Multimedia Networking

**Motivation**

- The ever-growing needs for various *multimedia communications* have made networks one of the most active areas for research and development.

- **Example 1: Multimedia streaming applications:**
  - Streaming means a user can listen (or watch) the file after the downloading has started.
  - 1) Streaming stored audio and video
    - On-demand requests for compressed audio/video files
  - 2) Streaming live audio and video
    - Broadcasting of radio and TV programs through the Internet
  - 3) Real-time interactive audio and video
    - Virtual classroom

- **Example 2:**
  - Multimedia application sharing
Network oriented classification of Media Types

- Non-Real Time
  - E.g. Text, Data
  - E.g. Images
    - Discrete
      - E.g. Text chat, Instant Messaging
      - E.g. Weather Updates
    - Delay Intolerant
      - E.g. Remote Desktop Applications
    - Delay Tolerant
      - E.g. Interactive Audio/Video
      - E.g. Streaming Audio/Video

- Real Time
  - Continuous
Characteristics of Multimedia Data

**Voluminous**

- They demand very high data rates, possibly dozens or hundreds of Mbps.

<table>
<thead>
<tr>
<th>Application</th>
<th>Speed Requirement</th>
</tr>
</thead>
<tbody>
<tr>
<td>Telephone</td>
<td>16 kbps</td>
</tr>
<tr>
<td>Audio-conferencing</td>
<td>32 kbps</td>
</tr>
<tr>
<td>CD-quality audio</td>
<td>128–192 kbps</td>
</tr>
<tr>
<td>Digital music (QoS)</td>
<td>64–640 kbps</td>
</tr>
<tr>
<td>H. 261</td>
<td>64 kbps–2 Mbps</td>
</tr>
<tr>
<td>H. 263</td>
<td>&lt; 64 kbps</td>
</tr>
<tr>
<td>DVI video</td>
<td>1.2–1.5 Mbps</td>
</tr>
<tr>
<td>MPEG-1 video</td>
<td>1.2–1.5 Mbps</td>
</tr>
<tr>
<td>MPEG-2 video</td>
<td>4–60 Mbps</td>
</tr>
<tr>
<td>HDTV (compressed)</td>
<td>&gt; 20 Mbps</td>
</tr>
<tr>
<td>HDTV (uncompressed)</td>
<td>&gt; 1 Gbps</td>
</tr>
<tr>
<td>MPEG-4 video-on-demand (QoS)</td>
<td>250–750 kbps</td>
</tr>
<tr>
<td>Videoconferencing (QoS)</td>
<td>384 kbps–2 Mbps</td>
</tr>
</tbody>
</table>
Characteristics of MM Data (Cont.)

- **Real-time and interactive**
  - They demand **low delay**. In addition, applications such as video conferencing and interactive multimedia also require **two-way traffic**.

- **Temporal relationship between data**
  - Sequencing (playing frames in **correct order/time** frame in video)
  - Synchronization (**inter-media scheduling**, e.g. Lip synchronization)

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

- **Delay sensitive, but loss tolerant**
  - Infrequent losses cause minor glitches
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**: The variability of packet delays within the same packet stream.
- **Sync skew**: a measure of multimedia data synchronization.
- ...
Latency and Jitter

Multimedia Networking

Latency and Jitter

Client-side buffering, playout delay compensate for network-added delay and variance of frame/packet delays.
Jitter (High or Low)

Jitters in frame playbacks. (a) High jitter, (b) Low jitter.
Multimedia Networking

Providing Real-time Traffic when Jitter Exists

- **Timestamp** the packets and separate the arrival time from the playback time.
- A playback **buffer** is required for real-time traffic.
- A **sequence number** on each packet is required for real-time traffic.
- Changing the **encoding** of a payload to a lower **quality** to match the bandwidth of the receiving network.
Perceived QoS

- Although QoS is commonly measured by the mentioned 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.
Streaming Stored Multimedia

- Downloading these types of files from a Web server can be different from downloading other types of files. To understand the concept, we will discuss several approaches, each with a different complexity:
  - **First Approach:** Using a Web Server
  - **Second Approach:** Using a Web Server with Metafile
  - **Third Approach:** Using a Media Server
  - **Fourth Approach:** Using a Media Server and RTSP
1. Using a Web Server

* Audio or video stored in file
* Files transferred as **HTTP** object
* Received in entirety at client then passed to player

Note: audio and video are not streamed, long delays until playout.

