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Sub:	MMC					Sub Code:	
Date:	20/11/2024	Duration:	90 Minutes	Max Marks:	50	Sem / Sec:	

Answer any FIVE FULL Questions

1 A digitized video is to be compressed using the MPEG-1 standard. Assuming a frame sequence of : IBB PBB PBB PBB, and average compression ratio of 10:1 (I), 20:1 (P) and 50:1 (B), derive the average bit rate that is generated by the encoder for both NTSC and PAL digitization formats. [10]

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:

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

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

Hence bit rate generated at 30 fps = 1.040 Mbps

2. **PAL (720 × 576)**:

- Resolution = $720 \times 576 = 414,720$ pixels per frame
- Frame rate = 25 fps
- Bits per pixel = 8 bits

$$\text{Uncompressed bit rate} = 414,720 \times 25 \times 8 \text{ bits per second} = 82,944,000 \text{ bits per second} \approx 82.94 \text{ Mb}$$

2	Explain principle of linear predictive coding with a neat block schematic.	[10]

Linear Predictive Coding (LPC) is a widely used technique in speech processing, particularly in speech compression, synthesis, and analysis. LPC models the speech signal as a linear combination of its previous samples, with the aim of predicting the current sample based on a weighted sum of past samples. The method is based on the assumption that speech signals are quasi-stationary over short periods, meaning that the speech signal can be approximated as a linear function of previous samples over small time frames.

Principle of LPC

The basic principle behind LPC is to predict the value of a sample in a speech signal as a weighted sum of previous samples. The weights, or **prediction coefficients**, are calculated such that the difference between the predicted value and the actual value is minimized. This difference is called the **residual signal**.

- **LPC Model**: The speech signal is modeled by a linear equation:

$$x[n] = \sum_{k=1}^{P} a_k \cdot x[n-k] + e[n]$$

where:

  - $x[n]$ is the current speech sample.
  - $x[n-k]$ are the previous speech samples.
  - a_k are the LPC coefficients (prediction coefficients).
  - P is the order of the predictor (the number of past samples used for prediction).
  - $e[n]$ is the residual error (the difference between the predicted and actual value).

The goal of LPC is to find the optimal set of coefficients $a_1, a_2, \ldots, a_P$ that minimize the prediction error, $e[n]$, or residual signal.

Diagram

```
+-----------------------------+
|   Speech Signal (x[n])      |
+-----------------------------+
          |
          v
   +-------------+
   | Pre-emphasis|
   +-------------+
          |
          v
+----------------------+
```

```
| Frame Blocking       |
| (Divide into frames) |
+----------------------+
          |
          v
+---------------------+
| Autocorrelation     |
| Computation         |
+---------------------+
          |
          v
+---------------------+
| LPC Analysis (Find  |
| Prediction Coeff.)  |
+---------------------+
          |
          v
+---------------------+
| Residual Signal     |
| (x[n] - predicted)  |
+---------------------+
          |
          v
+---------------------+
| Quantization        |
+---------------------+
          |
          v
+---------------------+
| Encoding & Output   |
+---------------------+
```

Explanation of the Diagram:

1. **Speech Signal (x[n])**: This is the raw speech signal, which is the input to the LPC system.
2. **Pre-emphasis**: A high-pass filter is applied to enhance the higher frequency components of the speech signal.
3. **Frame Blocking**: The speech signal is divided into small frames, typically lasting 20-30 ms. This is important as speech signals can change rapidly, and short frames allow the system to adapt to the changes.
4. **Autocorrelation Computation**: Autocorrelation is calculated for each frame, which measures the correlation between a signal and a delayed version of itself. This helps in capturing the linear dependence between different samples of the speech signal.
5. **LPC Analysis**: Using the autocorrelation values, the LPC analysis computes the prediction coefficients (or LPC coefficients) that minimize the prediction error.
6. **Residual Signal**: The residual signal is the difference between the actual speech signal and the predicted signal. This is the error signal that represents the unpredicted part of the speech.
7. **Quantization**: Both the LPC coefficients and the residual signal are quantized to reduce the data rate for transmission or storage. Quantization reduces the precision of the coefficients, leading to some loss of quality but improving compression.
8. **Encoding & Output**: The quantized coefficients and residual signal are encoded for transmission or storage.

