Computer and Multimedia Networks

- Basics of Computer and Multimedia Networks
- Multiplexing Technologies
- Quality of Multimedia Data Transmission
- Multimedia over IP
- Multimedia over ATM Networks
- Transport of MPEG-4
- Media-on-Demand (MOD)

Basics of Computer and Multimedia Networks

- Computer networks are essential to modern computing.
- Multimedia networks share all major issues and technologies of computer networks.
- The ever-growing needs for various multimedia communications have made networks one of the most active areas for research and development.
- Various high-speed networks are becoming a central part of most contemporary multimedia systems.

OSI Network Layers

OSI Reference Model has the following network layers:

1. **Physical Layer**: Defines electrical and mechanical properties of the physical interface, and specifies the functions and procedural sequences performed by circuits of the physical interface.

2. **Data Link Layer**: Specifies the ways to establish, maintain and terminate a link, e.g., transmission and synchronization of data frames, error detection and correction, and access protocol to the Physical layer.

3. **Network Layer**: Defines the routing of data from one end to the other across the network. Provides services such as addressing, internetworking, error handling, congestion control, and sequencing of packets.

OSI Network Layers (Cont'd)

4. **Transport Layer**: Provides end-to-end communication between *end systems* that support end-user applications or services. Supports either *connection-oriented* or *connectionless* protocols. Provides error recovery and flow control.

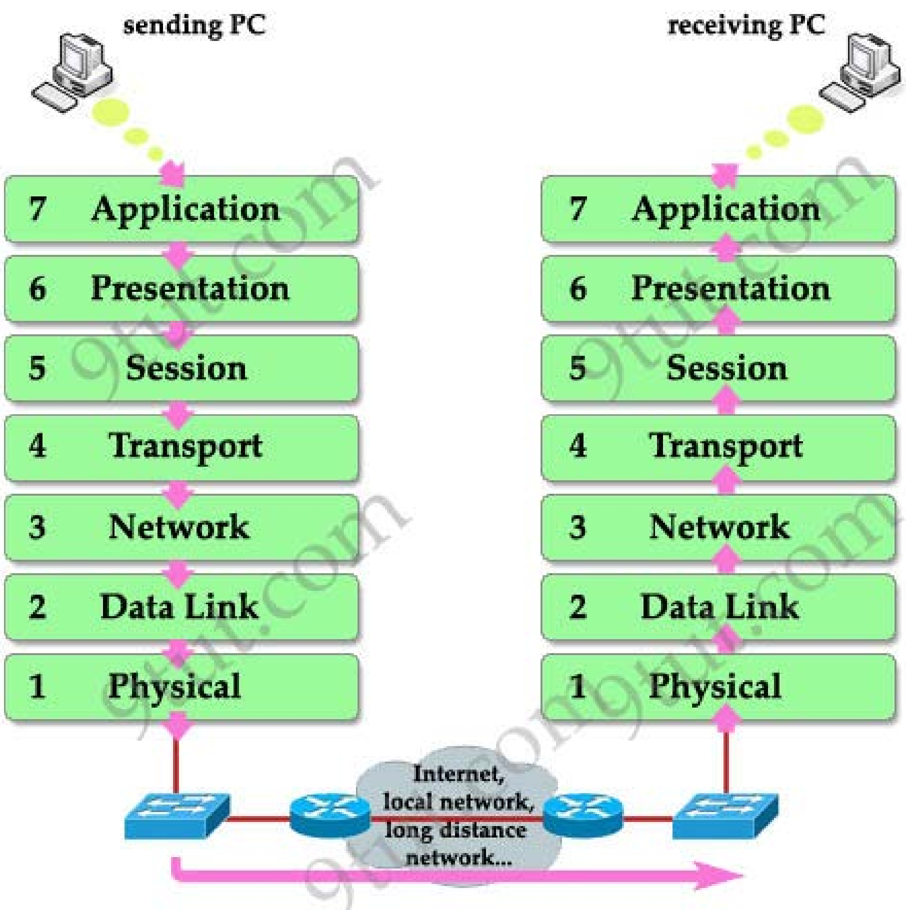
5. **Session Layer**: Coordinates interaction between user applications on different hosts, manages sessions (connections), e.g., completion of long file transfers.

6. **Presentation Layer**: Deals with the syntax of transmitted data, e.g., conversion of different data formats and codes due to different conventions, compression or encryption.

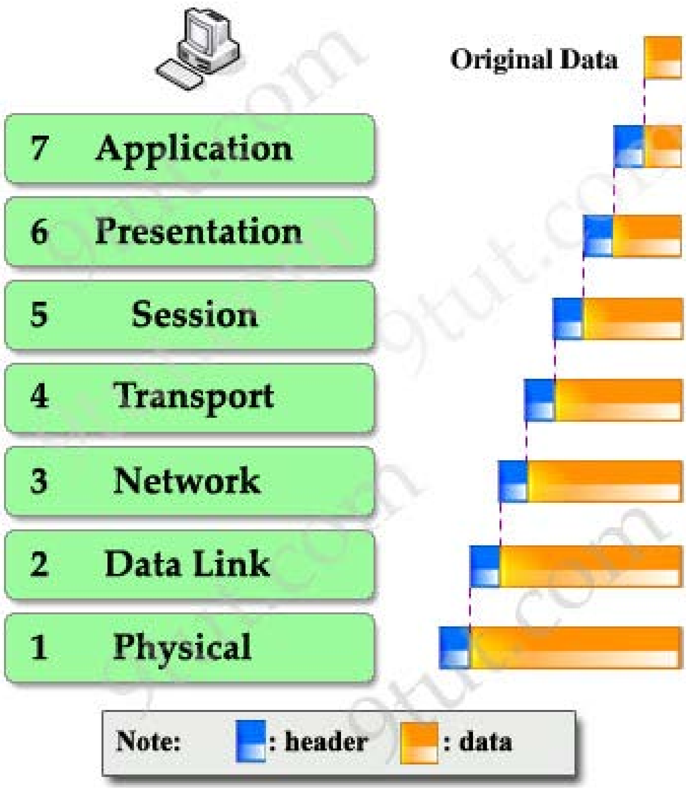
7. **Application Layer**: Supports various application programs and protocols, e.g., FTP, Telnet, HTTP, SNMP, SMTP/MIME, etc.

OSI Network Layers (Cont'd)

Please Do Not Throw Sausage Pizza Away



OSI Network Layers (Cont'd)



- When the information goes down through layers, a header is added to it. This is called "encapsulation".
- Each header can be understood only by the corresponding layer at the receiving side. Other layers only see that layer's header as a part of data.
- At the receiving side, corresponding header is stripped off in the same layer it was attached. This process is called "decapsulation".