2. Using a Web Server with Metafile

* Browser GETs metafile
* Browser launches player, passing metafile
* Player contacts web server
* Web server streams audio/video to player

Metafile is a generic term for a file format that can store multiple types of data. e.g. consider a container format whose specification describes how different data elements and metadata coexist in a computer file.
3. Using a Media Server

* Allows for non-HTTP protocol between media server and media player (e.g. UDP or TCP?)
4. Using a Media Server and RTSP

* Metafile communicated to web browser
* Browser launches player
* Player sets up an RTSP control connection, and data connection to streaming server

The Real Time Streaming Protocol (RTSP) is a network control protocol designed for use in entertainment and communications systems to control streaming media servers.
Real Time Streaming Protocol (RTSP)

The protocol is used for establishing and controlling media sessions between end points. Clients of media servers issue VCR-like commands, such as play and pause, to facilitate real-time control of playback of media files from the server.

RTSP requests: OPTIONS, DESCRIBE, SETUP, PLAY, PAUSE, RECORD, TEARDOWN

The transmission of streaming data itself is not a task of the RTSP protocol. Most RTSP servers use the Real-time Transport Protocol (RTP) for media stream delivery.
Example Media Servers

QuickTime Streaming Server (Apple)
- Advertised as: Using the RTP/RTSP open standard, QuickTime Streaming Server lets you deliver live or prerecorded content in real time over the Internet. With Instant-On, your content begins to play as soon its link is clicked; there’s no waiting for the file to download.

Windows Media Services (Microsoft)
- Advertised as: Windows Media Services 9 Series has advanced streaming functionality and native 64-bit support for even higher scalability.

Helix Universal Media Server (RealNetworks)
- Advertised as: The Helix Universal Media Server features full 64-bit support, eliminating file size limits and creating an HD-quality experience. Other features include bookmarking and mobile delivery of multi-format content – including Flash, H.264, MPEG-4 (MP4), QuickTime, MP3 and RealVideo – to PCs, smartphones, iPhones and tablets.
Multimedia Networking

Streaming Stored Multimedia

Media Player Functions
- Jitter removal
- Decompression
- Error concealment
- Graphical user interface with controls for interactivity

Examples
- VLC media Player
- Media Player Classic
- QuickTime
- Windows Media Player
- RealPlayer

QuickTime Player 7.6.6 running on Microsoft Windows
### Multimedia Networking

**Streaming Stored Multimedia**

<table>
<thead>
<tr>
<th>Media Servers</th>
<th>Media Players</th>
</tr>
</thead>
</table>

- QuickTime Streaming Server: Apple's closed-source streaming server that ships with Mac OS X Server.
- Darwin Streaming Server: Open-sourced version of QuickTime Streaming Server maintained by Apple.
- Pys3: Formerly called PacketVideo Streaming Server, this is Alcatel-Lucent's streaming server product.
- Helix Universal Server: RealNetworks' commercial streaming server for RTSP, RTMP, iPhone OS, Silverlight and HTTP streaming media clients.
- Helix DNA Server: RealNetworks' streaming server. Comes in both open-source and proprietary flavors.
- LIVE555: Open source C++ server and client libraries used in well-known clients like VLC and mplayer.
- Fang: Lean and mean streaming server with focus on rfc compliance.
- VideoLAN: Open source media player and streaming server.
- Windows Media Services: Microsoft's streaming server included with Windows Server.
- Xenon Streaming Server: Mobile streaming server from Vidiator Technology (US) Inc.
- RtpRtpStack: Streaming server which is designed for low footprint and high performance applications.
- Gst: GStreamer based RTSP Server and client.
- FFmpeg: includes fserver a GPL or LGPL RTSP streaming server.
- Envyvideo: has RTSP client and can retransmit video to other protocols.
- ViaMotion: integrated RTSP server for Video On Demand by Anevia

