Multimedia Communication

18EC743

Feb/Mar. 2022

Max. Marks: 100

Time: 3 hrs.

Solution and Scheme prepared by

Prof. RAGHAVENDRA. NAGARALLI.

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Module -1

1. a). Explain the communication modes available to transfer the information stream. → (10M).

Ans The transfer of the information streams associated with an application can take place in one of 5 modes.

i) Simplex: This means the information associated with the application flows in one direction only.

eg:- is the transmission of photographic images from a deep-space probe.

$$A \longrightarrow B$$

—2M

2) Half-Duplex: This means that information flows in both directions but alternatively. This mode is also known as two-way alternate.

eg:- User making a request for some info. from a server and the latter returning the requested info.

—2M

3) Duplex: This means that info. flows in both direction simultaneously. It is also known as two-way simultaneous.

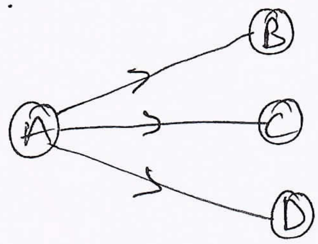
eg:- Two way flow of digitized speech and video associated with a video telephony application.

$$A \rightleftarrows B$$

—2M

4) Broadcast: This means that the info. output by a single source node is received by all the other nodes - computer and so on that are connected to the same network.

eg:- Broadcast of the television program over a cable network as all the television receivers that are connected to the n/w receive the same set of programs.



← 2M.

5) Multicast : This is similar to a broadcast except that the info. output by the source is received by only a specific subset of the nodes that are connected to the N/w.

eg:- video conferencing which involves a predefined group of terminals/computers connected to a N/w exchanging integrated speech and video streams.

— 2M.

1.b). Explain i) Data network and ii) Broadband multiservice network in detail with suitable figures.

→ (10M)

Ans. i) Data Networks

Data networks were designed to provide basic data communication services such as electronic mail (email) and general file transfers.

The two most widely deployed networks of this type are the X.25 network and the Internet.

The Internet is made up of a vast collection of interconnected networks all of which operate using the same set of communication protocols.

Access to the Internet is through an intermediate Internet service provider (ISP) N/w.

②

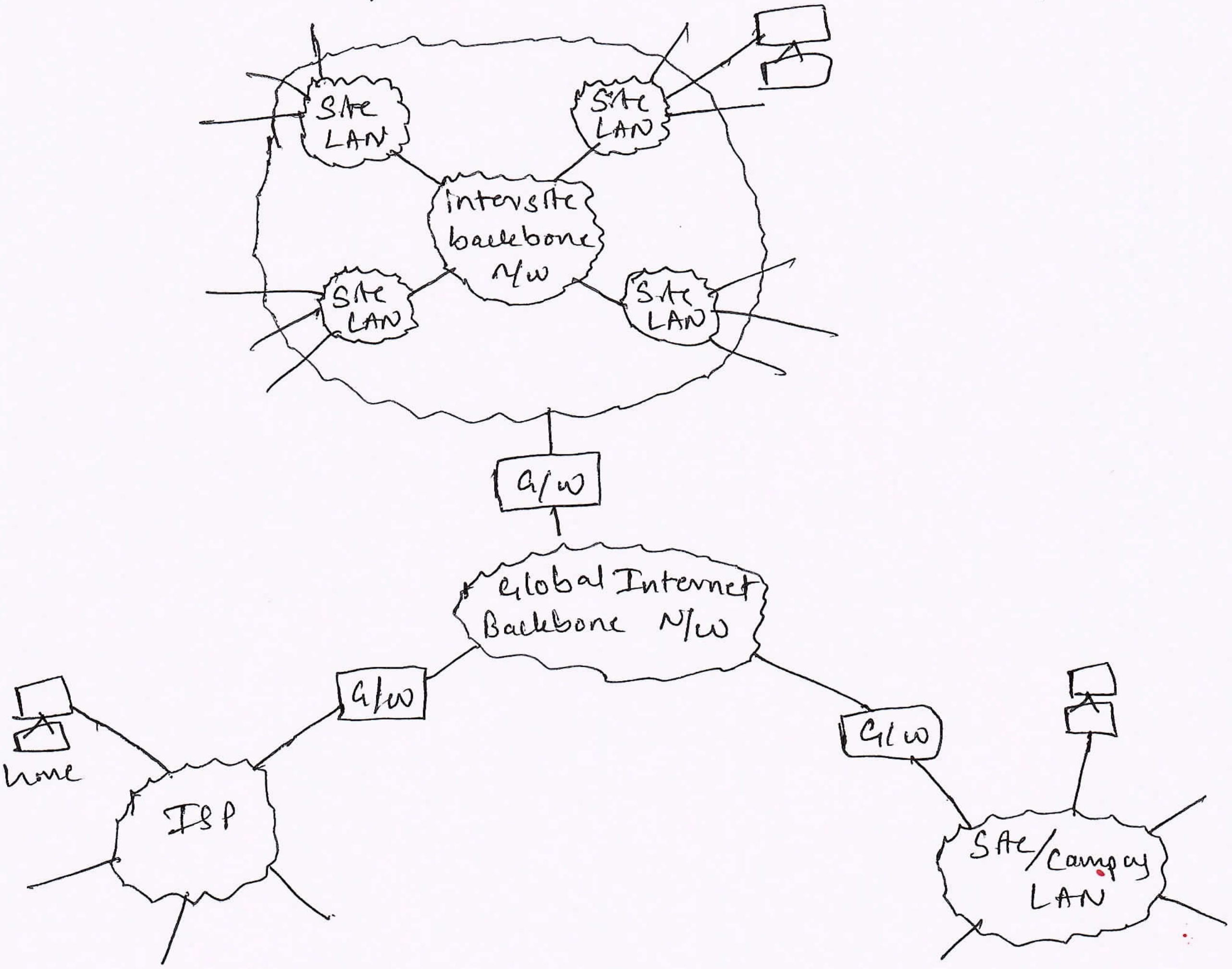
The user devices are connected to the ISP N/w either through a PSTN with modems or through an ISDN.

In case of single site, the N/w is known as local area N/w (LAN). For an wide N/w comprising multiple sites, the sites are interconnected together using an intersite backbone N/w.

The enterprize N/w is known as intranet

The diff. types of N/w are all connected to the Internet-backbone network through gateway, which is known as router.

All data N/w operate in what is called a packet mode.



—5M.

③

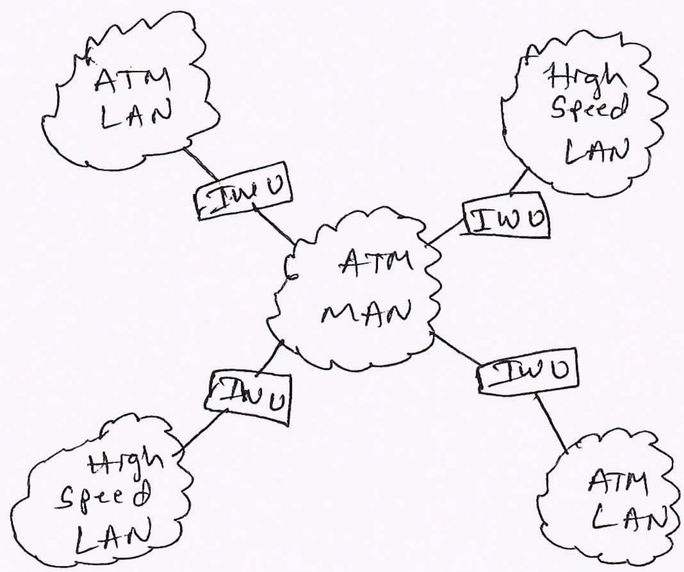
ii) Broadband multiservice N/w.