3 Explain MPEG-4 coding principles with the help of a neat diagram. [10]

MPEG-4 is a video and audio compression standard developed by the Moving Picture Experts Group (MPEG) to achieve efficient multimedia coding. It incorporates both video and audio compression techniques, and it is particularly used for streaming applications, multimedia communications, and digital television broadcasting. The coding principles of MPEG-4 revolve around several key concepts that help achieve high-quality video and audio while reducing the data rate for efficient transmission or storage.

Diagram:

```
      +-----------------------------+
      |     Video Input (Frames)    |
      +-----------------------------+
                  |
                  v
   +-------------------------------+
```

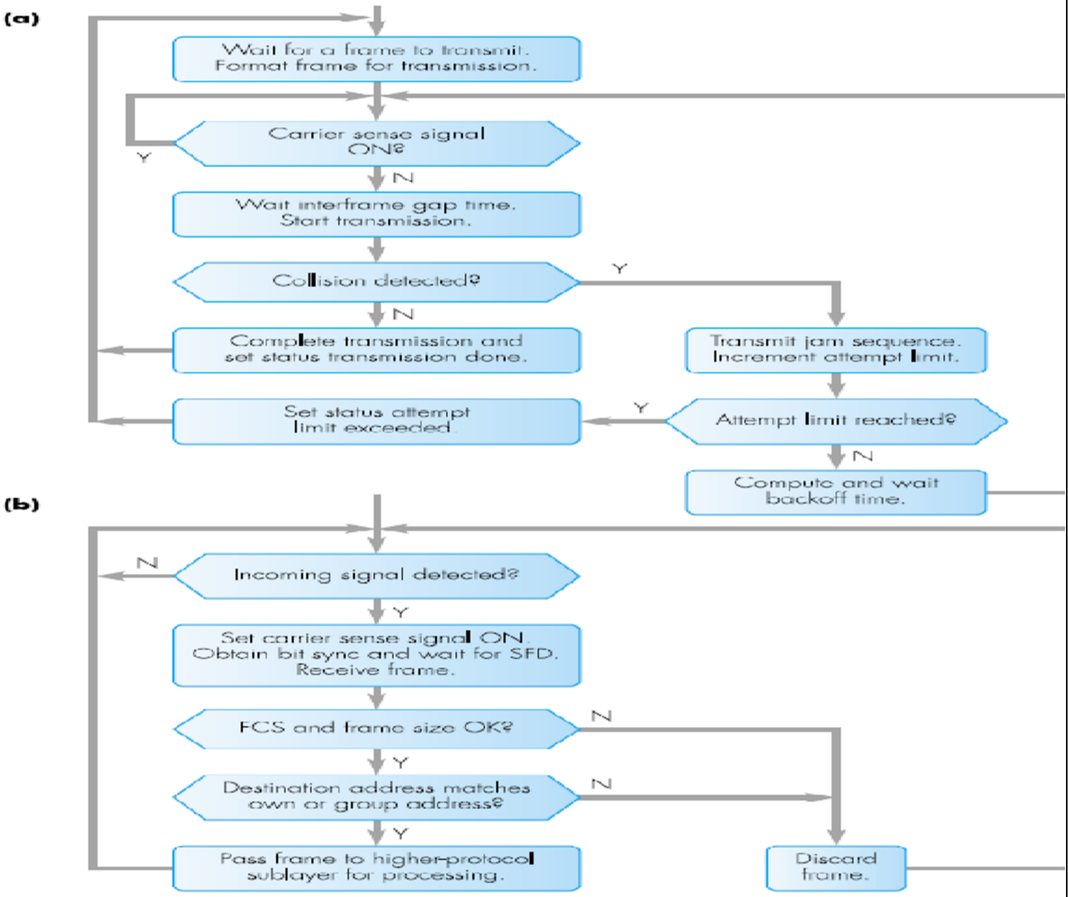
```
|    Motion Estimation (Find      |
|    Differences Between Frames) |
+-------------------------------+
                |
                v
+-------------------------------+
|    Motion Compensation (Apply |
|    Predicted Motion Vectors)  |
+-------------------------------+
                |
                v
+-------------------------------+
|      Transform Coding (DCT)    |
|   (Convert to Frequency Domain)|
+-------------------------------+
                |
                v
+-------------------------------+
|        Quantization (Reduce    |
|        Precision of Coefficients)|
+-------------------------------+
                |
                v
+-------------------------------+
|      Entropy Coding (Huffman/  |
|      Arithmetic Coding)        |
+-------------------------------+
                |
                v
+-------------------------------+
|    Compressed Video Output    |
|   (Encoded Bitstream)          |
+-------------------------------+
```