- cURL (beginning with version 7.20.0—9 February 2010[^1])
- FFmpeg (undocumented[^2])
  1. Live playback: ffmpeg rtsp://some-server.youtube.com/some-media.3gp
  2. Saving live broadcast to a file: ffmpeg -i rtsp://some-server.youtube.com/some-media.3gp -acodec copy -vcodec copy filename.3gp

- GStreamer
- Media Player Classic
- MPEG4IP
- MPlayer
- Winamp
- VLC media player
- Xine

Multimedia Networking

Streaming Live Multimedia

- **Example:** Broadcasting of radio and TV programs through the Internet

- **Similarities with stored multimedia streaming:**
  - Both are sensitive to delay
  - Both can not accept retransmission

- **Differences with stored multimedia streaming:**
  - **The type of communication:**
    - Stored MM streaming: Uni-cast and on-demand
    - Live MM streaming: Multi-cast and live

**Sending message over IP:**

- A **broadcast** message is sent to all nodes in the domain.
- A **unicast** message is sent to only one node.
- A **multicast** message is sent to a set of specified nodes.
Multimedia Networking

Real-time Interactive Multimedia

Example:
- Internet phone or voice over IP
- Video Conferencing

Later we will see more about this topic.
Computer Networks

Computer Networks
LAN and WAN

- LAN (Local Area Network) is restricted to a small geographical area, usually to a relatively small number of stations.

- WAN (Wide Area Network) refers to networks across cities and countries.

- MAN (Metropolitan Area Network) is sometimes also used to refer to the network between LAN and WAN.
Computer Networks

Internet: Network of Networks

- Server
- Workstation
- Mobile
- Router: Forward Packets (Chunks of data)

End Systems: Hosts

Communication Links: Copper, Coaxial Cable, Optical Fiber, Radio systems, Satellite

Bandwidth: Transmission Rate

Protocols: Control sending, receiving of msgs. e.g. TCP, IP, HTTP, FTP

local ISP

regional ISP

corporate network
Two types of data transmission through a network:

- **Circuit switching**: dedicated circuit per call: telephone net
- **Packet switching**: data sent thru net in discrete “chunks”
Circuit Switching

- An end-to-end circuit must be established that is dedicated for the entire duration of the connection at a guaranteed bandwidth.
- Initially designed for voice communications.
- Pros:
  - End-end resources reserved for “call” (Dedicated resources).
  - Circuit-like (guaranteed) performance.
    - Constant bit-rate, Short transmission delay, Small delay jitters.
- Cons:
  - No resource sharing (resource or resource "piece" idle if not used by owning call). resource "piece" is described in multiplexing slide.
  - Inefficient for general multimedia communications, especially for variable (sometimes bursty) data rates (Inefficient for multi-users with variable data rates).
In order to cope with multi-users and variable data rates, it adopts FDM or Synchronous TDM multiplexing techniques. 
Dividing link bandwidth into “pieces”: Frequency division and Time division

Example: 4 Users

FDM
(Frequency Division Multiplexing)

TDM
(Time Division Multiplexing)
## Multiplexing

### TDM

<table>
<thead>
<tr>
<th>Format</th>
<th>Num of channels</th>
<th>Data Rate (Mbps)</th>
<th>Format</th>
<th>Num of channels</th>
<th>Data Rate (Mbps)</th>
</tr>
</thead>
<tbody>
<tr>
<td>T1</td>
<td>24</td>
<td>1.544</td>
<td>E1</td>
<td>32</td>
<td>2.048</td>
</tr>
<tr>
<td>T2</td>
<td>96</td>
<td>6.312</td>
<td>E2</td>
<td>128</td>
<td>8.448</td>
</tr>
<tr>
<td>T3</td>
<td>672</td>
<td>44.736</td>
<td>E3</td>
<td>512</td>
<td>34.368</td>
</tr>
<tr>
<td>T4</td>
<td>4032</td>
<td>274.176</td>
<td>E4</td>
<td>2048</td>
<td>139.264</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>E5</td>
<td>8192</td>
<td>565.148</td>
</tr>
</tbody>
</table>

Comparison of TDM Carrier Standards
Packet Switching

- Each end-end data stream divided into packets.
  - Data is broken into small packets, usually of 1,000 bytes or less in length. The header of each packet will carry necessary control information such as destination address, routing, etc.

