1) What is band-limited signal?

A band-limited signal x(t) is one whose Fourier Transform is non-zero on only a finite interval of the frequency axis. If there exists a positive number B such that **X(f) is nonzero only in -B≤f≤B**, B is called the signal bandwidth

The discrete Fourier transform (DFT) is the counterpart of the continuous Fourier transform. It is defined for discrete-time signals. The transformed signal is also a discrete signal but in the frequency domain and it represents samples of the signal spectrum.

Let $x(nT_s)$ be a sequence of signal samples taken at time moments nT_s $(0 \le n \le N-1)$, then the DFT of $x(nT_s)$ is

$$y(k\Omega) = \sum_{n=0}^{N-1} x(nT_s)e^{-jkn\Omega T_s}, 0 \le k \le N-1. \tag{5.14}$$

2) What is the Nyquist sampling theorem? Provide a definition for the Nyquist frequency.

Nyquist theorem: If x(t) is a **band-limited signal** containing no frequencies higher than B. A **sufficient** sampling rate is **2B** or anything **larger**. 2B is called the Nyquist rate.

Nyquist frequency: **Maximum frequency** that will **not alias** given **a sampling rate**

3) What is an anti-aliasing filter?

Anti-Aliasing Filter (AAF) **removes** the selected **higher frequency components** from the source signal.

4) Define the meaning of the term "quantization interval" and how this influences the accuracy of the sampling process of an analogue signal.

If Vmax is the maximum positive and negative signal amplitude and n is the number of binary bits used then the quantization interval, q, is defined as

q = Vmax/2^n

How this influences ??? (HELP) n = number of binary bits used => **higher n -> higher accuracy ⇔ small q**. Noise (quantization error) is around +- q/2 => small q = lower noise. (same answer as question 5??)

5) Why does the quantization error gets worse with fewer bits to present an analogue signal?

Fewer bits -> the difference between the **actual signal amplitude** and the corresponding **nominal amplitude** is larger => error gets worse

6) Explain the run length coding with an example.

Runs of data (that is, sequences in which the same data value occurs in many **consecutive** data elements) are stored as a single data value and **count**, rather than as the original run

Ex: aaaaabbbbccc => 5a4b3c

7) Explain the Huffman coding with an example.

https://www.youtube.com/watch?v=dM6us854Jk0
Để lại link youtube điĐể hiểu còn làm bài tập ông

8) Explain the LZW coding with an example.
ABCBCABCABCD
https://www.youtube.com/watch?v=rZ-JRCPv_O8

9) What are DC and AC coefficients in image Compression?

The **coefficients** is the number that represents the **contribution** of each of **64 base cosine waves** blocks to the whole image.

Direct current (**DC**) coefficient: the **top left** block (F(0, 0)). Alternating current (**AC**): all the **other** block (i.e., F(u,v) (u,v≠0))

10)What characteristics of eye are exploited in the quantization of the image?

Eye is more **sensitive** to color pattern in **low spatial frequencies**.
In the human eyes, if the 2 colors are too near, the eyes can not distinguish.

The human eye is .... not so good at distinguishing the exact strength of a high frequency (rapidly varying) brightness variation. This fact allows one to reduce the amount of information required by ignoring the high frequency components.

11) Explain how the block preparation is performed in an image compression phenomenon?
(No idea)

12) Explain the process vectoring using a zigzag scan diagram.
By using zigzag scan, we get a huge list of **zero in a row**, which is easily compressed using run length coding or Huffman code

13) Why is DCT used in transform encoding?

DCT transforms a signal of image from the spatial domain to the frequency domain. DCT can **achieve compression** when **neglect** the **lower right** value that represent **higher frequencies** with little visible distortion.
----

14) In an image compression explain where information loss can occur?

If we have got an image where **lots of high frequency** information pixel changes are happening, that might get **significantly compressed**. The picture might get worse and information loss can occur

15) What is differential coding?