http://www.9tut.com/osi-model-tutorial

TCP/IP Protocols

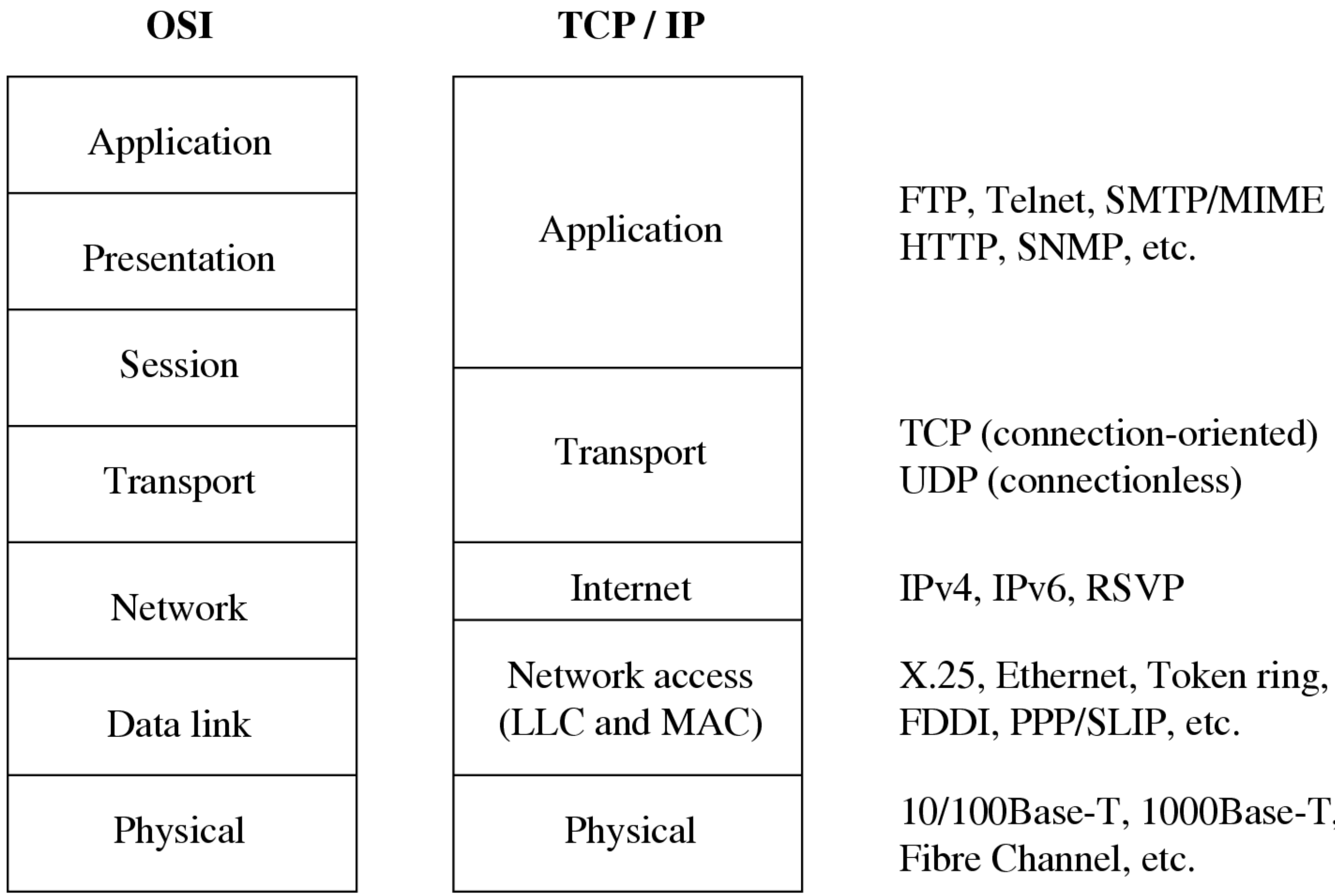


Fig: Comparison of OSI and TCP/IP protocol architectures

Transport Layer- TCP (Transmission Control Protocol)

- *Connection-oriented:* it provides reliable data transfer between pairs of communicating processes across the network.
- Established for packet switched networks only; data have to be packetized.
- Relies on the IP layer for delivering the message to the destination computer specified by its IP address.
- Provides message packetizing, error detection, retransmission, packet resequencing and multiplexing.
- Although reliable, the overhead of retransmission in TCP may be too high for many real-time multimedia applications such as streaming video —UDP can be used instead.

Transport Layer – TCP (Cont'd)

- Each TCP datagram header contains the source and destination port, sequence number, check sum, window field, acknowledgement number and other fields.

TCP header:

<table>
<tr><td>00</td><td>01</td><td>02</td><td>03</td><td>04</td><td>05</td><td>06</td><td>07</td><td>08</td><td>09</td><td>10</td><td>11</td><td>12</td><td>13</td><td>14</td><td>15</td><td>16</td><td>17</td><td>18</td><td>19</td><td>20</td><td>21</td><td>22</td><td>23</td><td>24</td><td>25</td><td>26</td><td>27</td><td>28</td><td>29</td><td>30</td><td>31</td></tr>
<tr><td colspan="16">Source Port</td><td colspan="16">Destination Port</td></tr>
<tr><td colspan="32">Sequence Number</td></tr>
<tr><td colspan="32">Acknowledgment Number</td></tr>
<tr><td colspan="4">Data Offset</td><td colspan="3">reserved</td><td colspan="3">ECN</td><td colspan="6">Control Bits</td><td colspan="16">Window</td></tr>
<tr><td colspan="16">Checksum</td><td colspan="16">Urgent Pointer</td></tr>
<tr><td colspan="32">Options and padding :::</td></tr>
<tr><td colspan="32">Data :::</td></tr>
</table>

http://www.networksorcery.com/enp/protocol/tcp.htm

Transport Layer - UDP (User Datagram Protocol)

- *Connectionless*: the message to be sent is a single Datagram.
- The only thing UDP provides is multiplexing and error detection through a Checksum.
- The source port number in UDP header is optional since there is no acknowledgment.
- Much faster than TCP, however it is unreliable:
  - In most real-time multimedia applications (e.g., streaming video or audio), packets that arrive late are simply discarded.
  - Flow control, and congestion avoidance, more realistically error concealment must be explored for acceptable Quality of Service (QoS).

Network Layer - IP (Internet Protocol)

- Two basic services: **packet addressing** and **packet fragmentation**.

- **Packet addressing**:

  - The IP protocol provides for a global addressing of computers across all interconnected networks.

  - For an IP packet to be transmitted within LANs, either broadcast based on hubs or point-to-point transmission based on switch is used.

  - For an IP packet to be transmitted across WANs, Gateways or routers are employed, which use routing tables to direct the messages according to destination IP addresses.

Network Layer — IP (Internet Protocol) (Cont'd)

- The IP layer also has to:
  - translate the destination IP address of incoming packets to the appropriate network address.
  - identify for each destination IP the next best router IP through which the packet should travel based on routing table.

- Since the best route can change depending on node availability, network congestion and other factors, routers have to communicate with each other to determine the best route for groups of IPs. The communication is done using *Internet Control Message Protocol (ICMP)*.

- IP is *connectionless* — provides no end-to-end flow control, packets could be received out of order, and dropped or duplicated.

Network Layer — IP (Internet Protocol) (Cont'd)

- **Packet fragmentation**: performed when a packet travels over a network that only accepts packets of a smaller size.
  - In that case, IP packets are split into the required smaller size, sent over the network to the next hop, and reassembled and resequenced.
- IP versions:
  - IPv4 (IP version 4): IP addresses are *32* bit numbers, usually specified using *dotted decimal notation* (e.g. 128.77.149.63) — running out of new IP addresses soon (projected in year 2008).
  - IPv6 (IP version 6): The *next generation IP (IPng)* - adopts 128-bit addresses, allowing $2^{128} \approx 3.4 \times 10^{38}$ addresses.

