MULTIMEDIA SYSTEMS AND APPLICATION REVISION

1. differentiate between lossy and lossless compression
   lossless compression retains all of the original data but may not achieve as high of a compression ratio as lossy compression, while lossy compression sacrifices some of the original data to achieve a smaller file size but may result in a loss of quality.

2. highlight various ways of display multimedia outputs
   multimedia outputs can be displayed in various ways depending on the type of content and the platform or device being used to display it. These include text-based displays, image displays, audio displays, video displays, and interactive displays.

3. discuss various evolving technologies for multimedia systems
   multimedia technology is constantly evolving, with new technologies emerging that allow for more immersive, efficient, and secure multimedia systems. These include VR/AR, HDR, cloud-based systems, AI, 5G networks, and blockchain technology.

4. describe the characteristics of rich text format
   RTF is a versatile document file format that supports text formatting, media integration, Unicode support, compact file size, and security features. It is widely used for creating and sharing documents across different platforms and applications.

5. outline two types of ISDN connection used in H323

   I. ***Basic Rate Interface (BRI):*** *BRI is a type of ISDN connection that provides two 64 Kbps data channels, known as B channels, and one 16 Kbps signaling channel, known as a D channel. BRI connections are commonly used in small to medium-sized video conferencing systems that require a moderate amount of bandwidth.*
   II. ***Primary Rate Interface (PRI):*** *PRI is a type of ISDN connection that provides 23 B channels and one D channel, with a total data rate of 1.544 Mbps. PRI connections are commonly used in large video conferencing systems that require a higher amount of bandwidth and support for multiple simultaneous calls.*

6. highlight four features of megaCo

   i. ***Call control signaling:*** *MegaCo provides a standardized way for controlling call signaling between media gateways and other endpoints in an IP telephony network. It supports a wide range of call types, including voice, video, and data calls.*
   ii. ***Scalability:*** *MegaCo is designed to be scalable, allowing it to handle large numbers of media gateways and endpoints. This makes it well-suited for use in large-scale IP telephony networks.*
   iii. ***Fault tolerance****: MegaCo includes mechanisms for detecting and handling faults in media gateways and other network components. It supports redundancy and failover mechanisms to ensure that calls can continue even in the event of equipment failures.*
   iv. ***Flexibility****: MegaCo is a highly flexible protocol that can be adapted to meet the specific requirements of different IP telephony networks. It supports a range of configuration options, including the ability to customize call flows and signaling messages, and can be used with a variety of transport protocols, including UDP and TCP/IP*.

7. discuss two types of SIP message

i. ***INVITE:*** *The INVITE message is used to initiate a new session. It includes details about the caller and the callee, as well as information about the type of session being requested. The callee can respond with a 2xx series response code to accept the call, or a 4xx or 5xx series response code to reject the call or indicate an error.*
ii. ***BYE:*** *The BYE message is used to terminate an existing session. It is sent by either the caller or the callee to signal that the session is ending. The message includes a reason for the session termination, such as "normal call clearing" or "user busy". The other party can respond with a 2xx series response code to acknowledge the session termination.*

8. explain the characteristics and goals of RSVP in multimedia systems

i. ***Reservation-based approach****: RSVP uses a reservation-based approach to establish network resources for multimedia sessions. This means that network resources are reserved in advance of a session, to ensure that the required bandwidth and network capacity are available.*
ii. ***Scalability****: RSVP is designed to be scalable and able to support large-scale multimedia networks. It uses a hierarchical signaling approach, where signaling messages are only propagated to the relevant nodes in the network.*
iii. ***Fine-grained QoS control****: RSVP allows fine-grained control over QoS parameters such as bandwidth, latency, and packet loss. This allows multimedia applications to specify their QoS requirements in detail, and ensures that these requirements are met during the session.*
iv. ***Flexibility:*** *RSVP is flexible and can be used in a variety of network topologies, including point-to-point, multicast, and multipoint-to-multipoint sessions. This makes it suitable for a wide range of multimedia applications, from video conferencing to real-time gaming.*

9. explain how black pressure is used in congestion control

i. ***Detection of congestion****: When a network experiences congestion, the routers in the network detect the congestion by monitoring the queue length and packet loss rate on each link. If the queue length exceeds a certain threshold, or if the packet loss rate increases significantly, the routers infer that congestion is occurring.*
ii. ***Setting the black pressure bit:*** *When congestion is detected, the routers set a "black pressure" bit in the packets that are being sent through the congested link. This bit indicates to the receiving nodes that the network is congested and that the packets are experiencing a longer delay than usual.*
iii. ***Backing off transmission****: The receiving nodes that receive packets with the black pressure bit set respond by reducing their transmission rate, in order to alleviate the congestion. This back-off mechanism ensures that the network remains stable and that the congestion does not worsen.*
iv. ***Gradual increase in transmission rate:*** *As the congestion subsides, the sending nodes gradually increase their transmission rate again, until the network reaches a stable state.*

