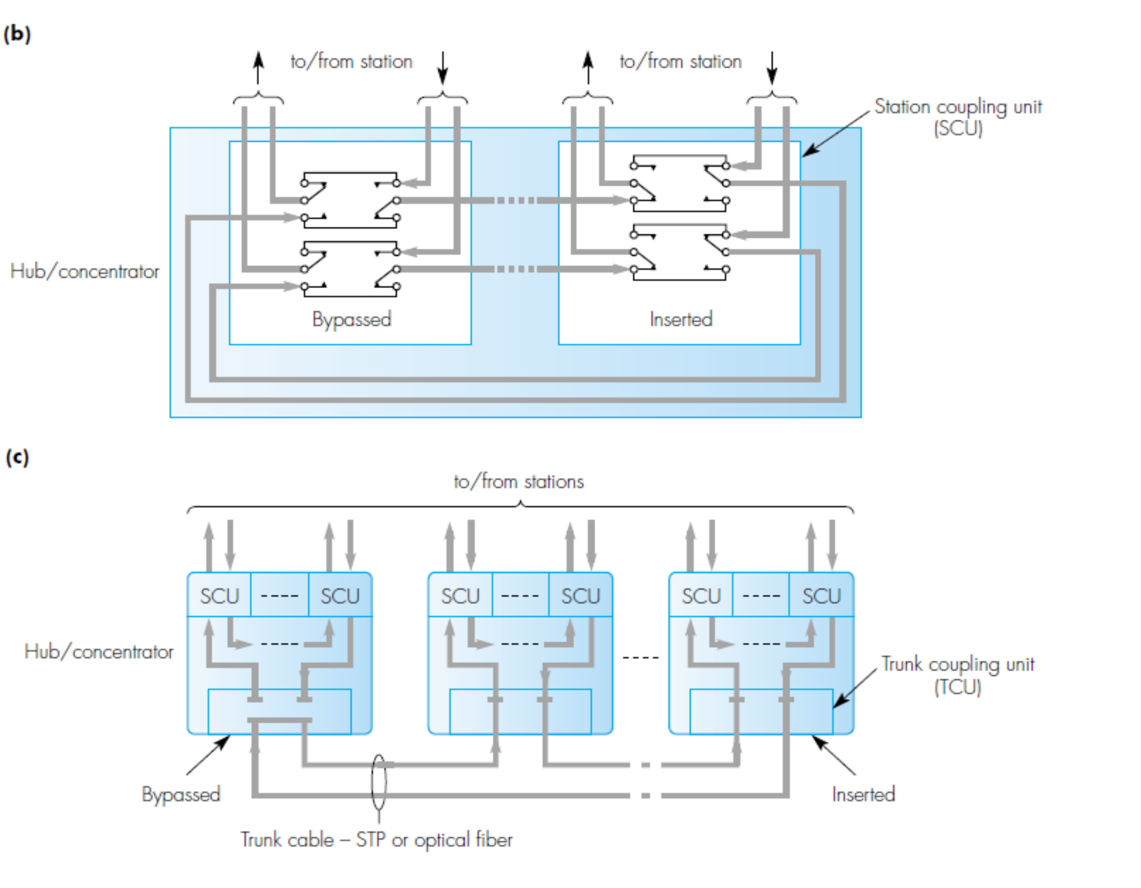


Figure 8.6 Token ring wiring configurations: (a) single hub; (b) station coupling unit; (c) multiple hubs/concentrators.

Concentrated / Single Hub – all stations attached by two twisted pairs

SCU- Station Coupling Unit – interconnected to form a unidirectional ring

It contains relay and driving circuits

When a station is switched off, SCU is in a bypass state and a transmission path exists through the SCU

When the station is switched on, insertion of a station into the ring is initiated

In this state, all signals are routed through MAC unit of the station, the received signal is sent to the transmit site if the station is not the originator of the ring or remove the received signal from the ring if it initiated transmission.

Two pairs of relays – to detect open ckt or short ckt faults

In bypassed state, MAC can conduct self test

Any data o/p on the transmit pair is looped back to receive pair

SCU is connected to STATION using three twisted pair, for transmission, reception and supply power

For larger configuration – connect multiple nodes/concentrators by STP/ Optical fibre

TCU-Trunk coupling unit – second relay unit

MAC unit does frame encapsulation, de-encapsulation, FCS generation, error detection and implementation of MAC algorithm.