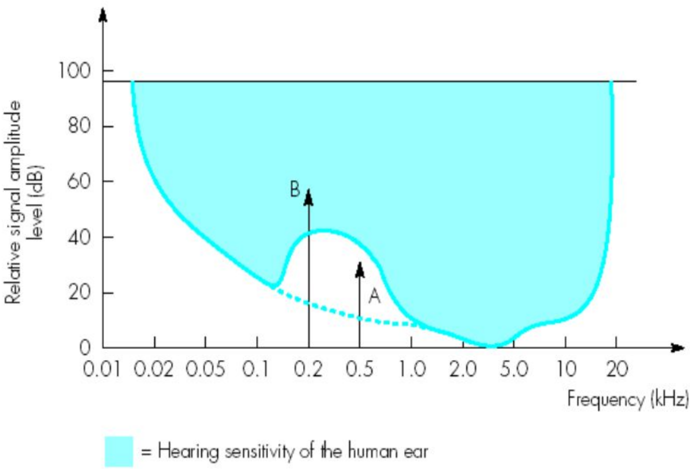
Encode the **differences** between the signal **itself** and its **prediction**. Also known as predictive coding.

16) Explain the term “critical bandwidth” and identify how this also varies with frequency.

The critical bandwidth is the width of each basic sensitivity curve at a particular signal level for that frequency.

17) Explain the meaning of the term “frequency masking”. Illustrate on your graph the masking effect of a loud signal on neighbouring signals.

frequency masking: When multiple signals are present, a **strong signal** may **reduce** the level of **sensitivity** of the ear to **other signals** which are **near** to it in frequency



Signal B is larger than signal A. This causes the basic sensitivity curve to be distorted.

Signal A will no longer be heard as it is within the distortion band.

18) Explain the meaning of the term “temporal masking”. What are the implications of exploiting this effect?

After the ear hears a **loud sound**, it takes a further **short time** before it can hear a **quieter sound**. This is known as the temporal masking
Implication: During this time, signals whose **amplitudes** are **less** than the decay envelope will **not be heard** and hence need **not** be **transmitted**

19) Explain the operation of a basic DPCM signal encoder and decoder. Include in your explanation the source of errors that can arise.
Encoder

- Previous digitized sample is held in the register (R).
- The DPCM signal is computed by subtracting the current content (R) from the new output by the ADC.

- The register value is then updated before transmission.

Decoder
- Decoder simply adds the previous register contents (PCM)
with the DPCM.
- Since ADC will have noise there will be cumulative errors in
the value of the register signal

20)Explain how a basic ADPCM scheme obtains improved performance over a DPCM scheme.
In ADPCM, **saving of bandwidth** is possible by **varying** the number of **bits** used for the difference signal **depending** on its **amplitude**.

21) Explain how better sound quality-for the same bit rate-can be obtained using a sub-band coding ADPCM. Give examples of the bit rates used for the lower and higher subbands and state an application of this type of codec.
How better sound quality ???
bit rates used for the lower and higher subbands: 48 kbps and 16 kbps
In some applications as voice coding, the subband that includes the voice is coded with more bits than the others. It is a way to reduce the file size

22) Explain the meaning of I, P and B frames of and the reasons for their use.
I-frames are encoded without reference to any other frames. Each frame is treated as a **separate picture** and the Y, Cb and Cr matrices are encoded separately using JPEG.
The encoding of the P-frame 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**
B-frames are predictions of how object have moved across the scene using the past and the future frame. It contains the **information** of **how a specific object move** across a small portion of the screen
We generate the I frame then the P frame, then based on the objects and how they moved, we construct the B frame.

23) Explain the terms of motion compensation and motion estimation in relation to the P-frames in video compression.

The **references** between the **different types of frames** are realized by a process called motion estimation. The **correlation** between two frames in terms of **motion** is represented by a **motion vector**.
Since the estimation is **not exact**, **additional** information must also be sent to indicate any **small differences** between the **predicted** and **actual** positions of the moving segments involved. This is known as the motion

compensation.

24)Explain group of pictures in relation to video compression? What happens in a fast moving scene.

The number of frames/pictures **between successive I-frames** is known as a group of pictures (GOP)
For fast moving video motion estimation will not work effectively. Hence **B-frames** (Bi-directional) are used. Their contents are **predicted** using the **past** and the **future** frames

25)What is a moving JPEG?
A **compression technique** for video images which applies the **JPEG compression** algorithm to each **still image** making up the video clip that is to be compressed.

26)Why do we use video object planes in MPEG-4? (slide 170)

Each audio & video object has a **separate object descriptor** associated with it, which allows the object to be **manipulated** by the viewer prior to being decoded & played out.