10. explain the use of watermarking in multimedia systems

i. ***Copyright protection****: Watermarking can be used to protect copyrighted multimedia content, such as music or videos, from unauthorized distribution or piracy. A digital watermark can be embedded in the content, which can be used to trace the source of the content and identify copyright violations.*
ii. ***Authentication****: Watermarking can be used to authenticate the source and integrity of multimedia files. For example, a digital watermark can be embedded in a digital*

photograph to verify that it was taken by a specific camera and has not been tampered with.

iii. ***Traceability:*** *Watermarking can be used to trace the distribution and usage of multimedia files. For example, a unique watermark can be embedded in a video, which can be used to track how the video is being shared and viewed online.*

iv. ***Content management****: Watermarking can be used as a tool for content management in multimedia systems. For example, a watermark can be used to indicate the version or the owner of a particular file, which can be useful for managing large collections of multimedia files.*

Section B.

11. describe six binary image compression schemes used in multimedia applications

i. ***Run-length encoding (RLE):*** *This is a simple compression scheme that works by replacing runs of identical pixels with a code that indicates the length of the run. This method is particularly effective for compressing binary images with long runs of black or white pixels.*

ii. ***Huffman coding:*** *This is a lossless compression scheme that uses variable-length codes to represent the most frequently occurring pixel values with shorter codes and the less frequent values with longer codes. This method is effective for compressing binary images with a large number of repeating patterns.*

iii. ***Arithmetic coding****: This is another lossless compression scheme that uses fractional values to encode the probability of each pixel value occurring. This method can achieve higher compression ratios than Huffman coding, but it is also more computationally intensive.*

iv. ***JBIG (Joint Bi-level Image Experts Group) compression****: This is a lossy compression scheme that uses a combination of RLE, Huffman coding, and arithmetic coding to compress binary images. It is particularly effective for compressing scanned documents, as it can achieve high compression ratios while maintaining high quality.*

v. ***CCITT Group 3 and Group 4****: These are lossless compression schemes used for fax transmission, which use a combination of RLE and Huffman coding to compress binary images. Group 3 is a simpler and faster algorithm that uses fixed-length codes, while Group 4 is more complex and uses variable-length codes.*

vi. ***Modified Huffman coding****: This is a lossy compression scheme that modifies the standard Huffman coding algorithm to allow for a small number of errors in the compressed image. This method is particularly effective for compressing binary images with a large number of repeating patterns, and can achieve high compression ratios with acceptable quality.*

12. discuss four key technologies used in multimedia input and output technologies

i. ***Voice recognition technology****: Voice recognition technology allows users to interact with multimedia systems using spoken commands, rather than relying on traditional input devices such as keyboards or mice. This technology is particularly useful in hands-free environments or for users with disabilities, and is widely used in smart home assistants and mobile devices.*

ii. ***Gesture recognition technology****: Gesture recognition technology allows users to interact with multimedia systems using hand movements, such as waving or pointing. This technology is particularly useful for virtual reality and augmented reality applications, where users can manipulate digital objects in a more natural and intuitive way.*

iii. ***Haptic feedback technology****: Haptic feedback technology provides physical feedback to users, such as vibration or force, in response to their actions in a multimedia system. This technology* *can enhance the sense of immersion and realism in virtual reality applications, and can also be used in gaming and mobile devices.*

iv. ***3D display technology**: 3D display technology allows users to view multimedia content in three dimensions, providing a more immersive and engaging experience. This technology can be used in virtual reality and augmented reality applications, as well as in 3D movies and gaming.*

13. discuss various retrieval and storage devices in multimedia system

i. ***Hard disk drives (HDDs):** HDDs are the most common type of storage device used in multimedia systems, providing large amounts of storage capacity for multimedia content such as images, videos, and audio files. They are relatively cheap and have fast access times, making them ideal for storing and retrieving multimedia content.*
ii. ***Solid-state drives (SSDs):** SSDs are similar to HDDs, but use flash memory instead of spinning disks to store data. They have faster access times and are more reliable than HDDs, making them well-suited for storing multimedia content that needs to be accessed quickly.*
iii. ***Optical storage devices**: Optical storage devices such as CDs, DVDs, and Blu-ray discs are still used for storing and distributing multimedia content such as movies, music, and software. They provide relatively large storage capacity and are portable and easy to distribute, but have slower access times than HDDs or SSDs.*
iv. ***USB flash drives:** USB flash drives are portable storage devices that are commonly used for transferring multimedia content between devices. They have relatively small storage capacity compared to HDDs or SSDs, but are convenient and easy to use.*
v. ***Cloud storage:** Cloud storage services such as Dropbox, Google Drive, and iCloud provide a convenient and flexible way to store and access multimedia content over the internet. Cloud storage allows users to access their multimedia content from anywhere and from any device with an internet connection.*
vi. ***Network-attached storage (NAS):** NAS devices provide a centralized storage solution for multimedia content that can be accessed by multiple users or devices on a network. They can be configured with multiple hard drives for increased storage capacity and redundancy, and provide advanced features such as backup and remote access.*
vii. ***Magnetic tape storage**: Magnetic tape storage is a high-capacity storage medium used primarily for long-term archival storage of multimedia content. It provides large storage capacity at a relatively low cost, but has slower access times than other storage devices.*