Multiplexing Technologies

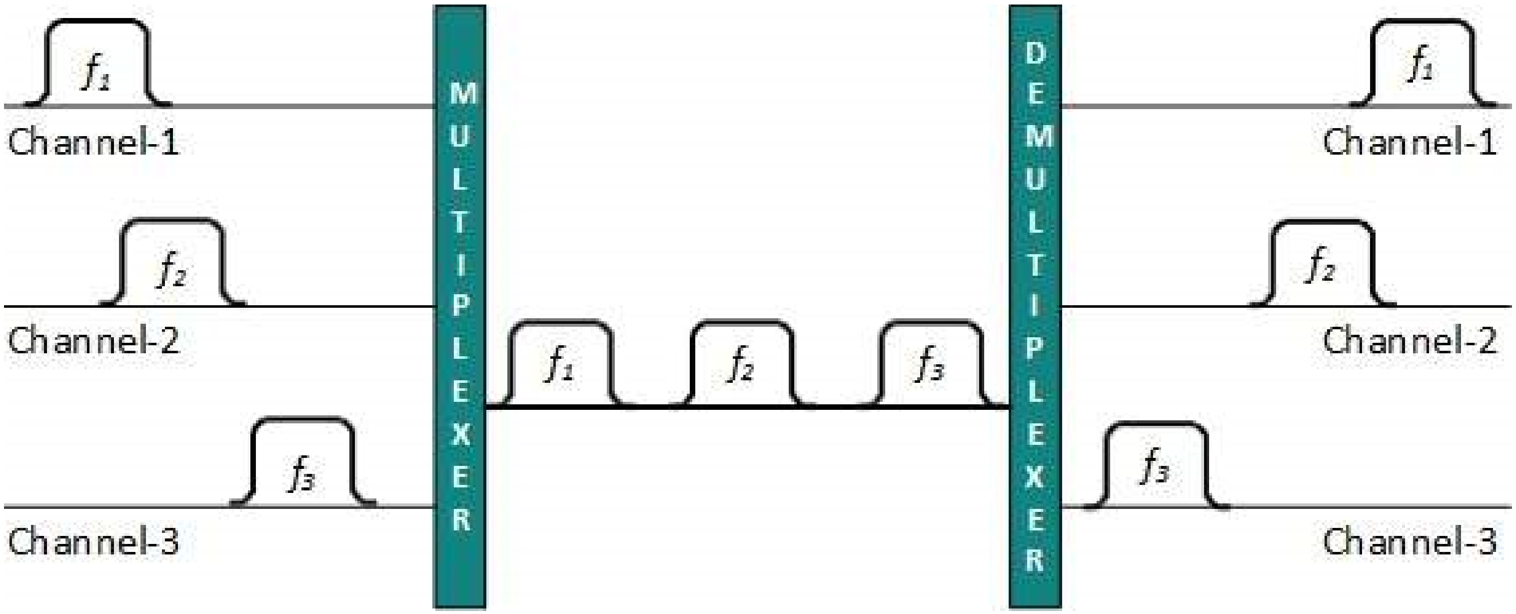
- **Basics of Multiplexing**

1. **FDM (Frequency Division Multiplexing)** — Multiple *channels* are arranged according to their frequency (e.g. radios and TVs):

   - For FDM to work properly, analog signals must be *modulated* so that the signal occupies a bandwidth B_s centered at f_c — *carrier* frequency unique for each channel.

   - The receiver uses a band-pass filter tuned for the particular channel-of-interest to capture the signal, and then uses a demodulator to decode it.

   - Basic modulation techniques: *Amplitude Modulation (AM)*, *Frequency Modulation (FM)*, *Phase Modulation (PM)*, and *Quadrature Amplitude Modulation (QAM)*.

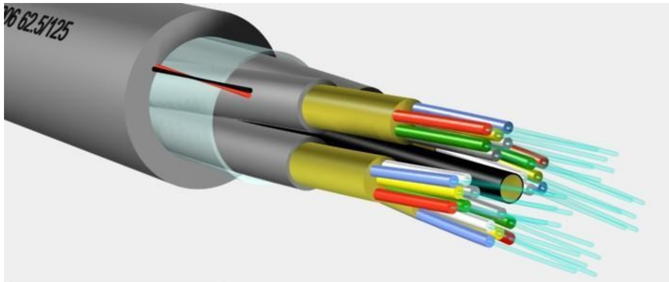

f1
Channel-1
f2
Channel-2
f3
Channel-3
MULTIPLEXER
f1
f2
f3
DEMULTIPLEXER
f1
Channel-1
f2
Channel-2
f3
Channel-3

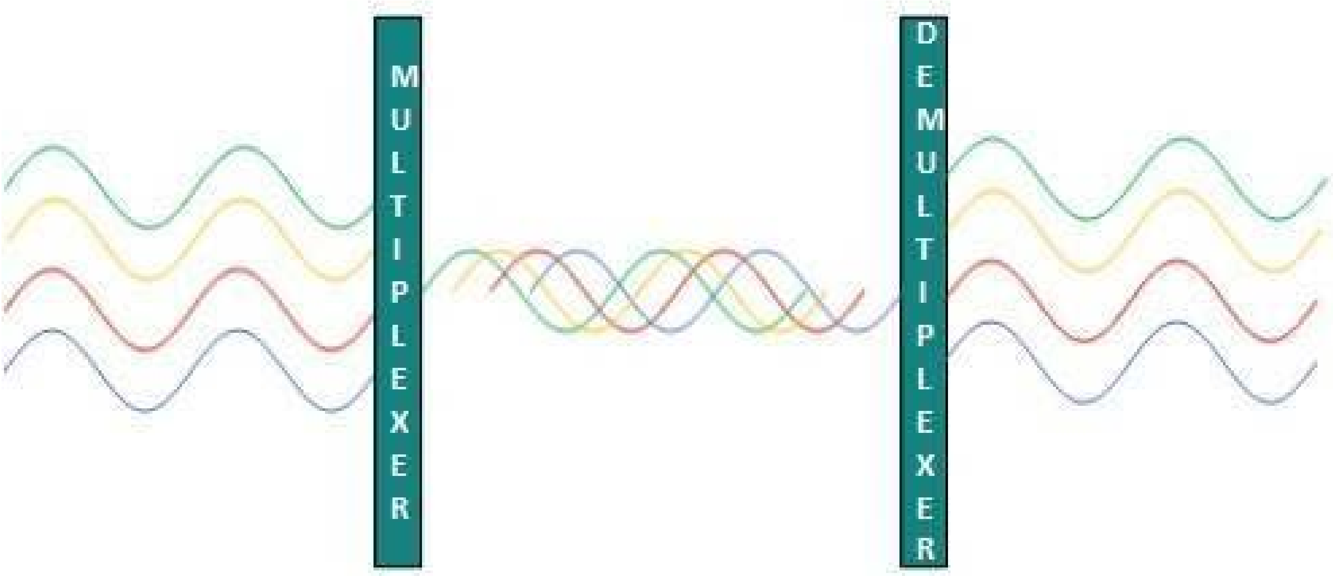
2. **WDM (Wavelength Division Multiplexing)**: A variation of FDM for data transmission in optical fibers:

- Light beams representing channels of different wave-lengths are combined at the source, and split again at the receiver.
- The capacity of WDM is tremendous — a huge number of channels can be multiplexed (aggregate bit-rate can be up to dozens of terabits per second).
- Two variations of WDM:

  a) **DWDM** (Dense WDM): employs densely spaced wavelengths so as to allow a larger number of channels than WDM (e.g., more than 32).

  b) **WWDM** (Wideband WDM): allows the transmission of color lights with a wider range of wavelengths (e.g., 1310 to 1557 nm for long reach and 850 nm for short reach) to achieve a larger capacity than WDM.

