CHAPTER 29 (corrisponde al cap. 28 italiano)

Multimedia

Solutions to Review Questions and Exercises

Review Questions

1. In ***streaming stored audio/video***, a client first downloads a compressed file and then listens to or watches it. ***In streaming live audio/video***, a client listens to or watches a file while it is being downloaded.
2. In ***frequency masking***, a loud sound partially or totally masks a softer sound. In ***temporal masking***, a loud sound blocks other sounds for a period of time.
3. A ***metafile*** contains information about a corresponding audio/video file.
4. ***RTSP*** is a control protocol that adds some functionalities to the streaming process. It is an out-of-band controlling protocol that functions like the FTP control connection.
5. ***Jitter*** manifests itself as a gap between what is heard or seen.
6. ***SIP*** is an application layer protocol that establishes, manages, and terminates a multimedia session.
7. ***JPEG*** is used to compress images. ***MPEG*** is used to compress video.
8. ***Blocking*** decreases the number of calculations.
9. The ***DCT*** reveals the number of redundancies of a block.
10. In ***spatial compression***, JPEG compresses each frame. In ***temporal compression***, redundant frames are removed.

Exercises

11.
    a. **9** packets played; **11** packets left
    b. **12** packets played; **8** packets left
    c. **17** packets played; **3** packets left
    d. **22** packets played; **8** packets left
12. ***TCP*** is not suitable for real-time traffic because it has no provision for timestamping, it does not support multicasting, and, most importantly, it retransmits lost or

corrupted packets. ***RTP*** is a protocol designed to handle real-time traffic. RTP handles timestamping, sequencing, and mixing. There is no retransmission when RTP is used with UDP.

13. We can say that ***UDP*** plus ***RTP*** is more suitable than ***TCP*** for multimedia communication. The combination uses the appropriate features of UDP, such as timestamp, multicasting, and lack of retransmission, and appropriate features of ***RTP*** such as error control.
14. ***RTCP*** is a control protocol that handles messages that control the flow and quality of data. It also allows recipient feedback. TCP allows for these types of messages, so it doesn't need RTCP.
15. The ***web server*** and ***media server*** can be two distinct machines since it is the metafile-data file combination that is important.
16. ***SIP*** can be modified to be used for interactive video such as teleconferencing.
17. Both ***SIP*** and ***H.323*** use the Internet as a telephone network. The main difference is that H.323 uses a gateway to transform a telephone network message to an Internet message. See Table 29.1.

Table 29.1 *Solution to Exercise 17*

Issues	***SIP***	***H.323***
Transport layer	UDP or TCP	UDP for data, TCP for control
Address format	IP address, e-mail address, or phone number	IP address
Establishment	3-way handshake	H.225, Q.931, H.245
Data exchange	UDP, TCP	RTP, RTCP, UDP, TCP
Termination	BYE message	Q.931

18. We can mention some of the problems involved in full implementation of ***voice over IP***:
    a. Your computer has to be on all the time as well as connected to the Internet.
    b. If the Internet connection is down, your phone service is also down.
    c. Voice quality can be a problem due to echoes or delays.
    d. There could be potential call degradation if the computer is also doing heavy processing.
19. ***H.323*** can also be used for video, but it requires the use of videophones. Currently most people don't have videophones.