The term broadband was used to indicate that the circuits associated with a call could have bit rates in excess of the maximum bit rate of 2Mbps - 30 X 64kbps. provided by an ISDN.

Multiservice N/w implies that the N/w must support multiple services.

Diff. multimedia applications require diff. bit rates, the rate being determined by the types of media that are involved. Hence the switching and transmission methods that are used within these N/w must be more flexible.

The rate of transfer of data through N/w varies with diff. media. Hence this mode of transmission is known as asynchronous transfer mode (ATM). Therefore these N/w also known as ATM N/w.



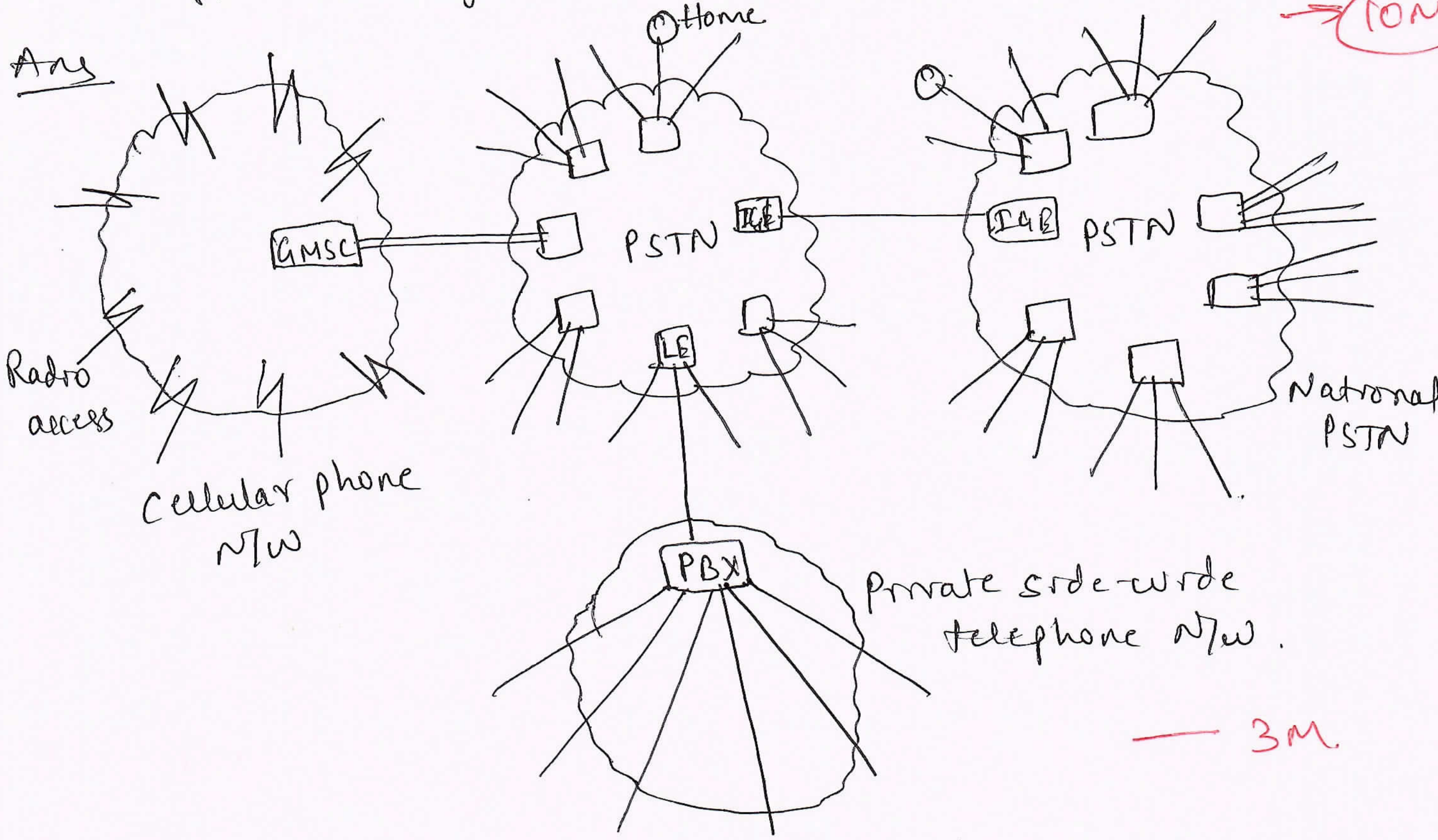
④

— 5M

OR

2.a). Explain with the aid of the diagram, how a PSTN can support range of multimedia common applications. → 10M.

Ans



— 3M.

PSTN have been in existence for many years. and have gone through many changes.

The term switched is used to indicate that a subscriber can make a call to any other telephone that is connected to the total N/W.

Telephones located in the home or in a small business are connected directly to their nearest local exchange/end office. Those located in a medium or large office/site are connected to a private switching office or PBX. The PBX provides a switched service between any two telephones that are connected to it. The PBX is connected to its nearest local exchange More recently, cellular phone N/w have been introduced which provide a similar service to mobile subscribers by means of handsets.

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The switches used in a cellular phone N/w are known as Mobile switching centers (MSC's)

Finally, international calls are routed to and switched by International gateway exchanges (IGE).

Telephone N/w operate in what is called a circuit mode. The access circuits that link the telephone handsets to a duration of the call. to PSTN or PBX were designed.

Although within a PSTN all the switches and the transmission circuits that interconnect them now operate in a digital mode, ~~to~~ to carry a digital signal a stream of binary 1's and 0s - over the analog access circuits requires a device known as a Modem.

→ 7M

2b). Explain in brief interactive applications over internet → (10M)

Ans Internet is used to support a range of interactive applications, the most widely used being for interactions with a WWW or simply web server.

This comprises a linked set of multimedia information servers that are geographically distributed around the internet.

Each document comprises a linked set of pages and the linkages between the page are known as hyperlinks. These are pointers or references, either to other pages of the same document or to any other document within the total web.

The linkage points within documents are defined by the creator of the document and are known as anchors.

— 5M

6

Documents comprising only text are created using what is called hypertext, while those comprising multimedia information are created using what is ~~known~~ known as hypermedia.

Each document has a unique address known as uniform resource locator or URL.

The 1st page of a document is known as the home page and all the hyperlinks on this and the other pages have similar URLs associated with them.

A standard format is used for writing document. This is known as the Hypertext Markup language (HTML) and its also used for writing client software to explore the total contents of the web. The client function is called a browser and there are number of user-friendly browsers.

-5M.

Module - 2

3 a). Illustrate the different types of text data representation. → 10M

Ans. There are three types of text representation.

1) Unformatted text: This is also known as plaintext and enables pages to be created which comprise of strings of fixed-sized characters from a limited character set.