27)What is FDDI? Describe the pros and cons of FDDI?

FDDI uses a **ring topology** of multimode or single mode **optical fiber transmission** links operating at 100 Mbps to span up to 200 km and permits up to 500 stations.
Cons:

– Higher **Cost** and Maintenance
– More **complex** to implement

Pros:

– Higher **Bandwidth**
– **Large distance** between FDDI nodes
-- Improved **Signal-to-Noise** ratio
– Difficult to **tap signal** from a **fiber cable**
– **Bit error rate** better than in copper and microwave system

28)Discuss the network requirements for multimedia
Communication.
xxxxxxxxxxxxxxxxxx
• **High speed and changing bandwidth**
  ▫ Multimedia applications, particularly those using video and images demand **large bandwidth**. However, bandwidth for the foreseeable future will be limited. The limitation arise from the cost of installing optical fiber transmission, terminal equipment complexity and speed, etc.

• **Quality of Services**
  ▫ The **availability** of multimedia sources places demands on

the service that a network must provide. The most importance of these are the bit error rate, the packet or cell loss, delay and delay variation.

• **Synchronization** of different information types

▫ Multimedia synchronization refers to temporal relationships between the media objects. A common example of temporal relationship is movie or television, where both audio and video objects are involved.

• **Reliable security** features and **firewalls**

▫ Security features include digital watermarking, data hiding, multimedia content protection, biometrics, multimedia human-computer interface.

▫ Firewall refers to a system, which controls the incoming and outgoing network traffic based on an applied rule set.

29) Explain why the ATM packet size is 53 Byte.

The design of ATM aimed for a low-jitter network interface. However, "cells" were introduced into the design to provide short queuing delays while continuing to support datagram traffic. ATM broke up all packets, data, and voice streams into 48-byte chunks, adding a 5-byte routing header to each one so that they could be reassembled later. The choice of 48 bytes was political rather than technical. When the CCITT (now ITU-T) was standardizing ATM, parties from the United States wanted a 64-byte payload because this was felt to be a good compromise in larger payloads optimized for data transmission and shorter payloads optimized for real-time applications like voice; parties from Europe wanted 32-byte payloads because the small size (and therefore short transmission times) simplify voice applications with respect to echo cancellation. Most of the European parties eventually came around to the arguments made by the Americans, but France and a few others held out for a shorter cell length. With 32 bytes, France would have been able to implement an ATM-based voice network with calls from one end of France to the other requiring no echo cancellation. 48 bytes (plus 5 header bytes = 53) was chosen as a compromise between the two sides. 5-byte headers were chosen because it was thought that 10% of the payload was the maximum price to pay for routing information. ATM multiplexed these 53-byte cells instead of packets which reduced worst-case cell contention jitter by a factor of almost 30, reducing the need for echo cancellers.

Shorter (in slide): The cell size is determined as a **trade-off** between **packetizing delay** and **cell overhead**. ATM packet size is 53 Byte because we can get the acceptable packetizing delay and cell overhead. A 53 Bytes cell contain 48 voice samples, which is only 48 x 125 μs = 6 ms of voice. The loss of a cell would be almost unnoticed.

30)What are MMDS and LMDS? How is MMDS different from LMDS.

- Multichannel Multipoint Distribution Service (MMDS), formerly known as Broadband Radio Service (BRS) and also known as Wireless Cable, is a wireless telecommunications technology, used for general-purpose broadband networking or, more commonly, as an alternative method of cable television programming reception.
- Local Multipoint Distribution Service (LMDS) is a broadband wireless access technology originally designed for digital television transmission (DTV). It was conceived as a fixed wireless, point-to-multipoint technology for utilization in the last mile.

Specifications	LMDS	MMDS
Full Form	Local Multipoint Distribution Service	Multichannel Multipoint Distribution Service
Architecture	The LMDS architecture consists of NOC (Network Operation Center), BS, CPE and Fiber backbone. It has cellular like architecture.	The MMDS architecture consists of tall antenna tower, backbone internet connectivity using router and network management system. It has microwave link like architecture.
Frequency of operation	28 GHz, 36 GHz	2.5 GHz, 3.5 GHz
Network Topology	P2MP (Point to Multi-point)	P2P (Point to Point)
Distance coverage	Good more smaller distances. (2 to 8 Km)	Covers larger distance. (50 to 100 Km)
Number of cells	more	very few
Data rate	1 to 10 Mbps	upto 2 Mbps
cost	CPE cost and deployment cost is medium to high.	CPE cost and deployment cost is low compare to LMDS.