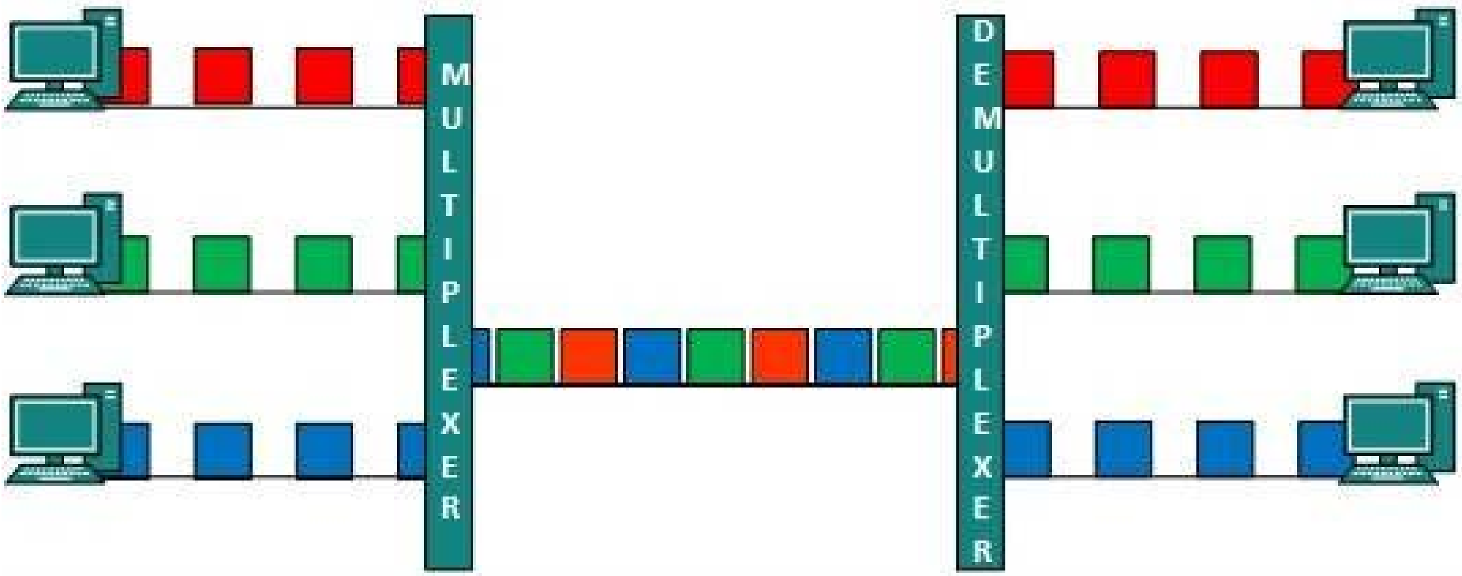



MULTIPLEXER
DEMULTIPLEXER

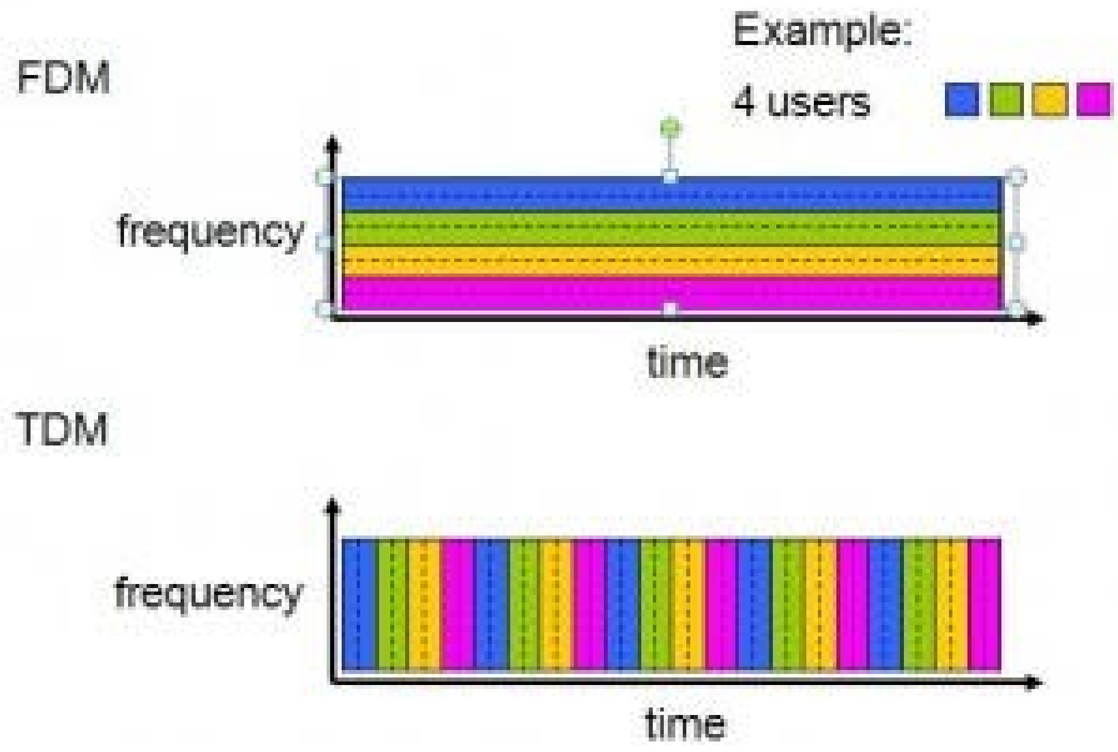
3. **TDM (Time Division Multiplexing)** — A technology for directly multiplexing digital data:

- If the source data is analog, it must first be digitized and converted into PCM (Pulse Code Modulation).
- Multiplexing is performed along the time (t) dimension.
  Multiple buffers are used for m ($m > 1$) channels.
- Two variations of TDM:
  a) **Synchronous TDM**: Each of the m buffers is scanned in turn and treated equally. If, at a given time slot, some sources (accordingly buffers) do not have data to transmit the slot is wasted.

  b) **Asynchronous TDM**: Only assign k ($k < m$) time slots to scan the k buffers that are likely to have data to send (based on statistics) — has the potential of having a higher throughput given the same carrier data rate.



MULTIPLEXER
DEMULTIPLEXER



FDM vs. TDM

ADSL (Asymmetric Digital Subscriber Line)

- Adopts a higher data rate downstream and lower data rate upstream, hence *asymmetric*.
- Makes use of existing telephone twisted-pair lines to transmit QAM (Quadrature Amplitude Modulated) digital signals.
- Bandwidth on ADSL lines: 1 MHz or higher.
- ADSL uses FDM to multiplex three channels:
  a) High speed (1.5 to 9 Mbps) downstream channel at the high end of the spectrum
  b) Medium speed (16 to 640 kbps) duplex channel.
  c) POTS (Plain Old Telephone Service) channel at the low end (next to DC, 0-4 kHz) of the spectrum.

ADSL (Asymmetric Digital Subscriber Line)

- Because signals attenuate quickly on twisted-pair lines, and noise increases with line length, ADSL has the distance limitation when using only ordinary twisted-pair copper wires.

Data rate	Wire Size	Distance
1.544 Mbps	0.5 mm	5.5km
1.544 Mbps	0.4 mm	4.6km
6.1 Mbps	0.5 mm	3.7km
6.1 Mbps	0.4 mm	2.7km

Table. History of Digital Subscriber Lines

Name	Meaning	Data Rate	Mode
V.32 or V.34	Voice Band Modems	1.2 to 56 kbps	Duplex
DSL	Digital Subscriber Line	160 kbps	Duplex
HDSL	High Data Rate Digital Subscriber Line	1.544 Mbps or 2.048 Mbps	Duplex
SDSL	Single Line Digital Subscriber Line	1.544 Mbps or 2.048 Mbps	Duplex
ADSL	Asymmetric Digital Subscriber Line	1.5 to 9 Mbps 16 to 640 kbps	Down Up
VDSL	Very high data rate Digital Subscriber Line	13 to 52 Mbps 1.5 to 2.3 Mbps	Down Up

Table offers a brief history of various digital subscriber lines (**xDSL**).

Characteristics of Multimedia Data

- **Voluminous** — they demand very high data rates, possibly dozens or hundreds of Mbps.

- **Real-time and interactive** — they demand low delay and synchronization between audio and video for "lip sync". In addition, applications such as video conferencing and interactive multimedia also require two-way traffic.

- **Sometimes bursty** — data rates fluctuate drastically, e.g., no traffic most of the time but burst to high volume in video-on-demand.

Quality of Multimedia Data Transmission

- **Quality of Service (QoS)** depends on many parameters:
  - **Data rate**: a measure of transmission speed.
  - **Latency (maximum frame/packet delay)**: maximum time needed from transmission to reception.
  - **Packet loss or error**: a measure (in percentage) of error rate of the packetized data transmission.
  - **Jitter**: a measure of smoothness of the audio/video playback, related to the variance of frame/packet delays.
  - **Sync skew**: a measure of multimedia data synchronization.

Multimedia Service Classes

- **Real-Time** (also *Conversational*): two-way traffic, low latency and jitter, possibly with prioritized delivery, e.g., voice telephony and video telephony.
- **Priority Data**: two-way traffic, low loss and low latency, with prioritized delivery, e.g., E-commerce applications.
- **Silver**: moderate latency and jitter, strict ordering and sync. One-way traffic, e.g., streaming video, or two-way traffic (also *Interactive*), e.g., web surfing, Internet games.
- **Best Effort** (also *Background*): no real-time requirement, e.g., downloading or transferring large files (movies).
- **Bronze**: no guarantees for transmission.

Requirement on Network Bandwidth / Bit-rate

Application	Speed Requirement
Telephone	16 kbps
Audio-conferencing	32 kbps
CD-quality audio	128–192 kbps
Digital music (QoS)	64–640 kbps
H. 261	64 kbps–2 Mbps
H. 263	$<$ 64 kbps
DVI video	1.2–1.5 Mbps
MPEG-1 video	1.2–1.5 Mbps
MPEG-2 video	4–60 Mbps
HDTV (compressed)	$>$ 20 Mbps
HDTV (uncompressed)	$>$ 1 Gbps
MPEG-4 video-on-demand (QoS)	250–750 kbps
Videoconferencing (QoS)	384 kbps–2 Mbps

Tolerance of Latency and Jitter in Digital Audio and Video

Application	Avg Latency Tolerance (msec)	Avg Jitter Tolerance (msec)
Low-end videoconf. (64 kbps)	300	130
Compressed voice (16 kbps)	30	130
MPEG NTSC video (1.5 Mbps)	5	7
MPEG audio (256 kbps)	7	9
HDTV video (20 Mbps)	0.8	1

Perceived QoS

- Although QoS is commonly measured by the above technical parameters, QoS itself is a "**collective effect of service performances that determine the degree of satisfaction of the user of that service**".
- In other words, it has everything to do with how the user *perceives* it. For example, in real-time multimedia:
  - Regularity is more important than latency (i.e., jitter and quality fluctuation are more annoying than slightly longer waiting).
  - Temporal correctness is more important than the sound and picture quality (i.e., ordering and synchronization of audio and video are of primary importance).
  - Humans tend to focus on one subject at a time. User focus is usually at the center of the screen, and it takes time to refocus especially after a scene change.