2) Formatted text: this is also known as richtext and enables pages and complete documents to be created which comprise of strings of characters of different styles, size and shape with tables, graphics and images inserted at appropriate points.

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3) hypertext: this enables an integrated set of documents to be created which have defined linkages between them. — 5M.

1). Two examples of character sets that are widely used to create pages consisting of unformatted text strings. i) Basic ASCII character set ii) supplementary set of mosaic characters.

ASCII is most widely used character sets which uses 7-bit binary codeword for each character.

2) An example of formatted text is that produced by most word processing packages. It is also used extensively in the publishing sector for the preparation of papers, books, magazines, journals etc. It enables documents to be created that consists of characters of diff. styles and of variable size and shape, each of which can be plain, bold or italicized. Other formatting options are supported to create chapters, sections and paragraphs, each with diff. headings and with tables, graphics and pictures inserted at diff. points.

3) Hypertext is a type of formatted text that enables a related set of documents, normally referred to as pages, to be created which have defined linkage points refered to as hyperlinks. between each other.

An example of hypertext language is HTML Associated with each link, is a unique network-wide name known as uniform resource locator. —5M.

8

3b). Describe the function of signal encoder with the associated waveform. → 10M

Ans The conversion of analog signal into a digital form is done by using encoder.

It consists of 2 main circuits, a bandlimiting filter and an analog to digital converter (ADC). the latter consists of sample and hold and a quantizer.

The role of filter is to remove selected higher freq. components from the source signal.

to represent amplitude of analog signal correctly, First the signal to be sampled at a rate which is higher than the maximum rate of change of the signal amplitude and secondly, the number of diff. quantization levels used to be as large as possible. — 2M.

Sampling rate :- Nyquist sampling theorem states that "in order to obtain an accurate representation of a signal, its amplitude must be sampled at a minimum rate that is equal to or greater than twice the highest sinusoidal freq. component that is present in the signal. This is known as Nyquist rate. →2M.

The bandlimiting filter is also known as antialiasing filter.

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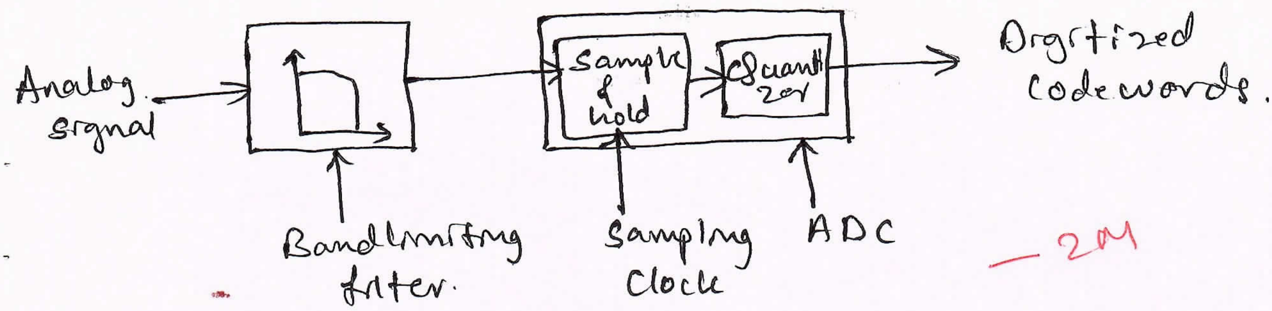
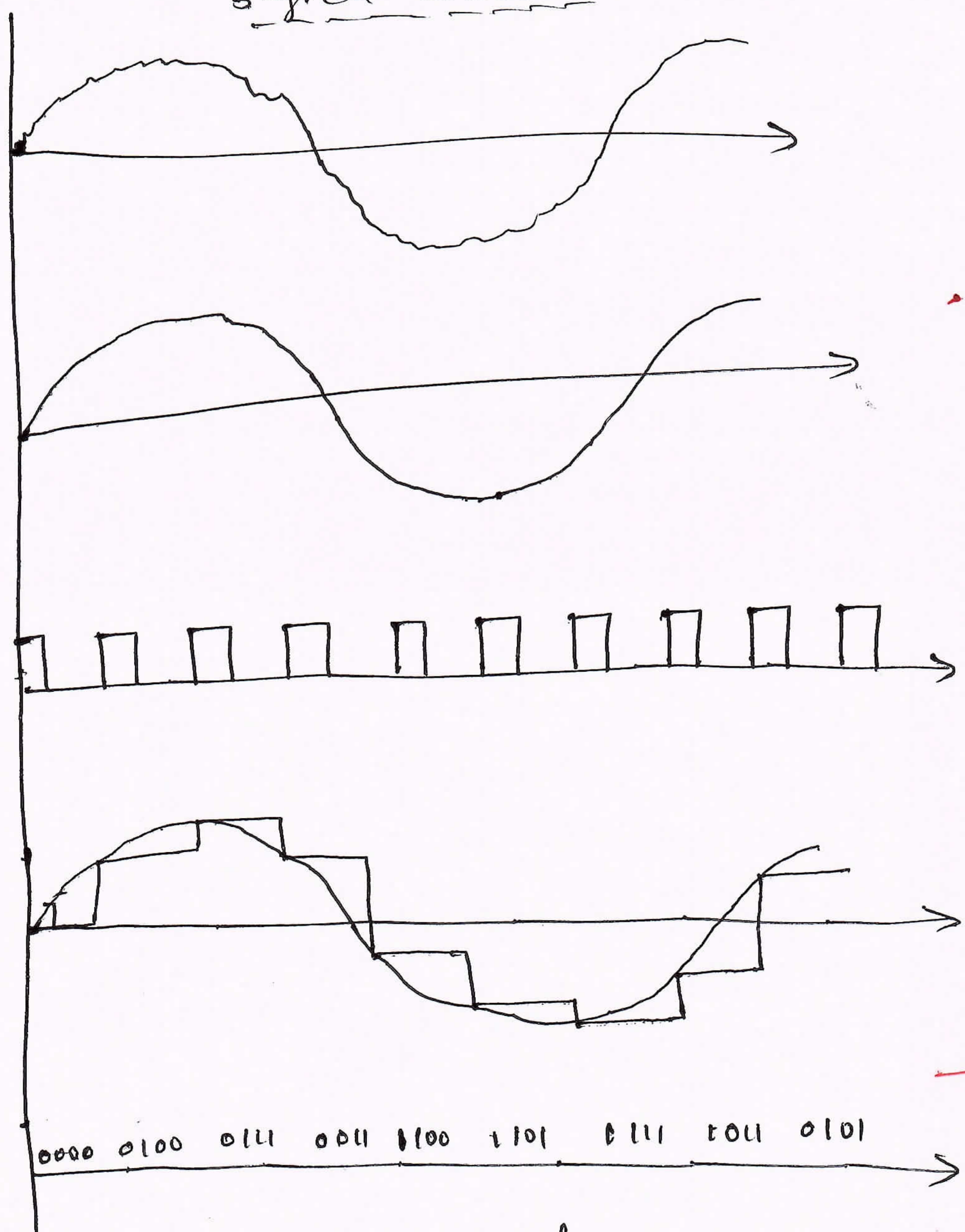


— 2M

signal encoder



— 4M.

Encoder waveform.