14. describe TCP implementation policy option as specified in TCP protocol entities

i. ***Maximum Segment Size (MSS):** MSS is a policy option that specifies the maximum size of the TCP data segment that can be sent over the network. This option is used to optimize TCP performance on networks with smaller Maximum Transmission Unit (MTU) sizes.*
ii. ***Selective Acknowledgement (SACK**): SACK is a policy option that allows TCP to acknowledge multiple packets at once, rather than waiting for each packet to be individually acknowledged. This option is used to reduce the overhead of packet acknowledgements and improve TCP performance on high-speed networks.*
iii. ***Time Stamps**: Time Stamps is a policy option that adds a timestamp to each TCP segment, allowing the receiver to calculate the Round Trip Time (RTT) of the connection*

more accurately. This option is used to improve TCP congestion control and reduce unnecessary retransmissions.

iv. ***Window Scaling****: Window Scaling is a policy option that allows the TCP window size to be increased beyond the default maximum of 64KB. This option is used to optimize TCP performance on high-speed networks where larger window sizes are necessary.*

v. ***Nagle Algorithm:*** *The Nagle Algorithm is a policy option that delays sending small TCP segments until a larger segment can be sent, reducing the number of small segments sent over the network. This option is used to improve TCP performance on networks with high latency.*

15. explain how congestion management is being addressed in random early detection

Random Early Detection (RED) is a congestion management mechanism used in packet-switched networks to reduce network congestion and improve overall network performance. RED works by monitoring the average queue length of a network router and selectively dropping packets before the queue becomes too congested. By dropping packets before the queue becomes too full, RED helps to prevent network congestion and packet loss.

In RED, packets are dropped randomly based on the average queue length of the router. When the average queue length exceeds a certain threshold, RED begins dropping packets randomly. The probability of dropping packets increases as the queue length increases, so that as the queue length grows, more packets are dropped.

The main goal of RED is to provide an early warning of network congestion, so that congestion can be avoided before it becomes too severe. By selectively dropping packets before the queue becomes too congested, RED helps to prevent network congestion and packet loss. This can improve overall network performance, reduce delays, and improve the reliability of the network.

One of the key benefits of RED is that it can be used to manage congestion without requiring complex network configuration or management. RED can be implemented on network routers and other network devices using standard network protocols, making it a relatively easy and cost-effective solution for managing network congestion.

16. explain how multicast IP is used in multimedia systems and applications

Multicast IP is a networking protocol that enables the transmission of data from a single sender to multiple receivers over a network. In multimedia systems and applications, multicast IP is commonly used to distribute multimedia content such as audio and video streams to multiple recipients simultaneously.

In a multicast IP network, a sender transmits a single copy of the multimedia data to a multicast IP address, which is then delivered to multiple receivers who have subscribed to that multicast group. This allows multimedia content to be transmitted efficiently to a large number of recipients without requiring multiple unicast transmissions.

Multicast IP is particularly useful in multimedia applications where the same content is being distributed to a large number of recipients at the same time, such as live video streaming or online gaming. Multicast IP can help to reduce network congestion, improve scalability, and reduce the cost of content distribution.

One of the key benefits of multicast IP in multimedia systems is its ability to deliver content in real-time. This is essential for applications such as live video streaming or real-time collaboration, where delays or interruptions can have a significant impact on the user experience.

However, there are some challenges associated with using multicast IP in multimedia systems. For example, multicast IP networks can be more difficult to configure and manage than traditional unicast networks, and multicast routing protocols can be more complex. Additionally, multicast IP may not be supported by all network devices or by all network service providers, which can limit its availability in some environments.

17. discuss any three-traffic condition function differential service

i. ***Classification:*** *Traffic classification is the process of identifying and categorizing different types of network traffic based on their source, destination, protocol, or other attributes. In DiffServ, classification is used to differentiate between different types of traffic and apply different QoS policies based on the importance of each type of traffic. For example, time-sensitive traffic such as video or voice may be given higher priority over less time-sensitive traffic such as email or file transfers.*
ii. ***Marking:*** *Traffic marking is the process of assigning a priority or class of service (CoS) value to network traffic. This value is used by network devices to determine how to handle the traffic and whether it should be given priority over other traffic. In DiffServ, marking is used to differentiate between different types of traffic and apply QoS policies based on their CoS values. For example, high-priority traffic may be marked with a higher CoS value than low-priority traffic, so that it is given higher priority in the network.*
iii. ***Policing****: Traffic policing is the process of enforcing network policies and ensuring that network traffic adheres to predefined QoS requirements. In DiffServ, policing is used to limit the amount of traffic that can be sent by a particular source or to enforce rate limits for certain types of traffic. For example, a network may enforce a rate limit for file transfers to prevent them from consuming too much bandwidth and degrading the performance of other types of traffic.*