31) Explain the Round Robin packet scheduling mechanism.

Buffer is organized in separate queues (each implemented FIFO) for each flow and a single packet is selected at time from queues with a circular mode.

32) Explain the term “Tail Drop” in network congestion. Why does Tail Drop lead to TCP global synchronization.

Tail Drop **drops arriving packets** when **buffers** in queue are **full**. It
may lead to network meltdown due to TCP global synchronization
Because, instead of discarding **many <u>segments</u> from one connection**, the router would tend to discard **one segment from each connection**.

33) What is IntServ? What is the main drawback of IntServ?

IntServ or integrated services is an **architecture** that **specifies** the elements to **guarantee** quality of service (**QoS**) on networks. Each applications that requires a service guarantee has to make a reservation by using Resource Reservation Protocol (RSVP) signaling

The main drawback of IntServ is its **lack of scalability**
(Routers have to classify, police and queue each flow)

34) Explain the terms “TSPEC” and “RSPEC” in Integrated Services.

TSPECs typically just specify the token rate and the bucket depth. For example, a video with a refresh rate of 75 frames per second, with each frame taking 10 packets, might specify a token rate of 750 Hz, and a bucket depth of only 10.
RSPECs specify what requirements there are for the flow: it can be normal internet 'best effort', in which case no reservation is needed.

Short: **TSPEC** describes the **flow traffic characteristics**. **RSPEC** describes the **service request** (request for controlled traffic and/or delay bound).

35) Describe the Token Bucket algorithm. What are the advantages of
Token Bucket over Leaky Bucket.

The token bucket algorithm can be conceptually understood as follows:

- A token is **added** to the bucket every 1/r seconds (r is token rate).
- The bucket can **hold** at the most b tokens. If a token **arrives** when the bucket is **full**, it is **discarded**.
- When a **packet** (network layer PDU) of **n bytes arrives**, **n tokens** are **removed** from the bucket, and the **packet** is **sent** to the network.
- If **fewer than n** tokens are **available**, **no tokens** are **removed** from the bucket, and the **packet** is considered to be **non-conformant**.

Main advantage of Token Bucket over Leaky Bucket:

- If bucket is **full** in Token Bucket, **token** are **discard** not packets. While in **Leaky Bucke**t, **packets** are **discarded**.
- **Token Bucket** can send **large bursts** at **faster rate** while **Leaky Bucket** always sends packets at **constant rate**.

36) Describe the RSVP mechanism? Why is RSVP receiver-oriented?

Sender sends PATH message to let the routers know on which links they should forward the reservation (RESV) message. PATH message contains TSPEC and specifies source traffic characteristics
Receiver requests for resources using RESV message. There are three reservation styles, which can be Fixed-filter, Wildcard-filter and Shared-explicit.
It is receiver-oriented because the receiver of a data flow initiates and maintains the resource reservation for that data flows.

37) Explain the terms “Stop-and-Wait ARQ”, “Go-back-N ARQ” and
“Selective ARQ”
A **stop-and-wait ARQ** sender sends one frame at a time. After sending each frame, the sender doesn't send any further frames until it receives an acknowledgement (ACK) signal. After receiving a valid frame, the receiver sends an ACK. If the ACK does not reach the sender before a certain time, known as the timeout, the sender sends the same frame again. The timeout countdown is reset after each frame transmission. The above behavior is a basic example of Stop-and-Wait.

Go-Back-N ARQ: the sending process continues to send a number of frames specified by a *window size* even without receiving an acknowledgement (ACK) packet from the receiver. It can transmit N frames to the peer before requiring an ACK. The receiver process keeps track of the sequence number of the next frame it expects to receive, and sends that number with every ACK it sends. The receiver will discard any frame that does not have the exact sequence number it expects (either a duplicate frame it already acknowledged, or an out-of-order frame it expects to receive later) and will resend an ACK for the last correct in-order frame. Once the sender has sent all of the frames in its *window*, it will detect that all of the frames since the first lost frame are *outstanding*, and will go back to the sequence number of the last ACK it received from the receiver process and fill its window starting with that frame and continue the process over again.