Master station - active ring monitor, supplies clock for the ring

Active monitor selection – bidding process

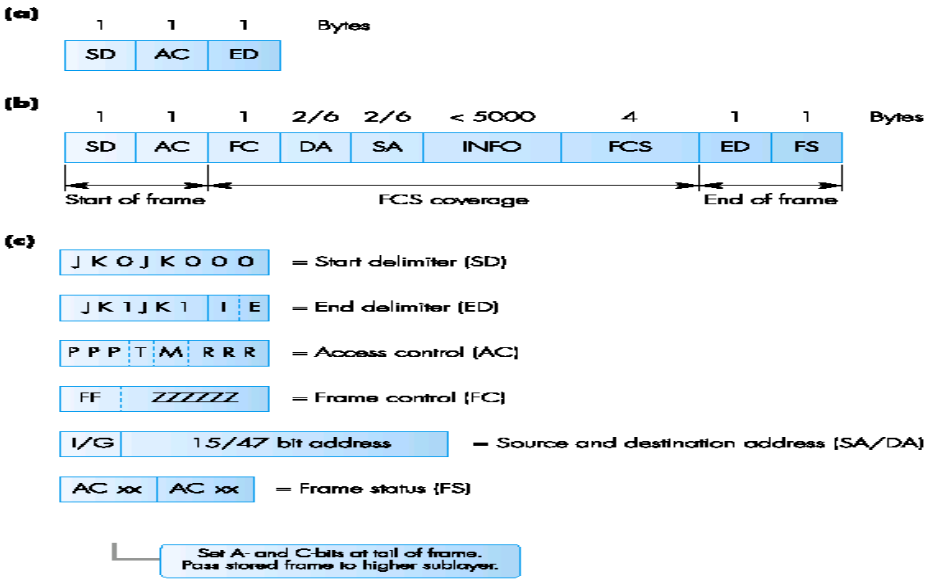
Active monitor ensures min latency time

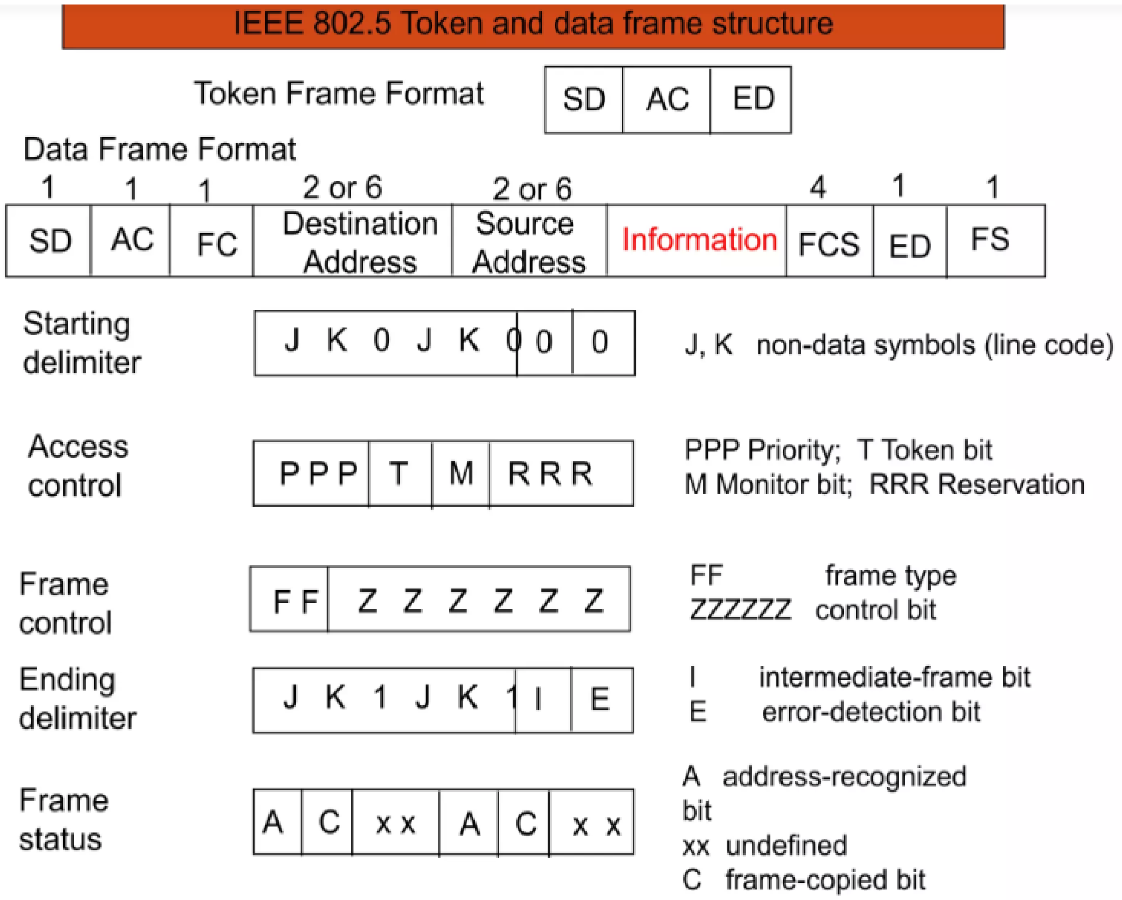
Ring should have min latency time to see that the token is not corrupted.