Quantization Intervals :- If Vmax is the max. positive and negative signal amplitude and n is no. of binary bits then magnitude of each quantization interval q is given by

$$q = \frac{2V_{max}}{2^n}$$

OR

4 a) Explain Raster-scan operation associated waveform. → (10M)

Ans The picture tubes used in most television sets operate using what is known as a raster-scan. This involves a finely-focussed electron-beam - the raster - being scanned over the complete screen.

Each complete scan comprises a number of discrete horizontal lines the 1st of which starts at the top left corner of the screen and the last of which ends at the bottom right corner.

At this point the beam is deflected back again to the top left corner and the scanning operation repeats in the same way. This type of scanning is called progressive scanning.

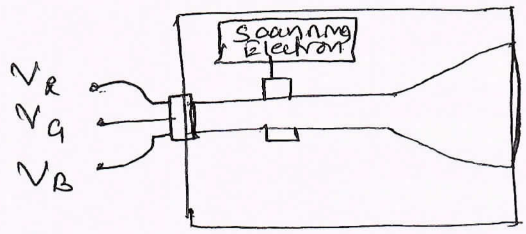
Each complete set of horizontal scan lines is called as frame.

Television picture tube were designed to display moving images. The persistence of light/color produced by the phosphor is designed to decay very quickly and hence it is necessary to continuously refresh the screen. — 3M

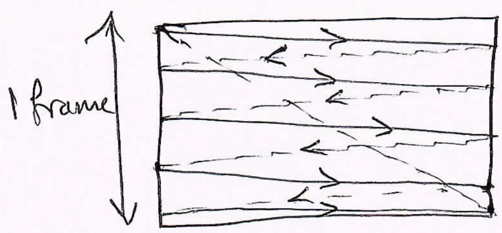
The frame refresh rate must be high enough to ensure the eye is not aware the display is continuously refreshed.

A refresh rate of at least 50 times per second is required.

The graphics program is used to create the high level version of the image interactively and the display controller part of the program interprets sequences of display commands and converts them into displayed objects. by writing the appropriate pixel values into the video RAM



CRT



raster-scan principles.

→ 4M

Pixel depth: The number of bits per pixel is known as the pixel depth and determines the range of different colors that can be produced.

Aspect ratio: Both the number of pixels per scanned line and the number of lines per frame vary. the actual number used being determined by what is known as the aspect ratio of the display screen.

→ 3M

4.b). Derive the bit rate and the memory requirements to store each frame that result from digitization of both 525 and 625-line system assuming a 4:2:2 format. Also find the total memory required to store a 1.5 hour movie/video

(10M)

Ans

525 line system.

$$Y = 720 \times 480$$

$$Cb = Cr = 360 \times 480$$

Bitrate : line sampling rate is fixed at 13.5MHz for Y and 6.75MHz for both Cb & Cr, all with 8 bits per sample.

$$\text{Bitrate} = 13.5 \times 10^6 \times 8 + 2(6.75 \times 10^6 \times 8)$$

$$= 216\,\text{Mbps}.$$

– 2½M.

$$\text{Memory required per line} = 720 \times 8 + 2(360 \times 8)$$

$$= 11520\ \text{bits}$$

$$\text{memory per frame} = 480 \times 11520$$

$$= 5.5296\ \text{M bits}$$

memory to store 1.5hr video

$$= 691.2 \times 60 \times 1.5 \times 3600\ \text{kbytes}$$

$$= 223.9488\ \text{G bytes}.$$

– 2½M

625 line system.

$$Y = 720 \times 576$$

$$Cb = Cr = 360 \times 576.$$

$$\text{Bitrate} = 13.5 \times 10^6 \times 8 + 2(6.75 \times 10^6 \times 8) = 216\,\text{Mbps}$$

– 2½M

$$\text{memory per frame} = 576 \times 11520 = 6.63555\ \text{Mbits}$$

$$\text{memory to store 1.5 hr video} = 829.44 \times 50 \times 1.5 \times 3600$$

$$= 223.9488\ \text{Gbytes}.$$

– 2½M

Module - 3

5 a). Give a brief description of the 5 main stages associated with the baseline mode of operation of JPEG. → (10M)

Ans There are 5 main stages associated with baseline mode, they are

1) Image / block preparation
2) Forward DCT
3) Quantization
4) Entropy encoding
5) Frame building.

1) Image / block preparation

The source image is made up of one or more 2D matrices of values. If it's monochrome image, just a single 2-D matrix is required. For a color image, if a CLUT is used just a single matrix is required. If the image is represented in R, G, B format, then 3 matrices are required. Also optionally Y, Cb, Cr form of representation can be used. — 2M.

2) Forward DCT

Each pixel value is quantized using 8-bits to produce a value in the range 0 to 255. For R, G, B or Y, the values in the range of -128 to +127 for the chrominance values Cb & Cr. If i/p 2D matrix is represented by P[x, y] and transformed matrix by F[i, j]. the DCT can be given as

$$F[i,j] = \frac{1}{4} C(i)\, C(j) \sum_{x=0}^{7} \sum_{y=0}^{7} P[x,y] \cos\frac{(2x+1)i\pi}{16} \cos\frac{(2y+1)j\pi}{16}$$

— 2M

Quantization

The main source of info. loss occurs during the quantization and entropy encoding stages where compression takes place.
If the magnitude of a higher freq. co-efficient is below a certain threshold the eye will not detect it. This property is exploited in the quantization phase by dropping those spatial freq. coefficients in the transformed matrix whose amplitudes are less than a defined threshold value.
— 2M

Entropy encoding

This comprises 4 steps
1) Vectoring: Representing values in the form of a single-dimension vector
2) Differential encoding: only the difference in magnitude of the DC coefficient in a quantized block relative to the value in the preceding block is encoded
3) Run-length encoding: The AC coefficients are encoded using this encoding procedure.
4) Huffman encoding: A table of codewords is used with the set of codewords precomputed using Huffman coding.
— 2M

Frame building

The role of frame builder is to encapsulate all the info. relating to an encoded image in a proper format
— 2M

(15)

5b). Explain CPU management and memory management in multimedia operating systems.

→ (10M)

Ans

CPU Management

* Real time processing can be achieved through efficient real-time scheduling

→ In the context of continuous media, a deadline can be the acceptable playback time of each frame

→ The challenges of multimedia scheduling are due to 2 conflicting goals: non real time process and real time processes.

→ The most important real-time scheduling approaches include Earliest Deadline First (EDF) and rate monitoring scheduling.

— 5M

Memory Management

→ The memory manager allocates memory to processes. Continuous media data is typically very large in size and requires stringent timing requirements.

→ One solution is to avoid swapping and to lock continuous media data in memory during the process.