Sequence of A & B packets does not have fixed pattern \( \rightarrow \) Statistical multiplexing.

In TDM each host gets same slot in revolving TDM frame.

Protocols: Internet protocol (IP), X.25 (older one)

x.25 is an ITU-T standard protocol suite for packet switched wide area network (WAN) communication. While X.25 has been, to a large extent, replaced by less complex protocols, especially the Internet protocol (IP), the service is still used and available in niche and legacy applications.
Packet Switching, Data Packet

- **Header**
  - Packet length, packet number
  - Source and destination routing information (IP addresses)
  - Synchronization, transmission protocol

- **Payload**
  - Packet body containing data to be transmitted

- **Trailer or footer**
  - Cyclic redundancy check: parity checking on the payload
Packet Switching

Pros:

- **Resource sharing**: Resources used as needed (Efficient use of the network resources); Suitable for applications which require dynamic bandwidth (e.g. VBR compressed video).
- **If** one branch gets too busy or **broken**, then the packets are automatically **routed through another path** instead.
- **As customers increase**, the network only has to expand slowly compared to circuit switching (Considering that the users are active e.g. 10% of the time).
Packet Switching

Cons:

- **Latency**: The time it takes to put back the data package, changes each time, which can be a problem for time-critical information such as an emergency signal.

- **Addressing Overhead**: Not very good for small data packages - for example if the data package itself is only 600 bytes long, then two packets of 512 bytes need to be used, plus the address information.

- **Resource Contention**: e.g. congestion may occur (packets queue, wait for link use)
  - If the link is busy, packets are queued in the buffer. Packets are **dropped** if buffer is full.
Packet Switching: Queuing

FIFO queues

Priority queues

Weighted fair queuing

The switch turns to the other queue when the current one is empty.

The turning switch selects 3 packets from first queue, then 2 packets from the second queue, then 1 packet from the third queue. The cycle repeats.
Goal: move packets through routers from source to destination

There are many path selection (i.e. routing) algorithms

**Virtual circuit network:**
- Fixed path determined at call setup time, remains fixed thru call
- Each packet carries tag (virtual circuit ID), tag determines next hop
- Routers maintain per-call state

**Datagram network:**
- Routes may change during session
- Destination address in packet determines next hop
- Analogy: driving, asking directions
Computer Networks

Network Taxonomy

Telecommunication networks

Circuit-switched networks
- FDM
- TDM

Packet-switched networks
- Networks with VCs
- Datagram Networks
Different Switching Techniques used in WAN:

- Circuit Switching
- Packet Switching
- Frame Relay
- Cell Relay (ATM, Asynchronous Transfer Mode)

Comparison of Different Switching Techniques. Compares the four switching technologies in terms of their bit rate and complexity. It can be seen that Circuit Switching is the least complex and offers constant (fixed) data rate, and Packet Switching is the opposite.
Frame Relay [Supplementary Materials]

- A cheaper version of packet switching with minimal services, working at the data link control layer.
- Frame Relay made the following major changes to X.25:
  - **Reduction of error-checking**: no more acknowledgement, no more hop-to-hop flow control and error control.
  - **Reduction of layers**: the multiplexing and switching of virtual circuits are changed from Layer 3 in X.25 to Layer 2. Layer 3 of X.25 is eliminated.
  - Frames have a length up to **1,600 bytes**. When a bad frame is received, it will simply be discarded | very high data rate: ranging from T1 (1.5 Mbps) to T3 (44.7 Mbps).
Cell Relay (ATM)

- Small and fixed-length (53 bytes) packets are adopted in cells.
- The small packet size is beneficial in reducing latency in ATM networks. When the darkened packet arrives slightly behind another packet of a normal size (e.g., 1 kB):
  - (a) It must wait for the completion of the other's transmission, hence serialization delay.
  - (b) Much less waiting time is needed for the darkened cell to be sent.
- Significantly increases the network throughput especially beneficial for real-time multimedia applications.