QoS for IP Protocols

- **IP** is a *best-effort* communications technology-hard to provide QoS over IP by current routing methods.
  - Abundant bandwidth improves QoS, but unlikely to be available everywhere over a complex networks.

Sol1:

- **DiffServ** (Differentiated Service) uses DiffServ code [*TOS* (Type of Service) octet in IPv4 packet, and Traffic Class octet in IPv6 packet] to classify packets to enable their differentiated treatment.
  - Widely deployed in intra-domain networks and enterprise networks as it is simpler and scales well.
  - Emerging as the QoS technology in conjunction with other QoS.

QoS for IP Protocols (Cont'd)

Sol2:

- **MPLS** (*Multiple Protocol Label Switching*) facilitates the marriage of IP to OSI Layer 2 technologies.
  - Creates tunnels: **Label Switched Paths (LSP)** — IP network becomes connection-oriented.
  - Main advantages of MPLS:

    a) Support Traffic Engineering (TE), which is used essentially to control traffic flow.

    b) Support VPN (Virtual Private Network).

    c) Both TE and VPN help delivery of QoS for multimedia data.

Prioritized Delivery

- Used to alleviate the perceived deterioration (high packet loss or error rate) in network congestion.
- **Prioritization for types of media**:
  - Transmission algorithms can provide prioritized delivery to different media.
- **Prioritization for uncompressed audio**:
  - PCM audio bitstreams can be broken into groups of every n^{th} sample.
- **Prioritization for JPEG image**:
  - The different *scans* in Progressive JPEG and different resolutions of the image in Hierarchical JPEG can be given different priorities.
- **Prioritization for compressed video**:
  - Set priorities to minimize playback delay and jitter by giving highest priority to I-frames for their reception, and lowest priority to B-frames.

Multimedia over IP

- A *broadcast* message is sent to all nodes in the domain, a *unicast* message is sent to only one node, and a *multicast* message is sent to a set of specified nodes.
- IP-multicast is vital for applications such as mailing list, bulletin boards, group file transfer, audio/video on demand, audio/videoconferencing

- **MBone** (Multicast Backbone) — based on the IP-multicast technology:
  - Used for audio and video conferencing on the Internet .
  - Uses a subnetwork of routers (*mrouters*) that support multicast to forward multicast packets.

Multimedia over IP

- **IP-Multicast**:
  - Anonymous membership: the source host multicasts to one of the IP-multicast addresses — doesn't know who will receive.
  - Potential problem: too many packets will be traveling and alive in the network — use time-to-live (TTL) in each IP packet.
  - based on UDP-limited reliability.

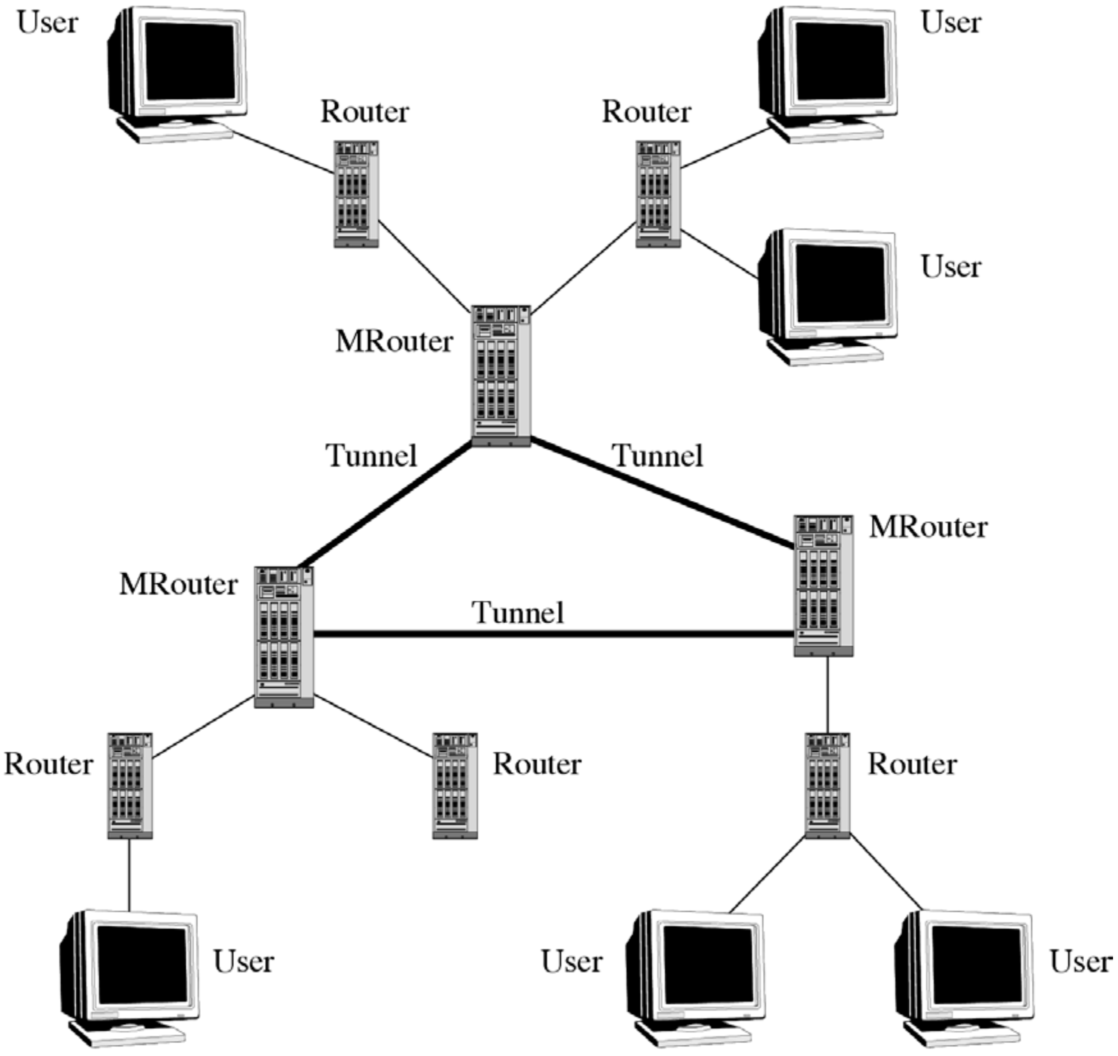


Fig: Tunnels for IP Multicast in MBone.

Internet Group Management Protocol (IGMP)

- Designed to help the maintenance of multicast groups.

- Two special types of IGMP messages are used:
  - **Query** messages are multicast by routers to all local hosts to inquire group membership.
  - **Report** is used to respond to a query and to join groups.

- On receiving a query, members wait for a random time before responding.

- Routers periodically query group membership, and declare themselves group members if they get a response to at least one query. If no responses occur after a while, they declare themselves as non members.