→ Other important practical implementation techniques include using scatter buffers and passing pointers

— 5M

OR

6 a) 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 the probability of each characters are
A & B = 0.25, C and D = 0.14, E, F, G & H = 0.055
i) Use Shannon's formula to derive minimum average number of bits per character.
ii) Use Huffman coding to derive a codeword set and prove this is the minimum set by constructing the Huffman code tree. → 10M

Ans

i) $H = -\left(2(0.25 \log_2 0.25) + 2(0.14 \log_2 0.14) + 4(0.55 \log_2 0.55)\right)$

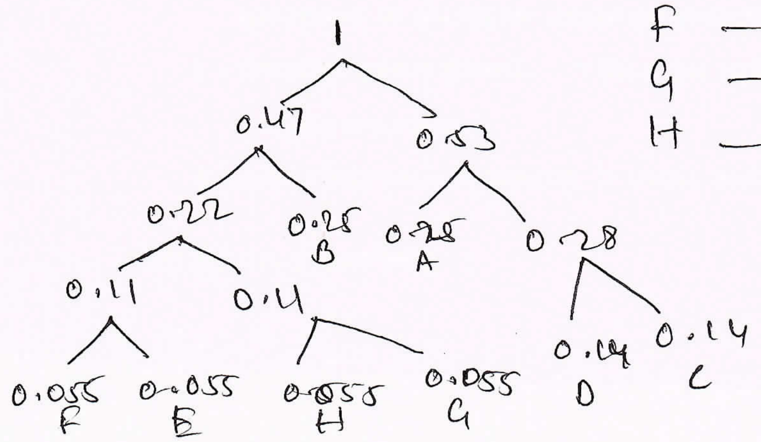
$H = 2.175$ bits per codeword — 3M.

ii)

A	0.25	0.25	0.25	0.25	0.28	0.47	0.53
B	0.25	0.25	0.25	0.25	0.25	0.28	0.47
C	0.14	0.14	0.14	0.22	0.25	0.25	
D	0.14	0.14	0.14	0.14	0.22		
E	0.055	0.11	0.11	0.14			
F	0.055	0.055	0.11				
G	0.055	0.055					
H	0.055						

A — 10
B — 01
C — 111
D — 110
E — 0001
F — 0000
G — 0011
H — 0010.

— 4M



→ 3M.

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6b). Define distributed multimedia system with neat block schematic and also highlight its features. → 10M

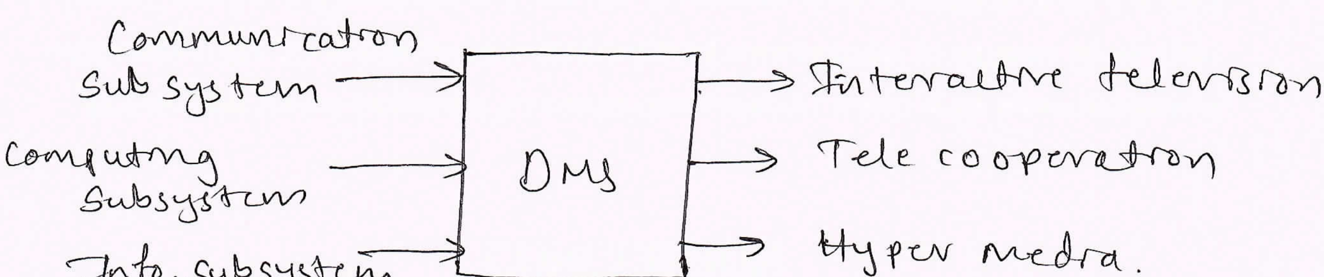
Ans DMS

→ A DMS is an integrated communication computing and information system that enables the processing, management delivery and presentation of synchronized multimedia information that the quality of service guarantees.

→ It integrates and manages the info. communication and computing subsystems to realize multimedia applications

→ An example of DMS is a number of multimedia PC's and/or workstations connected with continuous media servers using the internet.

→ The inputs of the system consists of the factors that derive a DMS from concepts to reality, and the output consists of a wide range of distributed multimedia applications. — 4M



— 2M

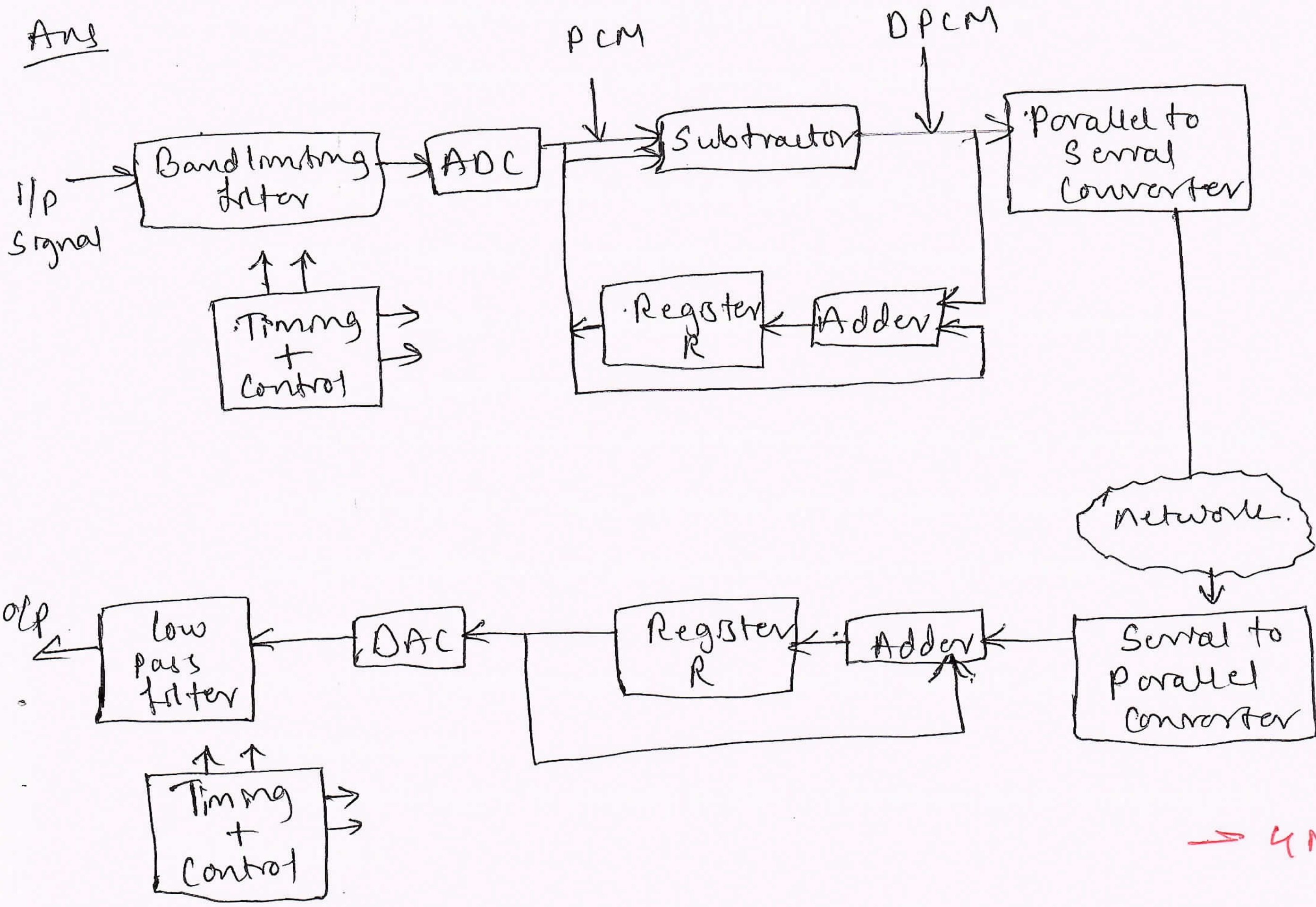
Main Features of DMS

1) Technology Integration
2) Multimedia Integration
3) Real time performance
4) System wide QOS support
5) Interactivity
6) Multimedia synchronization support
7) Standardization support. — 4M.