Explanation of the Diagram:

1. **Video Input (Frames)**: This is the raw video, typically a sequence of frames that are going to be compressed.
2. **Motion Estimation**: The encoder looks at neighboring frames to find areas where the image has moved, calculating motion vectors.
3. **Motion Compensation**: The encoder predicts the current frame by applying the motion vectors to previous frames, reducing redundant information.
4. **Transform Coding (DCT)**: The residual errors after motion compensation are transformed using techniques like Discrete Cosine Transform (DCT) to convert the pixel data into frequency components.
5. **Quantization**: The transformed coefficients are quantized, reducing their precision to further compress the data.
6. **Entropy Coding**: Finally, the quantized data is encoded using entropy coding (Huffman or Arithmetic) to compress the data efficiently.
7. **Compressed Video Output**: The final output is the compressed video bitstream, which is much smaller in size compared to the raw input video.

4	Explain the following terms : 1) Group of Pictures, 2) Prediction Span, 3) Motion Compensation, 4) Motion Estimation, 5) Temporal Masking. 1) Group of Pictures: A GOP is the distance between two I-frames measured in the number of frames. I-frames are sometimes referred to as “Keyframes” as they are the key that the other types of frames are structured around. It begins with the keyframe (blue) and the white frames contain the information used to create the appearance of motion when referencing the keyframe.	[10]

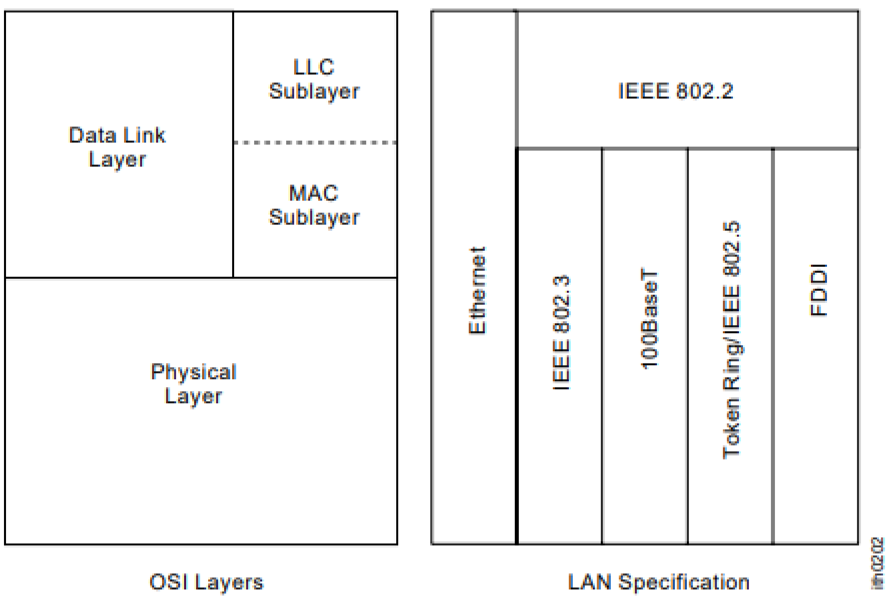
	2) Prediction Span: This typically refers to the length or range over which a prediction is made in video compression or signal processing. In video codecs, prediction is used to estimate pixel values based on neighboring pixels or frames. The prediction span determines how far back or forward the encoder looks to make these predictions, which can influence compression efficiency and quality. 3) Motion Compensation: This is a technique used in video compression to reduce temporal redundancy between consecutive video frames. Motion compensation involves predicting the movement of objects between frames and compensating for the differences by encoding only the motion vectors (the direction and magnitude of movement). This significantly reduces the amount of data needed to represent video. 4) Motion Estimation: This is the process of analyzing video frames to detect and estimate the motion of objects between consecutive frames. In video compression, motion estimation helps in creating the motion vectors that are used in motion compensation. The more accurate the motion estimation, the better the video quality and the higher the compression efficiency. 5) Temporal Masking: In the context of video and audio processing, temporal masking refers to the phenomenon where certain sounds or visual details become imperceptible to the human senses due to the presence of a stronger signal or noise at a particular point in time. In video compression, temporal masking can be used to discard less important information that is masked by stronger visual or motion signals, thereby improving compression efficiency without significantly affecting perceived quality.	
5	Explain the working of CSMA/CD protocol with suitable diagrams. 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 frame 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. CSMA/CD Working: Step 1: Check if the sender is ready to transmit data packets. Step 2: Check if the transmission link is idle. If idle then repeat the above process. The sender sends dummy data on the link. If it does not receive any collision signal, it sends the data. Otherwise, it refrains from sending data. Step 3: Transmit the data & check for collisions. During transmission, if a collision signal is received by the node, transmission is stopped. The station then transmits a jam signal onto the link and waits for random time intervals before it resends the frame. After some random time, it again attempts to transfer the data. Step 4: If no collision was detected in propagation, the sender completes its frame transmission and resets the counters.	[10]



