Chapter 2: Multimedia Streaming (Part B)

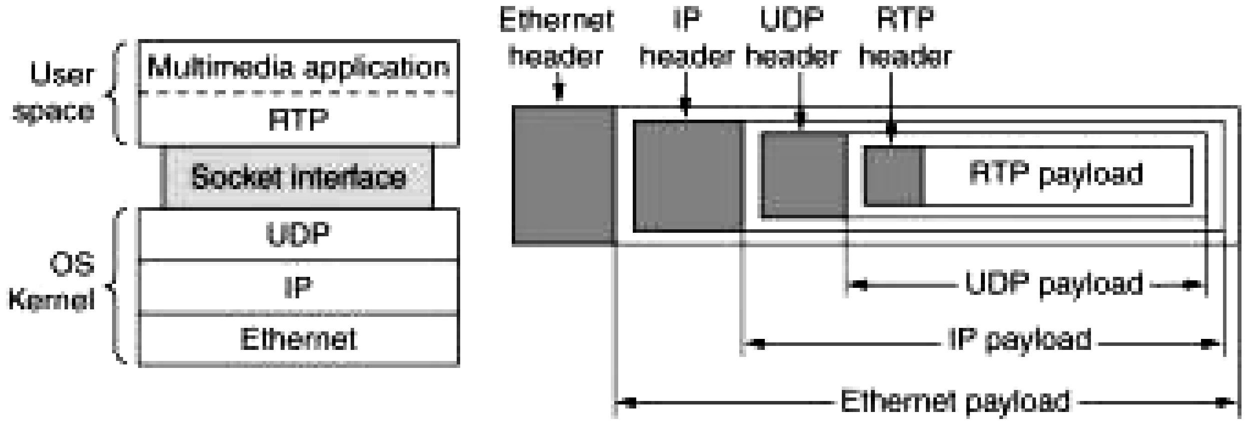
V. Multimedia Streaming

- **Streaming** breaks data into many packets. When the client has received enough packets, the user software will start playing packets. The user can begin listening/watching almost immediately without having to download the entire media file.
- **Multimedia over Internet**
  - Running multimedia applications over packet-switched networks like the Internet is very attractive.
    - The infrastructure often is already in place => save expensive software development.
    - LAN and WAN technologies provide a relatively inexpensive, plentiful, but shared bandwidth over bigger and bigger networks.
  - On the Internet, packets are routed independently across shared networks, so transit times vary significantly. Variations in transit delays are called **jitter**. A playback application is designed to solve the jitter problem by buffering at the receiver.
  - To cope with congestion, several approaches have been proposed (e.g., the application adapts to the available bandwidth by switching to a different encoding).

- **Multimedia Networking: Goals and Challenges**
  - To build the multimedia on network and distributed systems, so different users on different machines can share image, sound, video, voice, etc, and to communicate with each under these tools.
  - Typical applications: internet telephony, multimedia conferencing, distributed simulations, network games, etc.
- **Challenges of Real-time**
  - **Real-time data over non-real-time network**: The Internet is not a real-time computer network.
  - **High Bandwidth Requirement:** multimedia applications usually require much higher bandwidth than traditional textual applications. Examples of high-bandwidth network protocols: Gigabit Ethernet, FDDI, ATM, etc.
  - **Real-time Character:** If the network is congested, real-time data becomes obsolete if it doesn't arrive in time.