38) Explain the terms “Expedited Forwarding PHB” and “Assured

Forwarding PHB” in computer networking.

- Expedited Forwarding (EF) has the characteristics of **low delay**, **low loss** and **low jitter**. **EF traffic** is given **strict priority queuing above all** other traffic classes. Typical networks will limit EF traffic to no more than 30% of the capacity of a link.

- Assured Forwarding (AF) **defines four separate classes**. When congestion occurs, the traffic in the **higher class** is given **higher priority** (WFQ), and the **packets** with the **higher drop precedence** are **discarded** (RED)

39) Discuss the issues of multimedia synchronization.

- Content Relations

It defines a dependency of media objects on some data. An example of a content relation is two graphics that are based on the same data but show different interpretations of the data.

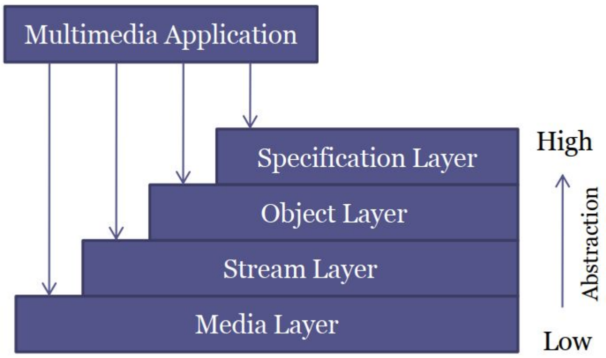
- Spatial Relations

It defines the space used for the presentation of a media object on an output device at a certain point of time in a multimedia presentation.

- Temporal Relations

It defines the temporal dependencies between media objects. They are of interest whenever time-dependent media objects exist.

40) Discuss the reference model for multimedia synchronization.



The model of Gerold Blakowski and Ralf Steinmetz, "A Media Synchronization Survey: Reference Model, Specification, and Case Studies, " IEEE Journal on Selected Areas in Communications, vol. 14, no. 1, Jan. 1996.

- Each layer implements synchronization mechanisms, which are provided by an appropriate interface. Each interface defines services, offering the user a means to define his requirements. Each interface can be used by an application or by the next higher layer.

41) Why is synchronization in a distributed environment more complex than in a local environment?

Synchronization in a distributed environment is more complex than in a local environment. This is caused by the **distributed storage** of **synchronization** information and the **distributed locations** of **source** and the **sink** (receiver). Even different media objects involved in the presentation may be located at different places.

42) How is the synchronization specification delivered between the source and the sink?

Transport of the Sync Specification

• **Delivery** of the **complete synchronization** information **before** the **start** of the presentation.

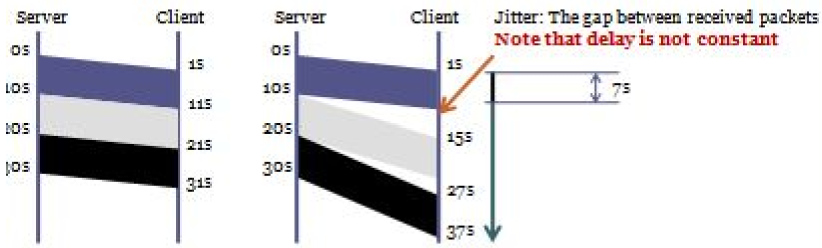
• Use of an **additional channel**.
 ▫ +: No addition delay
 ▫ -: Errors caused by delay or loss of synchronization units
 ▫ -: Difficult to handle multiple source nodes

• **Multiplexed data streams**
 ▫ +: No additional channel and delay
 ▫ +: MPEG bit stream combines audio, video and sync info.
 ▫ -: Difficult to select an appropriate QoS
 ▫ -: Difficult to handle multiple source nod

43) What is multi-step synchronization?

+) Synchronization during **object acquisition**, e.g., during digitizing video frames.
+) Synchronization of **retrieval**, e.g., synchronized access to frames of a stored video.
+) Synchronization during **delivery** of the Local Data Units to the network.
+) Synchronization during **transport**, e.g., by isochronous protocols.
+) Synchronization at the **sink**, i.e., synchronized delivery to the output devices.
+) Synchronization within the **output device**.