RTP (Real-time Transport Protocol)

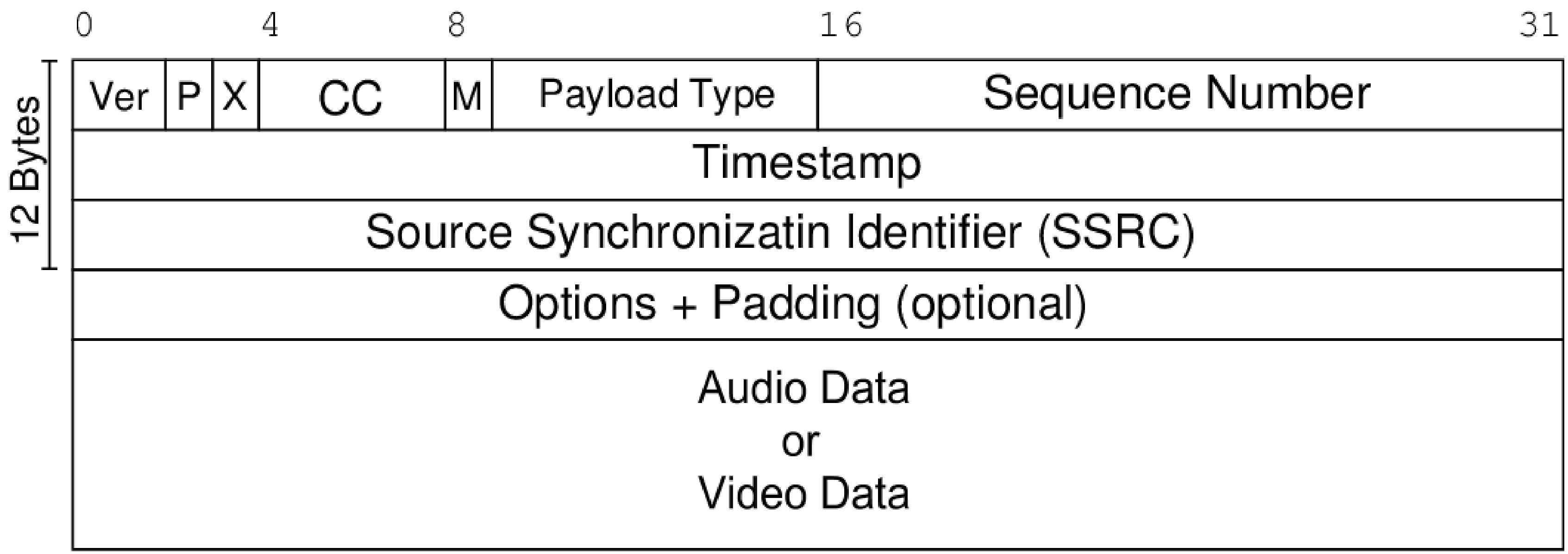
- Designed for the transport of real-time data, such as audio and video streams:
  - Primarily intended for multicast.
- Usually runs on top of UDP which provides efficient (but less reliable) connectionless datagram service:
  - RTP must create its own *timestamping* and *sequencing* mechanisms to ensure the ordering.

RTCP (Real Time Control Protocol)

- A companion protocol of RTP:
  - Monitors QoS in providing feedback to the server (sender) and conveys information about the participants of a multi-party conference.
  - Provides the necessary information for audio and video synchronization.

- RTP and RTCP packets are sent to the same IP address (multicast or unicast) but on different ports.

RTCP (Real Time Control Protocol)



RTP packet header

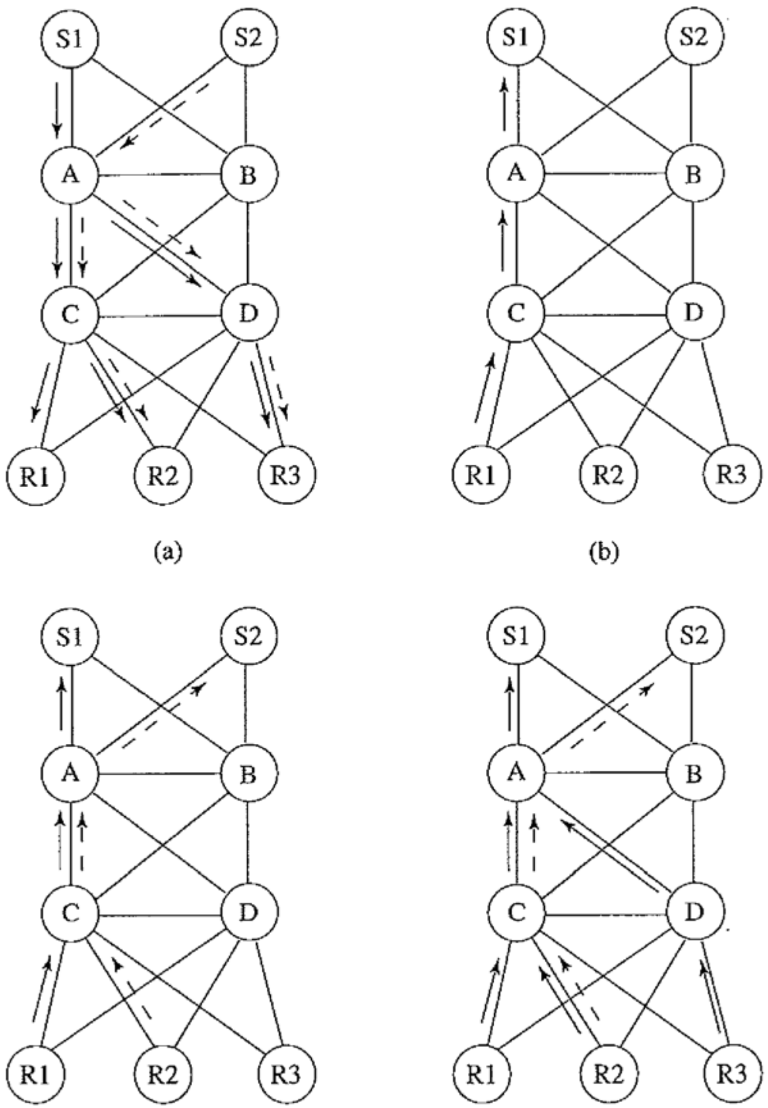
RSVP (Resource ReSerVation Protocol)

- Aim to guarantee desirable QoS, mostly for multicast but also applicable to unicast.

  - A general communication model supported by RSVP consists of m senders and n receivers, possibly in various multicast groups.

- The most important messages of RSVP:
  1. A **Path** message is initiated by the sender, and contains information about the sender and the path (e.g., the previous RSVP hop).
  2. A **Resv** message is sent by a receiver that wishes to make a reservation.

RSVP



(a) (b)

(c) (d)

Main Challenges of RSVP

- Plenty of senders and receivers compete for the limited network bandwidth.

- The receivers can be heterogeneous in demanding different contents with different QoS.

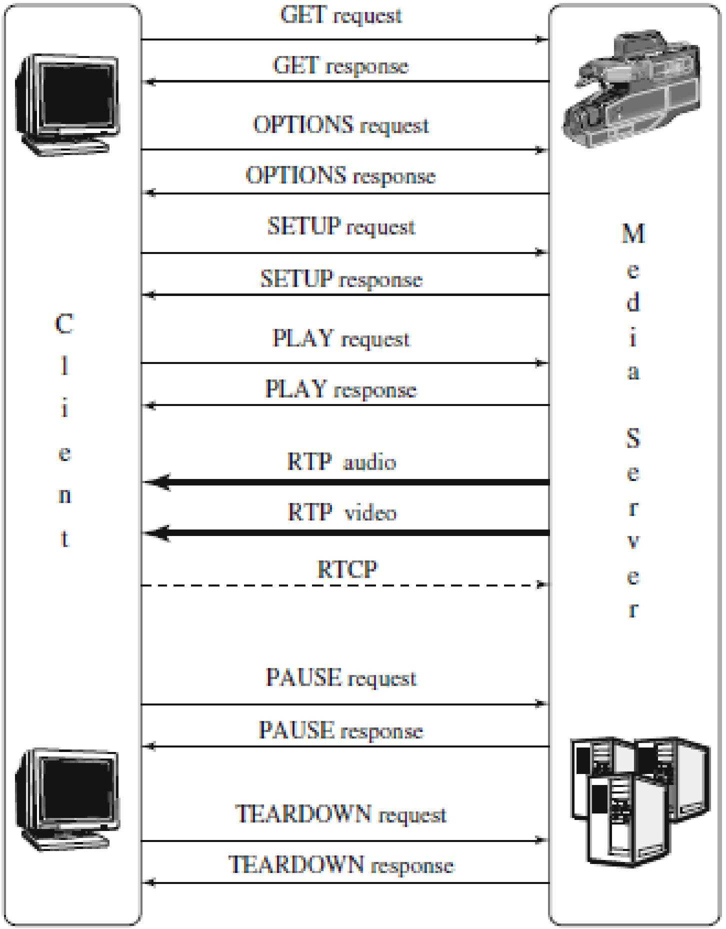
- They can be dynamic by joining or quitting multicast groups at any time.