Module - 4

7. a). Discuss the principles of differential pulse code modulation with block diagram. → 10M

Ans



→ 4M

(19)

DPCM is a derivative of standard PCM and uses the fact that, 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.

Hence if only the digitized difference signal is used to encode the waveform then fewer bits are required.

—2M

A DPCM encoder and decoder are shown above The previous digitized sample of the analog input signal is held in the register, and the difference signal is computed by subtracting the current contents from the new digitized sample o/p by the ADC

—2M

The value in the register is then updated

The decoder operates by simply adding the received difference signal (DPCM) to the previously computed signal held in the register (PCM)

In another way. the difference signal is computed by subtracting varying proportions of the last three predicted values from the current digitized value output by ADC.

—2M

(20).

7.b) Explain principle of linear predictive coding with block schematic → 10M

Ans. An approach which involves the source simply analysing the audio waveform to determine a selection of the perceptual features it contains.

These are then quantized and sent and the destination uses them, together with a sound synthesizer, to regenerate a sound. This is the basis of the linear predictive coding (LPC).

The 3 features which determines the perception of a signal by the ear are its 1) pitch 2) period 3) loudness

In addition, the origins of the sound are important. These are known as vocal tract excitation parameters.

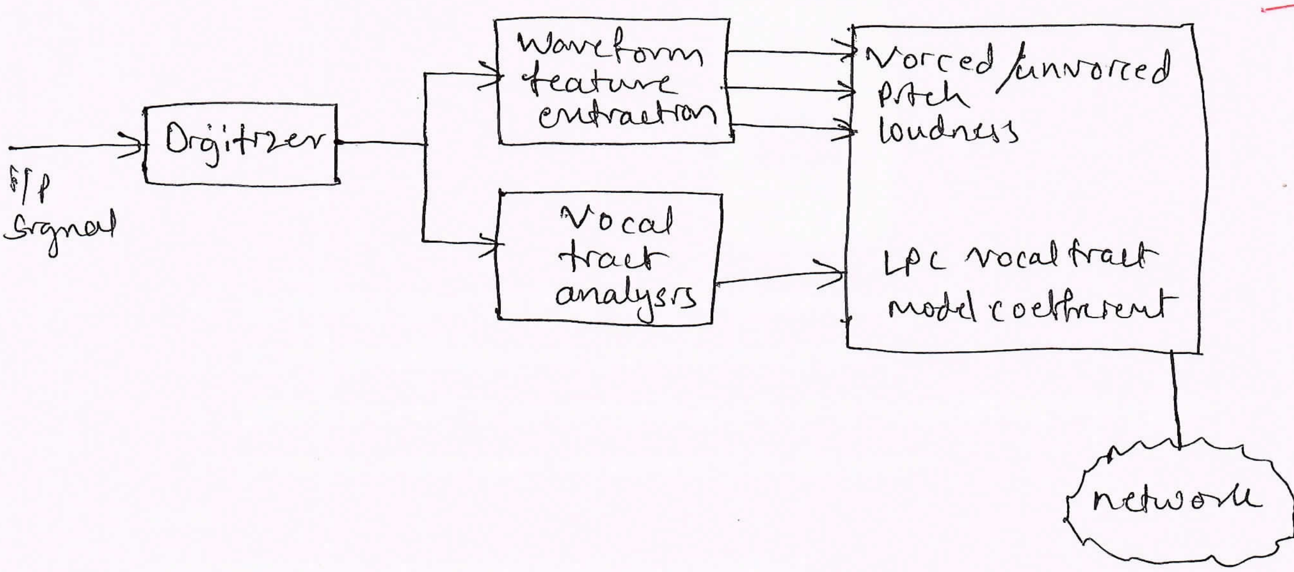
1) voiced sounds
2) Unvoiced sounds. — 3M.

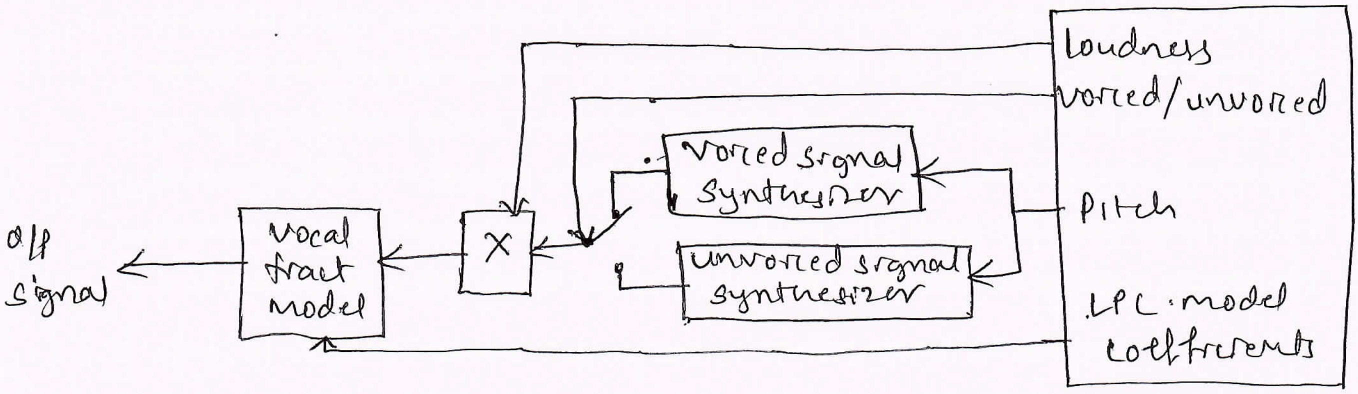
Once these have been obtained from the source waveform, it is possible to use them with a suitable model of the vocal tract, to generate a synthesized version of the original speech signal.

The input speech waveform is 1st sampled and quantized at a defined rate. A block of digitized samples known as a segment is then analyzed to determine the various perceptual parameters of the speech that it contains.

The speech signal generated by the vocal tract model in the decoder is a function of the present output of the speech synthesizer plus a linear combination of the previous set of model coefficient. Hence the vocal tract model used in adaptive and the encoder determines and sends a new set of coefficients for each quantized segment.

—3M





— 4M

Linear predictive coding

OR

8a) What are the video compression principles, explain with example frame sequence i) I & P frames ii) I-P-B frames iii) PB frames.

→ 10M

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Ans Video compression principles

The quality of the video used in various applications varies and is determined by the digitization format and frame refresh rate used.

The digitization format defines the sampling rate that is used for the luminance. Y and two chrominance Cb & Cr. signals and their relative position in each frame.

One approach to compressing a video source is to apply the JPEG algorithms to each frame independently. it is known as JPEG or MJPEG.

Only a small portion of each frame is involved with any motion that is taking place.

The technique that is used to exploit the high correlation between successive frames is to predict the content of many of the frames.

Instead of sending the source video as a set of individually compressed frames, just a selection is sent in this form and for the remaining frames, only the difference between the actual frame contents and the predicted frame content are sent.

Since the estimation ~~of~~ motion process is not exact additional info. must also be sent. This is known as motion compensation.