(a) Serialization delay in a normal packet switching network.  
(b) Lower latency in a cell network.
A fixed format: 53 bytes, of which the first 5 bytes are for the cell header, followed by **48 bytes of payload**.

The ATM Layer has two types of interfaces: UNI (User Network Interface) is local, between a user and an ATM network, and NNI (Network-Network Interface) is between ATM switches.

The structure of an ATM UNI cell header:

<table>
<thead>
<tr>
<th>GFC</th>
<th>VPI</th>
<th>VCI</th>
<th>PT</th>
<th>CLP</th>
<th>HEC</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>8</td>
<td>16</td>
<td>24</td>
<td>32</td>
</tr>
</tbody>
</table>

| 0   | 8   | 16  | 24 | 32  | 40  |

- **GFC** = General Flow Control
- **VPI** = Virtual Path Identifier
- **VCI** = Virtual Channel Identifier
- **PT** = Payload Type
- **CLP** = Cell Loss Priority
- **HEC** = Header Error Check
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.
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.
# Computer Networks

## TCP/IP Protocols

### Comparison of OSI and TCP/IP protocol architectures

<table>
<thead>
<tr>
<th>OSI</th>
<th>TCP/IP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Application</td>
<td>FTP, Telnet, SMTP/MIME</td>
</tr>
<tr>
<td>Presentation</td>
<td>HTTP, SNMP, etc.</td>
</tr>
<tr>
<td>Session</td>
<td>TCP (connection-oriented)</td>
</tr>
<tr>
<td>Transport</td>
<td>UDP (connectionless)</td>
</tr>
<tr>
<td>Network</td>
<td>IPv4, IPv6, RSVP</td>
</tr>
<tr>
<td>Data link</td>
<td>X.25, Ethernet, Token ring</td>
</tr>
<tr>
<td>Physical</td>
<td>X.25, Ethernet, Token ring, FDDI, PPP/SLIP, etc.</td>
</tr>
<tr>
<td></td>
<td>10/100Base-T, 1000Base-T, Fibre Channel, etc.</td>
</tr>
</tbody>
</table>
Transport Layer, TCP and UDP

TCP (Transmission Control Protocol)

- **Goal:** data transfer between end systems.
- **Connection-oriented**
  - **Handshaking:** setup (prepare for) data transfer ahead of time.
- Established for packet switched networks only.
- 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.
Transport Layer, TCP and UDP

TCP (Transmission Control Protocol)

**TCP, Pros**
- **Reliable**, in-order byte-stream data transfer.
- **Loss**: acknowledgements and retransmissions.
- **Flow control**: sender won’t overwhelm receiver.
- **Congestion control**: senders “slow down sending rate” when network congested.

**TCP, Cons**
- Although reliable, the **overhead of retransmission** in TCP may be too high for many real-time multimedia applications such as streaming video (This is why TCP is not suitable for interactive multimedia applications), UDP can be used instead.

**App:**
- HTTP (Web), FTP (file transfer), Telnet (remote login), SMTP (email).
UDP (User Datagram Protocol)

- **Goal:** data transfer between end systems (same as TCP)
- **Connectionless**
  - No handshaking between UDP sender, receiver
  - Each UDP segment handled independently of others
- **Pros:** Much faster than TCP
- **Cons:**
  - Unreliable data transfer
  - No flow control
  - No congestion control
Transport Layer, TCP and UDP

UDP (User Datagram Protocol)

- Note: UDP is more suitable than TCP for interactive multimedia traffic. However, we need the services of RTP, another transport layer protocol, to make up for the deficiencies of UDP.

- Note: In most real-time multimedia applications (e.g., streaming video or audio), packets that arrive late are simply discarded. Error concealment must be explored for acceptable Quality of Service (QoS).

- App: Streaming media, teleconferencing, DNS, Internet telephony
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.
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.
- 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.
  - 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 $2^{128} \sim 3.4*10^{38}$ addresses.
Computer Networks

Find out more at...


Thank You

Next Session: Multimedia Networking, Part II

FIND OUT MORE AT...

1. http://ce.sharif.edu/~m_amiri/