RTSP (Real Time Streaming Protocol)

- **Streaming Audio and Video**: Audio and video data that are transmitted from a stored media server to the client in a data stream that is almost instantly decoded.
- **RTSP Protocol**: for communication between a client and a stored media server.

1. **Requesting presentation description**: the client issues a DESCRIBE request to the Stored Media Server to obtain the presentation description — media types, frame rate, resolution, codec, etc.
2. **Session setup**: the client issues a SETUP to inform the server of the destination IP address, port number, protocols, TTL (for multicast).
3. **Requesting and receiving media**: after receiving a PLAY, the server started to transmit streaming audio/video data using RTP.
4. **Session closure**: TEARDOWN closes the session.

RTSP (Real Time Streaming Protocol)



A possible scenario of RTSP operations

Internet Telephony

- Main advantages of Internet telephony over *POTS* (*Plain Old Telephone Service*):
  - Uses *packet-switching* — network usage is much more efficient (voice communication is bursty and VBR-encoded).
  - With the technologies of *multicast or multipoint* communication, multi-party calls are not much more difficult than two-party calls.
  - With advanced multimedia data compression techniques, various degrees of *QoS* can be supported and dynamically adjusted according to the network traffic.
  - Good *graphics user interfaces* can be developed to show available features and services, monitor call status and progress, etc.

Internet Telephony (Cont'd)

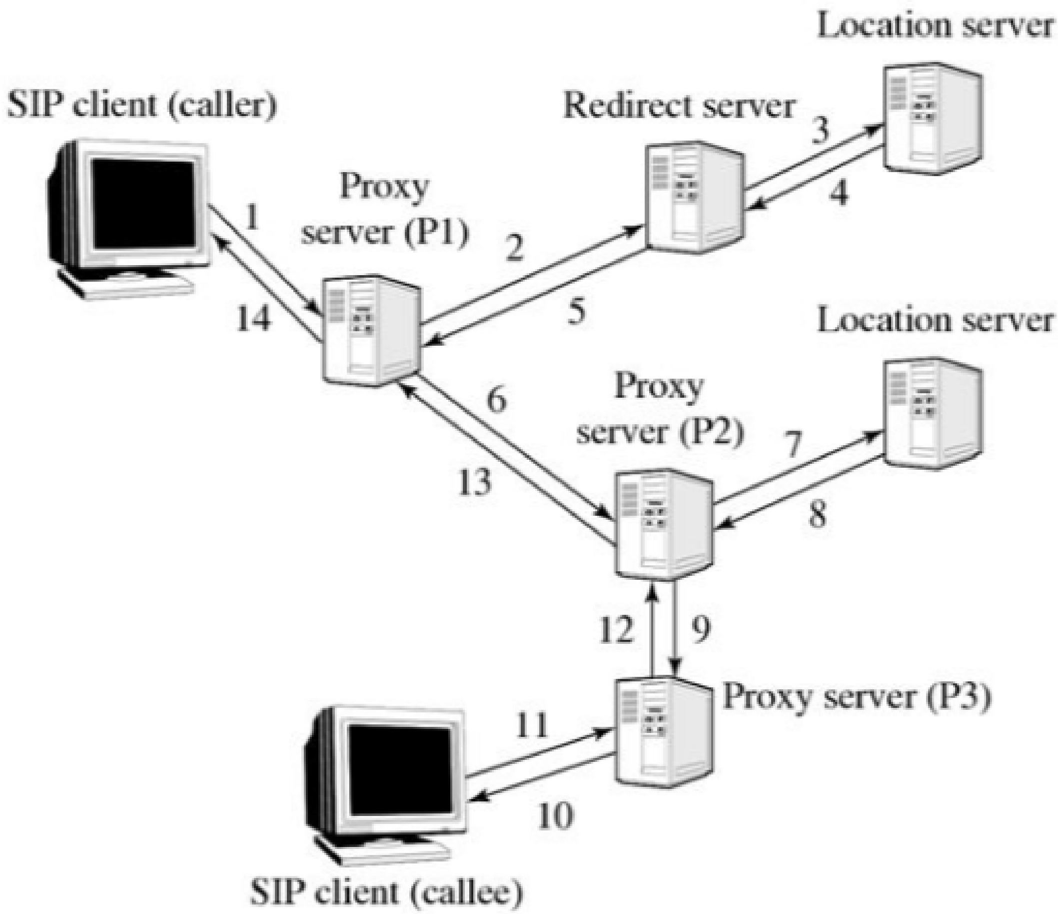
- The transport of real-time audio (and video) in Internet telephony is supported by RTP (whose control protocol is RTCP).

- Streaming media is handled by RTSP and Internet resource reservation is taken care of by RSVP.

SIP (Session Initiation Protocol)

- An *application-layer* control protocol in charge of the establishment and termination of sessions in Internet telephony.
  - SIP is a text-based protocol, also a client-server protocol.
- SIP can advertise its session using email, news group, web pages or directories, or *SAP* — a multicast protocol.
- The *methods* (commands) for clients to invoke:
  - **INVITE**: invites callee(s) to participate in a call.
  - **ACK**: acknowledges the invitation.
  - **OPTIONS**: enquires media capabilities without setting up a call.
  - **CANCEL**: terminates the invitation.
  - **BYE**: terminates a call.
  - **REGISTER**: sends user's location info to a Registrar (a SIP server).

SIP


Location server
SIP client (caller)
Redirect server
3
4
Proxy
server (P1)
1
2
14
5
Location server
6
Proxy
server (P2)
7
13
8
12
9
Proxy server (P3)
11
10
SIP client (callee)

Multimedia over ATM Networks

- ATM forum supports various **Video Bit-rates:**
  - **CBR** (Constant Bit Rate): if the allocated bit-rate of CBR is too low, then cell loss and distortion of the video content are inevitable.
  - **VBR** (Variable Bit Rate): the most commonly used video bit-rate for compressed video, can be further divided into:

    * rt-VBR (real-time Variable Bit Rate): for compressed video.

    * nrt-VBR (non real-time Variable Bit Rate): for specified QoS.
  - **ABR** (Available Bit Rate): data transmission can be backed off or buffered due to congestion. Cell loss rate and minimum cell data rate can sometimes be specified.
  - **UBR** (Unspecified Bit Rate): no guarantee on any quality parameter.

Multicast over ATM

- Multicast in ATM networks had several challenges:
  - ATM is connection-oriented; hence ATM multicasting needs to set up all multipoint connections.
  - QoS in ATM must be negotiated at the connection set-up time and be known to all switches.
- It is difficult to support multipoint-to-point or multipoint-to-multipoint connections in ATM
  - Do not keep track of multiplexer number or sequence number. It cannot reassemble the data correctly at the receiver side if cells from different senders are interleaved at their reception.

Transport of MPEG-4

- **DMIF**: An interface between multimedia applications and their transport. It supports:

  1. Remote interactive network access (IP, ATM, PSTN, ISDN, mobile).
  2. Broadcast media (cable or satellite).
  3. Local media on disks.

- The following fig. shows the integration of delivery through three types of communication mediums.

- **MPEG-4 over IP**: MPEG-4 sessions can be carried over IP-based protocols such as RTP, RTSP, and HTTP.