— 4M

Frame types

I - Frames : These are encoded without reference to any other frames

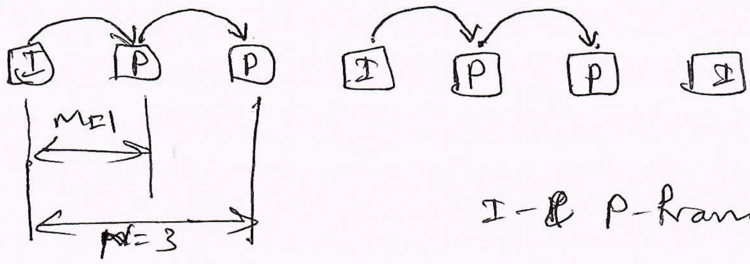
(28)

Each frame is treated as a seperate picture and Y, Cb and Cr matrices are encoded independently using JPEG algorithm — 2M

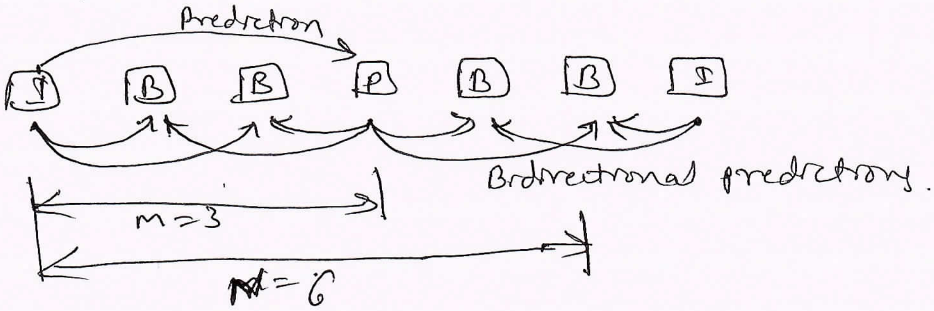
P-frames: The encoding of P frames is relative to the contents of either a preceding I-frame or a preceding P-frame.

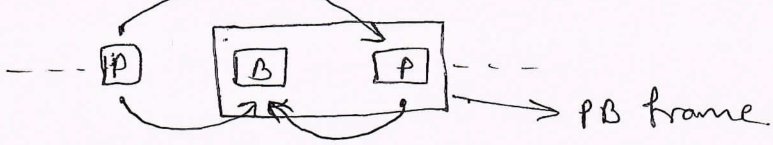
P frames are encoded using a combination of motion estimation and motion compensation and hence significantly higher levels of compression can be obtained with them. —

B-frames: Motion estimation involves comparing small segments of two consecutive frames for different and should a diff. be detected, a search is carried out to determine to which neighbouring segment the original segment has moved. — 2M



I- & P-frames only.





— 2M

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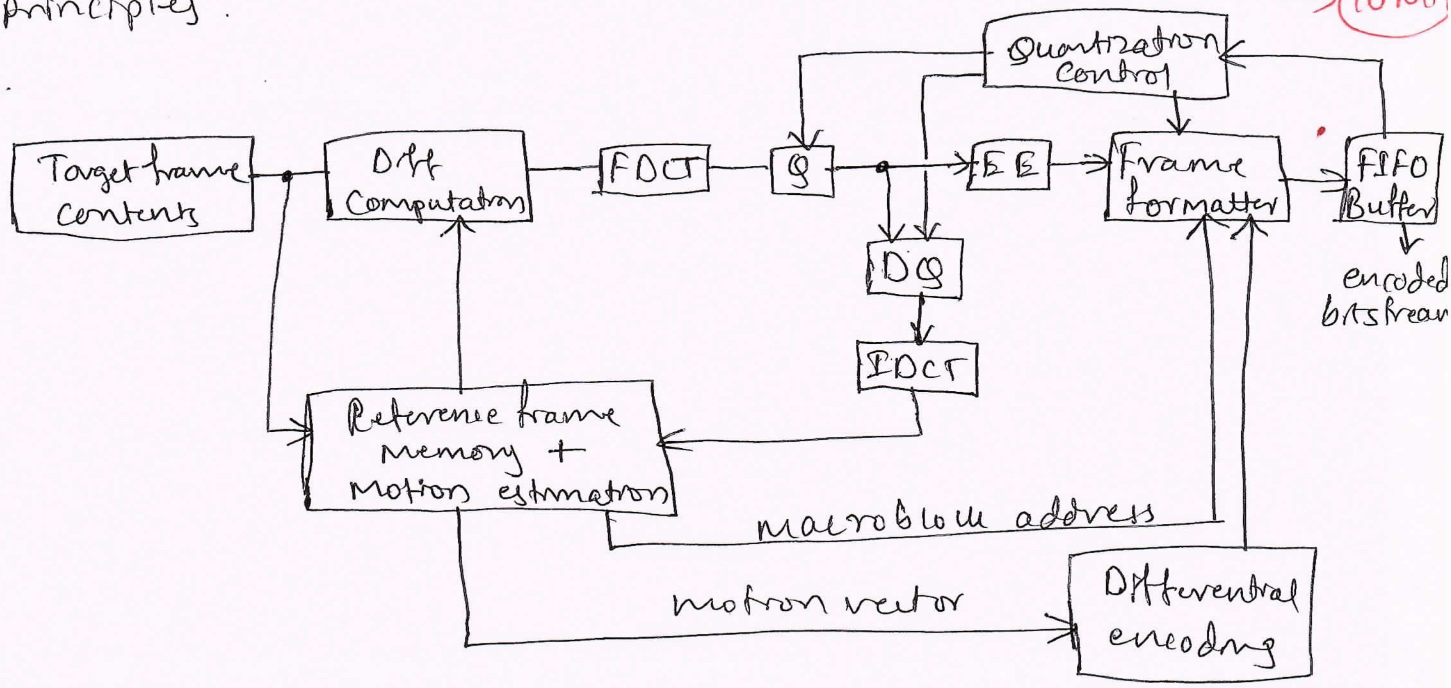
8b) Using Block diagram, explain H-261 video encoder principles. (10M)

Ans.



— 4M.

<u>H 261 video encoder principles implementation</u>

H 261 video compression standard has been defined by the ITU-T for the provision of video telephony and videoconferencing services over an ISDN.

The digitization format used is either the common Intermediate format (CIF) or quarter CIF (QCIF). Normally the CIF is used for video conferencing and QCIF for video telephony.

The O/P bitrate produced by the encoder is determined by the quantization threshold values that are used, the higher the threshold the lower the accuracy and hence the lower is the O/P bit rate.

It is possible to obtain a constant O/P bit rate from the encode by dynamically varying the quantization threshold used. This is the role of the FIFO buffer. — 3M

The order of the O/P from a FIFO buffer is the Same as that of I/P.

However since the O/P rate from the buffer is a Constant determined by the bitrate of the transmission Channel the I/P rate temporarily exceeds the O/P rate then the buffer will start to fill.

conversly if the I/P rate falls below the O/P rate then the buffer contents will decrease. In order to use this property, two threshold level is derived. the low threshold and the high threshold.
→ 3M

Module - 5

9a). what is a LAN? Explain LAN topologies and LAN media access methods. → 10M

Ans LAN's are used to interconnect distributed communities of end systems. Typically these are distributed around an office, single building or an enterprise.
—

The early LAN's operate using a shared, high bit rate, transmission medium to which all the stations are attached and the info frames relating to all calls are transmitted. — 2M

To ensure the transmission BW is shared fairly between all of the attached stations, a number of different medium access Control (MAC) methods are used

These include (CSMA/CD) and token ring. Both of which have a defined maximum number of attached stations and length of transmission medium associated with them.