44) Explain the term “jitter”. Explain how the use of timestamp may overcome the jitter problem.
(page 237)



Timestamp is used in receiver’s buffer to **reorder** packets.
(**personal understanding) Set an initial delay so that the information can be shown to user continuously. In the example, set the initial delay to 7s instead of only 1s.

45) Explain why the real –time data can not be TCP?

• TCP forces the sink application to wait for retransmission(s) in the case of packet loss, causing large delays.
• TCP cannot support multicast, which is a basic requirement of video conferencing applications.

• TCP congestion control mechanisms decreases the congestion window when packet losses are detected. Audio and video on the other hand have bitrates that cannot be suddenly decreased.
• TCP headers are larger than a UDP header.
• TCP does not contain the timestamp and encoding parameters, needed by the receiver.
• TCP does not allow packet loss. In A/V, a loss of 1-20% is tolerable. The loss can be compensated by FEC.

46) What is RTP? What are the main functions of RTP? (slide 242)

RTP is a **network protocol** for **delivering audio and video** over **IP** network. RTP is used in conjunction with the Real-Time Control Protocol (RTCP). While RTP carries the media streams, RTCP is used to monitor transmission statistics and QoS and aids synchronization of multiple streams.

RTP does not ensure real-time delivery, but it provide means for:

- **Jitter elimination/reduction** by using **playback buffer**.
- **Synchronization** of several audio and video streams.
- **Multiplexing** of audio and video streams.
- **Translation** of audio and video streams.

47) What is the marker bit in RTP header? What is the market bit good for?

The interpretation of the marker is defined by a profile. It is intended to allow significant events such as frame boundaries to be marked in the packet stream. A profile MAY define additional marker bits or specify that there is no marker bit by changing the number of bits in the payload type field

48)What is RTCP? What are the main functions of RTCP? (slide 245)

RTCP provides **out-of-band statistics** (e.g., packet loss, packet delay variation, round-trip delay time) and **control information** for an **RTP session.**

The functionalities of RTCP include:

- **Gathering statistics** on quality aspects of the **media distribution** and **transmitting** this data to the **session media source** and other session **participant**.

- Provisioning **session control functions**. RTCP is a convenient means to reach all session participants. RTP is only transmitted by a media source.

49) Explain why the fraction of the RTCP traffic must be Limited?

The primary function of the transport protocol to carry data is not impaired and the control traffic can be included in the bandwidth specification given to a resource reservation protocol, and so that each participant can independently calculate its share. It is suggested that the fraction of the session bandwidth allocated to RTCP be fixed at 5%. While the value of this and other constants in the interval calculation is not critical, all participants in the session must use the same values so the same interval will be calculated. Therefore, these constants should be fixed for a particular profile.

50) What is FEC? How does FEC work? What are the disadvantages of FEC?

- Forward error correction (FEC) is a method of obtaining error control in data transmission in which the source (transmitter) sends redundant data and the destination (receiver) recognizes only the portion of the data that contains no apparent errors. Because FEC does not require handshaking between the source and the destination, it can be used for broadcasting of data to many destinations simultaneously from a single source.
- In the simplest form of FEC, each character is sent twice. The receiver checks both instances of each character for adherence to the protocol being used. If conformity occurs in both instances, the character is accepted. If conformity occurs in one instance and not in the other, the character that conforms to protocol is accepted. If conformity does not occur in either instance, the character is rejected and a blank space or an underscore (_) is displayed in its place.
- Disadvantages
    - Computation may be more difficult implement
    - Still add bandwidth
    - Add decoder complexity
    - Lower quality (vs. other methods of repair)

51) How does interleaving increase the robustness of FEC? What are the disadvantages of interleaving?
- Increase the robustness: **arrange the packet** to reduce burst errors
- Advantages
    - Most audio compression schemes can do interleaving without additional complexity
    - No extra bandwidth added
- Disadvantages
    - Delay of interleaving factor in packets
        - Even when not repairing!
    - Gains to quality can be moderate

52) What is RTSP? Explain the operation of RTSP. How is it compared with HTTP streaming? (slide 253)

RTSP is a network control protocol (port number is 554), designed for controlling streaming media servers. It is used to establish and control media session between end-points. Most RTSP servers use the RTP and RTCP for media stream delivery.