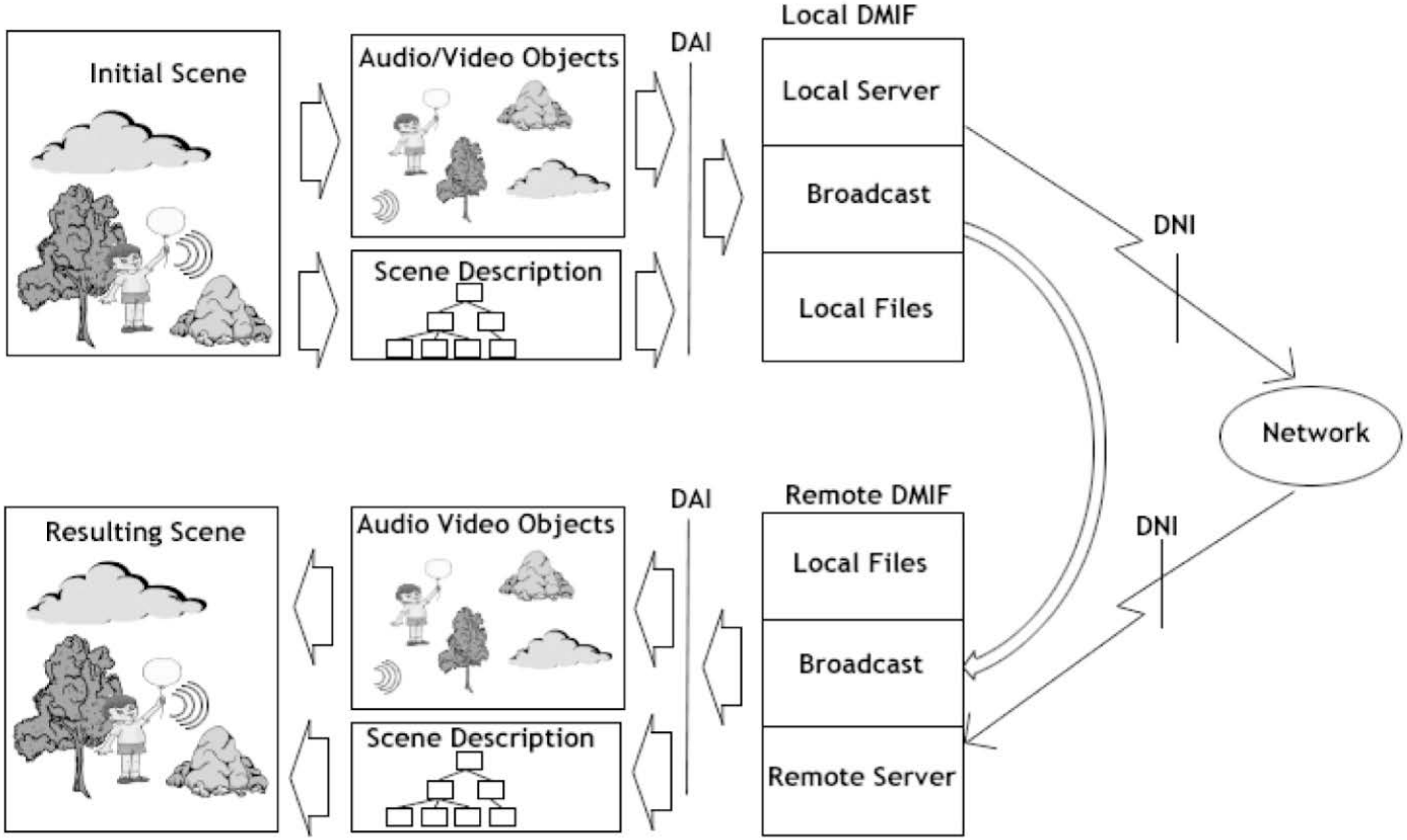


Fig. DMIF — the multimedia content delivery integration framework.

Media-on-Demand (MOD)

- **Interactive TV (ITV) and Set-top Box (STB)**
  - ITV supports activities such as:
    1. TV (basic, subscription, pay-per-view).
    2. Video-on-demand (VOD).
    3. Information services (news, weather, magazines, sports events, etc.).
    4. Interactive entertainment (Internet games, etc.).
    5. E-commerce (on-line shopping, stock trading).
    6. Access to digital libraries and educational materials.

Set-top Box (STB)

- Set-top Box (STB) generally has the following components:

1. **Network Interface and Communication Unit**: including tuner and demodulator, security devices, and a communication channel.

2. **Processing Unit**: including CPU, memory, and special-purpose operating system for the STB.

3. **Audio/Video Unit**: including audio and video (MPEG-2 and 4) decoders, DSP (Digital Signal Processor), buffers, and D/A converters.

4. **Graphics Unit**: supporting real-time 3D graphics for animations and games.

5. **Peripheral Control Unit**: controllers for disks, audio and video I/O devices (e.g., digital video cameras), CD/DVD reader and writer, etc.

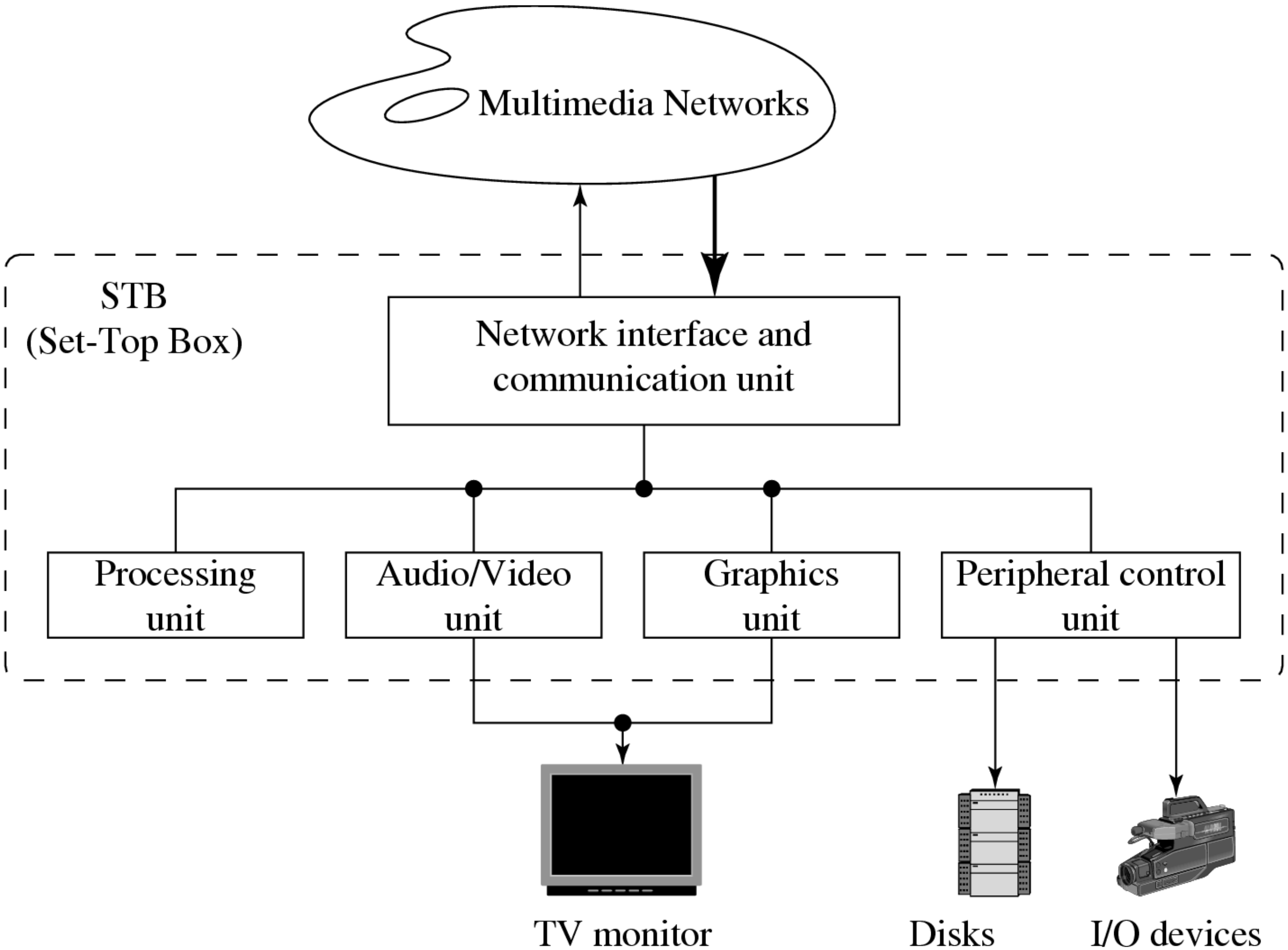


Fig: General Architecture of STB (Set-top Box).

Further Exploration

- **Text books:**
  - *Computer Networks* by A.S. Tanenbaum
  - *Data & Computer Communications* by W. Stalling

- **Further Topics**:
  - SONET FAQ, etc.
  - xDSL introductions at DSL Forum website.
  - Introductions and White Papers on ATM.
  - FAQ and White Papers on 10 Gigabit Ethernet at the Alliance website.
  - IEEE 802 standards.
  - IETF RFCs: IPv6 (Internet Protocol, Version 6).