To maintain const ring latency additional elastic (variable) buffer with a length of 6 bits is added to the fixed 24 bit buffer

The buffer bits are altered based on the received signal.

Figure 8.7 Token ring network frame formats and field descriptions: (a) token format; (b) frame format; (c) field descriptions.





Two types of formats – Normal token and Control Token

Control Token – Right to transmit is passed from one station to other

Normal Frame – used by station to send data or MAC information round the ring

SD & ED – bit sequences for data transparency

The J & K symbols follow other than normal encoding. J symbol has same polarity as the preceding symbol. K symbol has opposite polarity to the preceding symbol

In token both I & E are Zero

In normal frame, I is used to indicate whether the frame is first or last and E is for error detection.

AC field consists of priority bits, token , monitor bits and reservation bits.

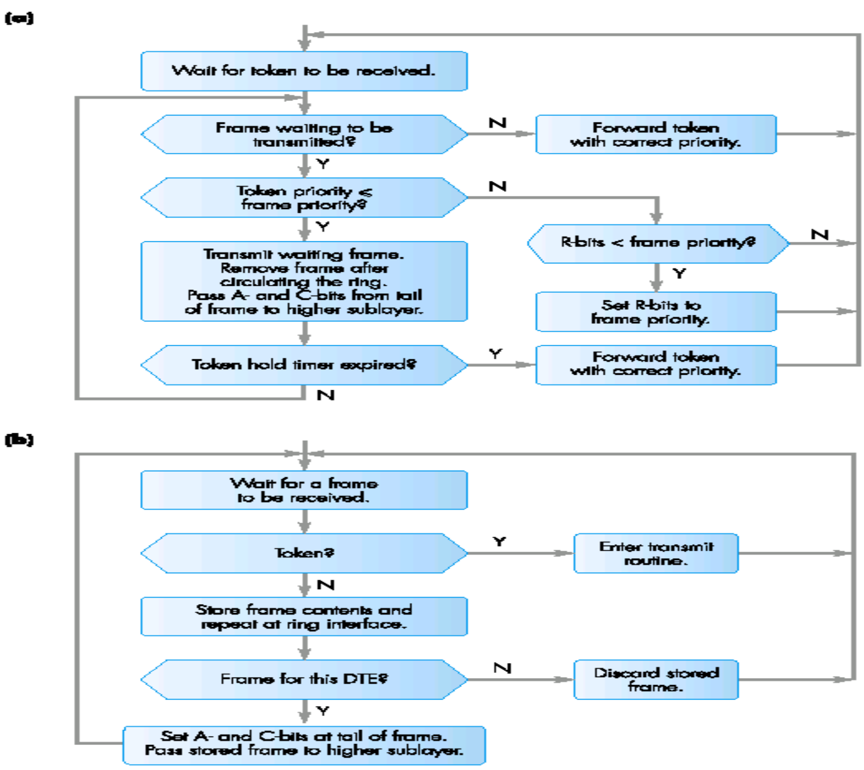
FC field defines type of frame and other control functions

INFO field use to carry user data or control information

FCS – for error checking

FS – consists of two fields, the address recognized bits A and the frame copied bits C.