How RTSP works: (Source: http://www.informit.com/articles/article.aspx?p=169578&seqNum=3)

- The client establishes a TCP connection to the servers, typically on TCP port 554, the well-known port for RTSP.
- The client will then commence issuing a series of RTSP header commands that have a similar format to HTTP, each of which is acknowledged by the server.
- Once the negotiation of transport parameters has been completed, the client will issue a PLAY command to instruct the server to commence delivery of the RTP data stream.
- Once the client decides to close the stream, a TEARDOWN command is issued along with the Session ID instructing the server to cease the RTP delivery associated with that ID.

Compare to HTTP:

- Similarly: RTSP defines control sequences useful in controlling multimedia playback and uses TCP to maintain end-to-end connection.
- Difference: RTSP has state. Request can be made by both the streaming server and client.

53) What is the relationship between RTP, RTCP, and RTSP?

- RTCP is a part of RTP and helps with lip synchronization and QOS management, among others.
- RTSP is a control protocol that initiating and directing delivery of streaming multimedia from media servers, the "Internet VCR remote control protocol". RTSP does not deliver data (though the RTSP connection may be used to tunnel RTP traffic for ease of use with firewalls and other network devices). RTP and RTSP will likely be used together in many systems, but either protocol can be used without the other. The RTSP draft contains a section on the use of RTP with RTSP.

- It's important to distinguish between RTP and Real-Time Streaming protocol (RTSP), another transfer protocol. RTSP is used when viewers communicate with a unicast server. RTSP allows two-way communication; that is, viewers can communicate with the streaming server and do things like rewind the movie, go to a chapter, and so on. By contrast, RTP is a one-way protocol used to send live or stored streams from the server to the client.

54) Describe the basic network elements of H.323. (slide 259-262)

H.323 terminal can be either a personal computer (PC) or a stand-alone device, running H.323 and the multimedia application.

H.323 gateway connects two dissimilar networks.

H.323 gatekeeper is an optional component in the H.323 network that provides a number of services to terminals, gateways and MCU devices.

H.323 MCU is responsible for managing multipoint conferences. It consists of two logical entities, i.e., Multipoint Controller (Call signaling, conference control) and Multipoint Processor (switching/mixing of media streams).

55) Describe the basic network elements of SIP (Session Initiation Protocol). (slide 276-279)

A User Agent (UA) takes input from a user and acts as an agent on his behalf to set up and tear down media sessions with other UAs. The user can be a human or another protocol (e.g., gateway). UAs typically register with a proxy server in their domain.

A SIP Proxy Server receives a request from a UA or another proxy and acts on behalf of the UA in forwarding or responding to the request. It does not issue requests and has no media capabilities. Stateful proxy server (e.g., forking proxy server) keeps track of requests and responses received in the past.

A SIP Redirect Server accepts a SIP request, maps the SIP address of the callee into zero (if there is no known address) or a new address and returns it to the UA. It does not pass the request to other servers.

A SIP Registrar Server accepts REGISTER requests for the purposes of updating a location database with the contact information of the user, specified in the request.

SIP Back-to-back User Agent: A B2BUA is a type of UA that receivers a SIP request, then reformulates the request and sends it out as a new request.

A SIP gateway is an application that interfaces a SIP network to a network utilizing another signaling protocol. SIP gateway is a special type of UA, there the UA acts on behalf on another protocol.