In practice the maximum distance is relatively small and hence most LANs of this type comprise multiple LAN segments that are interconnected together using either repeaters or devices known as bridges and a high bit rate backbone subnetwork. Recently high bit rate versions known as legacy LAN's are available.

— 4M

Newer hubs operate in duplex mode and allow the frames relating to multiple calls to be transmitted concurrently. Examples include fast Ethernet hubs and Ethernet switching hubs.

In terms of the link layer protocol associated with LAN's, the various LAN types all use a standard LLC sublayer and there is a different MAC sublayer for each of the LAN types.

In multisite enterprise N/W, the LANs associated with the different sites are interconnected together using various methods determined by the volume of traffic involved.

— 4M

9 b). Explain the devices commonly used in LAN → 10M

Ans

The common network devices used in LAN are

1) Hub :- Hubs connect multiple computer networking devices together. A hub also acts as a repeater. It connects LAN components with identical protocol. Hubs do not perform packet filtering or addressing functions, they just send data packets to all connected devices. Hubs operate at the physical layer of OSI model

2) Switch :- A switch is a multiport device that improves network efficiency. The switch maintains limited routing info. Segments of LAN's are usually connected using switches. Switches can read the MAC address to pass the packets to the appropriate destination. A switch can work at either data link layer or the network layer of OSI model. — 3M

3) Routers :- Routers help transmit packets to their destination by routing through networking devices using network topologies. They store info about the networks they are connected to. using routing tables. Routers mainly work at network layer. Routers can also operate as packet-filtering firewalls and use access control lists. Routers also used to divide large networks into 2 or more subnetworks. They use various protocols for their operations

— 2M

4) Bridge :- Bridges are used to connect 2 or more hosts or network segments together. The basic role of bridges in network architecture is storing and forwarding frames between different segments. They use MAC addresses for transferring frames. Bridges work only at the physical and datalink layers of the OSI model. Bridges are used to divide large networks into smaller segments. They connect different LAN components.

5). Gateway :- Gateways normally work at the transport and session layers of the OSI model. Gateways connect two or more autonomous networks, each with its own routing algorithms, protocols, topology, domain name service etc...

— 3M

6). Modem :- Modems are used to transmit digital signals over analog telephone lines. Modems do modulation and demodulation operations.

7). Repeater :- It's a device that regenerate the weak signals to regain their shape. They work on the physical layer.

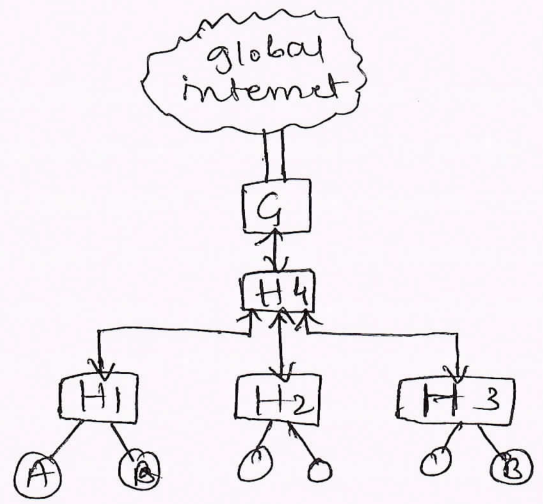
8). Access point :- It's commonly a wireless device which operate as a bridge connecting a standard wired network to wireless devices. They work ~~at the~~ data link layer of OSI model. wireless access point consists of a transmitter and receiver devices used to create a wireless LAN

— 2M

OR

10).a). Explain Address Resolution protocol. Briefly describe ARP functionality. → (10M)

Ans. Associated with each host are 2 addresses.
1) IP address 2) MAC address



— 2M

Example topology for describing operation of ARP.

In above figure 3 Ethernet hubs H1, H2 and H3 are interconnected to 4th hub H4. There is a connection from H4 and site gateway (G).

Associated with each ARP is a routing table known as the ARP cache.

This contains a list of the IP/MAC address-pair of those hosts with which host A has recently communicated, and when the host is 1st switched on, it contains no entries.

On receipt of the 1st datagram from the IP in host A, the ARP in A reads the destn IP address of B from datagram header and determines it is not in the cache. — 2M.

Hence it broadcasts an ARP request message in a broadcast frame over the LAN and waits

The request message contains both its own IP/MAC address pair and the IP address of destination.

This message is received by the ARP in all hosts attached to the LAN.

— 2M

The ARP in host B recognizes its own IP address in the request message and process it.

It 1st checks the entry of source address-pair is within its own cache. and if not, enters them.

The ARP in host B then responds by returning an ARP reply message to the ARP of host A.

On receipt of reply message, the ARP in host A 1st makes an entry in the cache and then passes the waiting datagram to LLC sublayer or to MAC sublayer.

Being on the same broadcast N/W, gateway receives a copy of all broadcast frames containing ARP request & reply messages.

On receipt of each message, the ARP of gateway checks its own cache for address-pair entry. If it is not, then it adds them to the cache.

— 4M.

10b). Explain IPV4 addressing and IP datagram format

→ 10M

Ans. Each host, gateway and router has a unique Internet wide IP address assigned to it that comprises a netid and hostid part.

Each gateway/router port has a different netid associated with it.

One of 3 different address formats can be used. Each format is known by an address class.

Each of these classes is intended for use with a different size of network.

The class to which an address belongs can be determined from the position of the 1st zero bit in the 1st 4 bits.

The remaining bits specify the netid and hostid parts.

Class A addresses have 7 bits for netid and 24 bits for the hostid.

Class B addresses have 14 bits for netid and 16 bits for hostid.

Class C addresses have 21 bits for netid and 8 bits for hostid.

Class D addresses are reserved for multicasting.

— 4M

IPV4 datagram

0 – 4	4 – 8	8 – 16	16 – 32
version	IHL	Type of service	Total length.
Identification			D M · Fragment offset
Time to live		Protocol	Header checksum
Source IP address			
Destination IP address			
Options			
Payload (≤ 65535 bytes)			

— 2M.

All user info. is transfered in the payload part of what is known as a datagram.

- Version field contains the version of the IP used.
- Intermediate header length field specifies the actual length of the header.

The type of service field shows the priority of the application data.

The total length field defines the total length of the datagram

Identification field helps destination to relate each received packet fragment to the same original datagram.

- The next 3 bits are flags. D flag defines don't fragment and M flag defines more fragment.

The fragment offset is used by the same procedure to indicate the position of the 1st byte of the fragment.

— 2M

The value in the time to live field defines the maximum time for which a packet can be in transit across the internet.

The value in the protocol field is used to enable the destn IP to pass the payload within each received packet to the same protocol that sent the data.

- The header checksum applies just to the header part of datagram and is a safeguard against corrupted packets.

Source & destn addresses are IP addresses of Source & destn

Finally the options field is used to carry additional info. relating to 1) security 2) source routing 3) loose source routing 4) route recording 5) stream identification 6) time stamp.

— 2M

(34)