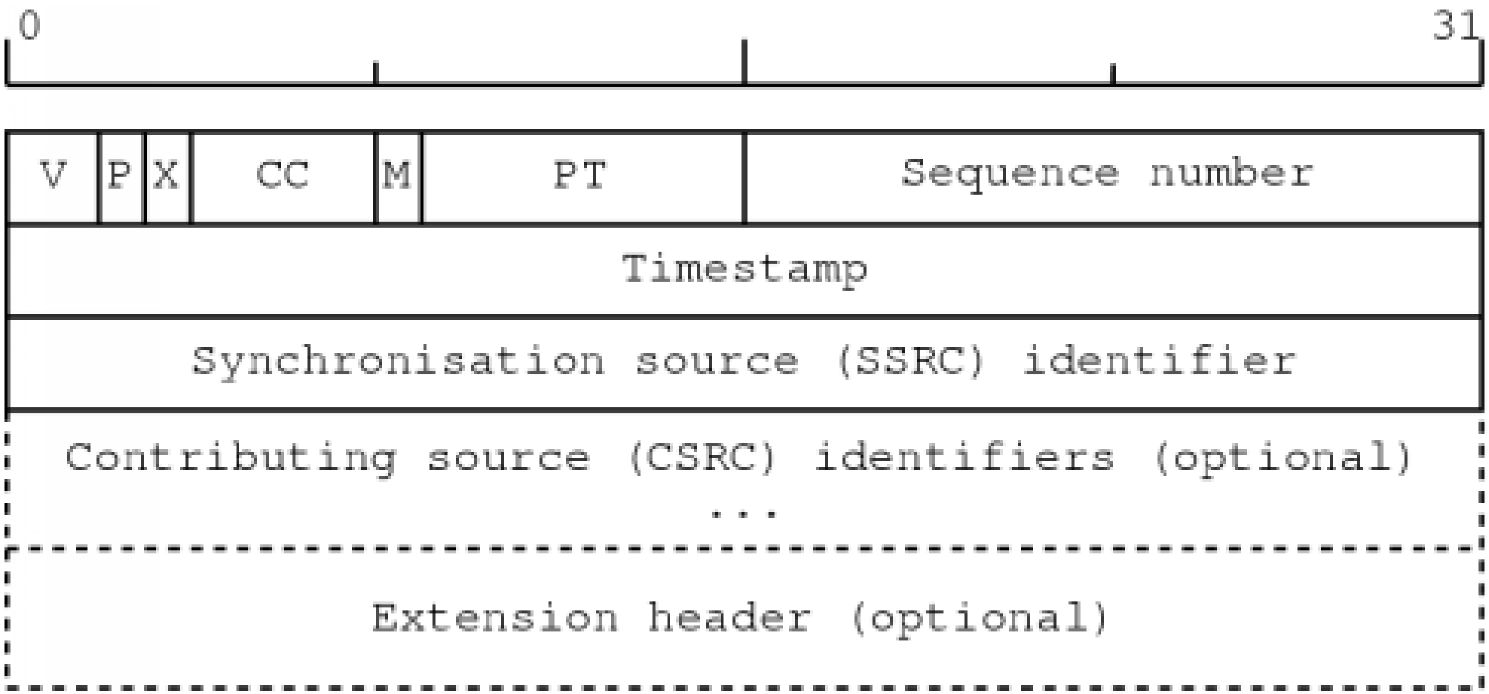
- **Solutions for Multimedia Streaming**
  - 4 protocols have been designed to work together to provide real-time service.
    - Resource Reservation protocol (RSVP);
    - Work together with Real-Time Transport Protocol (RTP), Real-Time Transport Control Protocol (RTCP), Real Time Streaming Protocol (RTSP);
    - It is a comprehensive approach to provide applications with the type of service the people need in the quality they choose.
- RTP (**Real-Time Transport Protocol**) is a protocol which provides end-to-end delivery services for data (such as interactive audio and video) with **real-time** characteristics. It was primarily designed to support multiparty multimedia conferences.
  - **Real-Time:** The correctness of the data depends not only on whether the result is the correct one, but also on the time at which the result is delivered.
- RTP is basically a combination of two parts -
  - Real-Time Transport Protocol (RTP): It carries real-time data.
  - Real-Time Transport Control Protocol (RTCP): It monitors the quality of service by periodically sending statistics information to participants.

- RTP is usually implemented within the application. RTP is designed to be independent from the underlying transport protocol and can be used over unicast as well as multicast.



- In a multimedia session, each medium is carried in a separate RTP session, with its own RTCP packets reporting the reception quality for that session. For example, audio and video would travel on separate RTP sessions, enabling a recipient to select whether or not to receive a particular medium.

- **RTP Header**
  - The RTP header has the following format:



  - The first 12 bytes are present in every RTP packet. The fields have the following meaning:
    - Version (V): Version of RTP.

- Padding (P): If set, the packet contains one or more additional padding octets at the end which are not part of the payload.
- Extension (X): If set, the fixed header is followed by exactly one header extension.
- CSRC Count (CC): The number of CSRC identifiers that follow the fixed header.
- Marker (M): To allow significant events such as frame boundaries to be marked in the packet stream.
- Payload Type (PT): Identifies the format of the RTP payload.
- Sequence Number: Increments by one for each RTP data packet sent. The initial value is randomly set.
- Timestamp: The sampling instant of the first octet in the RTP data packet. Designed for synchronization, etc.
- SSRC: Randomly chosen number to distinguish synchronization sources within the same RTP session.
- CSRC List: To identify the contributing sources for the payload. Can be up to 16 sources, 32 bits long for each.

- **RTP Features**
  - RTP itself does not provide any mechanism to ensure timely delivery or provide other quality of service guarantees, but relies on lower-layer services to do so.
  - RTP is a protocol framework that is deliberately not complete. A particular application should customize its profile.

- RTP provides functionality suited for carrying real-time content (e.g. timestamp and control mechanisms for synchronizing different streams).

▪ **Real-Time Transport Control Protocol (RTCP)**

- The primary function is to provide information (statistics) to an application regarding the quality of data distribution. These statistics include number of packets sent, number of packets lost, inter-arrival jitter, etc.
- Identify RTP source: RTCP carries a transport-level identifier for an RTP source, called the canonical name (CNAME). Receivers use the CNAME to associate multiple data streams from a given participant in a set of related RTP sessions (e.g., to synchronize audio and video).
- Control RTCP transmission interval: To prevent control traffic from overwhelming network resources and to allow RTP to scale up to a large number of session participants.

▪ **Real Time Streaming Protocol (RTSP)**

- RTSP is a network control protocol designed to control streaming media servers.
- 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.
- Most RTSP servers use RTP for media stream delivery.
- Basic RTSP requests are as follows:

  (a) **OPTIONS:** Returns the request types the server will accept.

  (b) **DESCRIBE:** Returns the presentation description, typically in Session Description Protocol (SDP) format.

  (c) **SETUP:** Specifies how a single media stream must be transported. This must be done before a PLAY request is sent. The request contains the media stream URL and a transport specifier (which typically includes port numbers for receiving RTP and RTCP data).

  (d) **PLAY:** To play one or all media streams.

  (e) **PAUSE:** To temporarily halts one or all media streams.

  (f) **RECORD:** To send a stream to the server for storage.

  (g) **TEARDOWN:** To terminate the session.