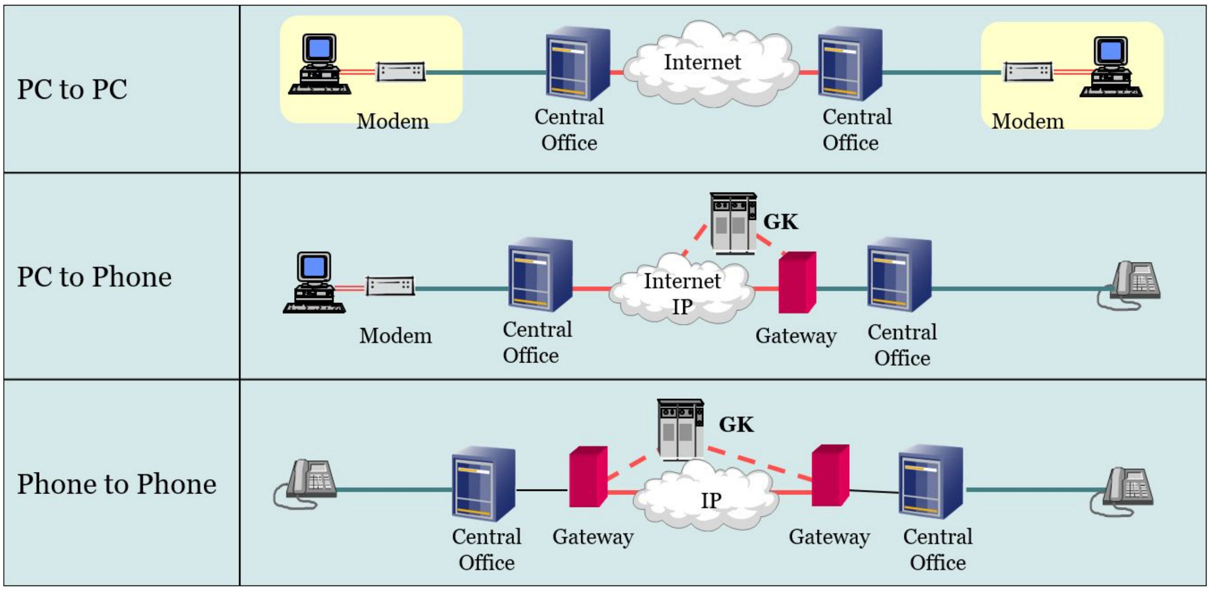
56)How do RSTP, RTP and H.323 relate to one another?

H.323 is a recommendation define the protocols providing communication sessions on any packet network. H.323 was the first VoIP standard to adopt RTP (Real-time Transport Protocol) to transport audio and video over IP networks. RTCP is a sister of RTP, it provide out-of-band statistic for RTP sessions (packet count, packet loss, packet delay variation,...)

57)What are the differences between VoIP and PSTN? (slide 288)

Feature	VoIP	PSTN
Connectivity type	Internet connectivity	Dedicated telephone lines
Required bandwidth	~ 10 Kbps in each direction	~ 64 Kbps in each direction
Pricing	Free VoIP calling (local and international), Calls to mobile and landline phones have nominal subscription fees	No free calls can be made Costly international calling and monthly phone plans.
Scalability	More bandwidth and simple software updates	More dedicated lines and hardware
Remote extensions	Typically standard	Require dedicated lines for each extension and is very pricey
Disaster recovery	Service terminates when internet connectivity is lost	Service remains active during power outages. But cordless phones would be unusable

58)Discuss the different VoIP scenarios. (slide 290)



59)How are the Video Conferences (VC) classified? (slide 295-296)

- Ad-hoc Conference
- Scheduled Conference
- Video Conferencing over ISDN
- Video conferencing over IP networks (e.g., ADSL, FTTH)

60) What are the advantages and disadvantages of the distributed VC over the centralized VC?

Advantages of DVCS (compared with centralized systems) include:

- Allows users to work productively when not connected to a network.
- Common operations (such as commits, viewing history, and reverting changes) are faster for DVCS, because there is no need to communicate with a central server.With DVCS, communication is only necessary when sharing changes among other peers.
- Allows private work, so users can use their changes even for early drafts they do not want to publish.
- Working copies effectively function as remote backups, which avoids relying on one physical machine as a single point of failure.
- Allows various development models to be used, such as using development branches or a Commander/Lieutenant model.
- Permits centralized control of the "release version" of the project
- On FOSS software projects it is much easier to create a project fork from a project that is stalled because of leadership conflicts or design disagreements.

Disadvantages of DVCS (compared with centralized systems) include:

- Initial checkout of a repository is slower as compared to checkout in a centralized version control system, because all branches and revision history are copied to the local machine by default.

- The lack of locking mechanisms that is part of most centralized VCS and still plays an important role when it comes to non-mergeable binary files such as graphic assets.
- Additional storage required for every user to have a complete copy of the complete codebase history.

- Advantage:
    - Centralized MCU need to send multiple stream of video directly to many users at another campus, and it waste a huge amount of bandwidth. On the other hand, in distributed VC, there is only one video stream from one MCU to another MCU through WAN connection, and that reduce a lot of bandwidth consumption
    - Reduce the workload of a single MCU unit
- Disadvantage: Costly since we need to deploy multiple MCU