6 Explain LAN protocol in detail with various layers and suitable figures. [10]

A LAN is a high-speed, fault-tolerant data network that covers a relatively small geographic area. It typically connects workstations, personal computers, printers, and other devices. LANs offer computer users many advantages, including shared access to devices and applications, file exchange between connected users, and communication between users via electronic mail and other applications.

Popular LAN protocols mapped to the OSI reference model.



LAN protocols are concerned principally with lower layers of the OSI model. This architecture was developed by the IEEE 802 LAN standards committee It is generally referred to as the IEEE 802 reference model.

	Working from the bottom up, the lowest layer of the IEEE 802 reference model corresponds to the physical layer of the OSI model and includes such functions as : Encoding / decoding of signals Preamble generation/removal Bit transmission / reception In addition, the physical layer of the 802 model includes a specification of the transmission medium and the topology.Above the physical layer are the functions associated with providing service to LAN users. These include the following: 1. On transmission, assemble data into a frame with address and error-detection fields. 2. On reception, disassemble frames and perform address recognition and error detection. 3. Govern access to the LAN transmission medium . 4. Provide an interface to higher layers and perform flow and error control . The set of functions in the last point are grouped into a logical link control (LLC) layer. The functions in the first three points are treated as a separate layer, called medium access control (MAC). The separation is done for the following reasons: 1. The logic required to manage access to a shared access medium is not found in traditional layer 2 data link layer . 2. For the same LLC, several MAC options may be provided	
7	a) Assuming signal propagation delay in the fiber of 5 microseconds per 1 km, derive the latency of the following FDDI ring configurations in both time and bits assuming a usable bit rate of 100 Mbps. 1) 2 km ring with 20 stations, 2) 20 km ring with 200 stations, 3) 100 km ring with 500 stations. b) Explain the various types of high speed LANs. Ans a) Ring latency, Tl = Signal propagation delay, T p + N × station latency, T s where N is the number of stations. (i) Tl = 2 × 5 + 20 × 1 = 30µs or 3000 bits (ii) Tl = 20 × 5 + 200 × 1 = 300µs or 30000 bits (iii) Tl = 100 × 5 + 500 × 1 = 1000µs or 100000 bits Note that the above values assume that the primary ring only is in use. If a fault occurred, the three signal propagation delay values would each be doubled. Also, for dual attach stations, the station latency would be doubled. Ans b) High Speed LANs are classified as •8.7.1:Fast Ethernet: •8.7.2:Switched Fast Ethernet •8.7.3:Gigabit Ethernet •Fast Ethernet: It was to use the same shared, half-duplex transmission mode as Ethernet but to obtain a*10 increase in operational bit rate over 10BaseT while at the same time retaining the same wiring systems , MAC method , and frame format. (10-Speed-10Mbps, T=twisted pair cable) •The major technological hurdle to overcome with Fast Ethernet was how to achieve a bit rate of 100Mbps over 100m of UTP cable. •Switched Fast Ethernet: In order to allow multiple access/transfers to be in progress concurrently, two developments have been made: –Switch hub architecture –Duplex working over the circuits that connect the stations to the hub. •Gigabit Ethernet: A transmission technology based on the Ethernet frame format and protocol used in local area networks (LANs), provides a data rate of 1 billion bits per second, or 1 gigabit (Gb).	[10]