Figure 8.8 Token ring MAC sublayer operation: (a) transmit; (b) receive.



Priority Operation:

Frames with higher priority than the current ring service priority are always transmitted on the ring first

All stations holding frames with the same priority have equal access rights to the ring

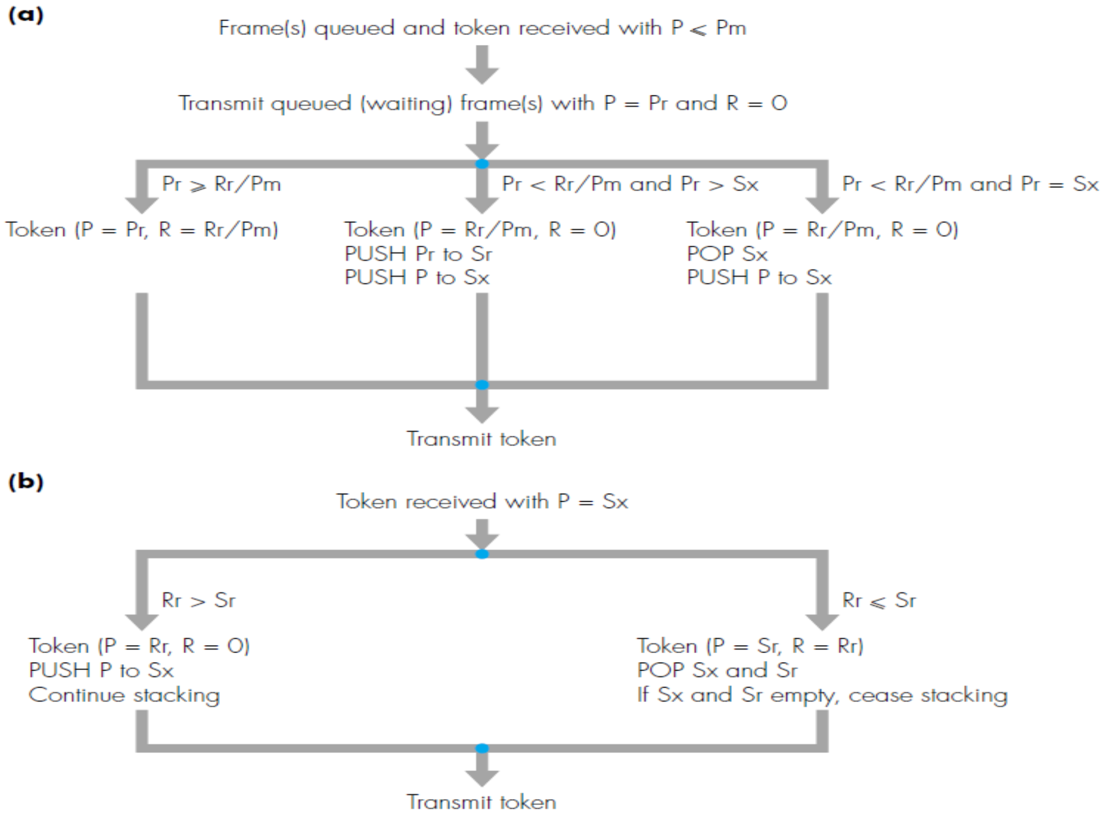
Two bits R & P are used in this operation

Each MAC unit has two sets of values. First set with three variables, Pm, Pr and Rr

Pm – Specifies highest priority values within the frame

Pr & Rr – priority registers

The second set of values consists of two stacks Sr and Sx



Ring Management:

Connected with transmission of frames and tokens during normal operation of the ring and defining initialization procedure for the station which joins an already operation ring

Initialization: A station added to the ring enters an initialization sequence and conforms that no stations in the ring are using the same address and also informs its entry to the ring

Stand by Monitor: To monitor the correct operation of the ring. Monitors passage of tokens and special active monitors present.

Active Monitor: An active monitor station inserts its latency buffer and enables its own clock. It ensures that there are no other tokens or frames on the ring before it initiates transmission of a new token

Beaconing: This is to inform each station , the suspension of token passing protocol.