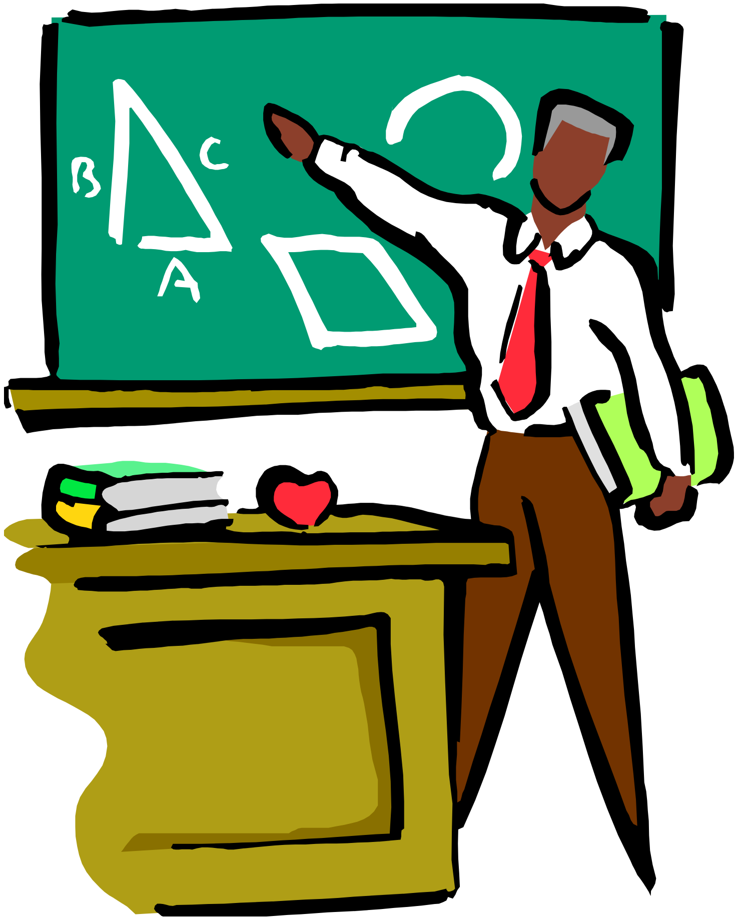
CONCORDIA UNIVERSITY
Concordia Institute for Information Systems Engineering

Chapter VI
Some Other Transport Protocols

Roch H. Glitho

The Other Transport Protocols



- 1 - Motivations and taxonomy
- 2 - Building on UDP: RTP / RTCP
- 3 - Building from scratch: SCTP
- 4 - Building from scratch: DCCP

Motivations and Taxonomy

Key characteristics of TCP

- Reliability
  - Three way handshake connection
  - Re-transmission
- Congestion control
  - Windows
    - Transmission rate reduction
- Uni-homing

Motivations and Taxonomy

Key characteristics of UDP

- No reliability
- No congestion control
- Uni-homing

Motivations and Taxonomy

The one size (either TCP or UDP) fits all philosophy does not always work

- What about
  - Applications requiring reliability but real time delivery (i.e. no retransmission)?
    - Interactive audio/video (e.g. conferencing)
  - Applications requiring more reliability than what is provided by TCP?
    - Multimedia session signalling
  - Applications requiring real time delivery, low reliability, but congestion control?
    - Multi party games

Motivations and Taxonomy

Two possible approaches

- Build a new transport protocol that complements / runs on top of existing transport protocols (e.g. UDP)
  - RTP/RTCP on top of UDP and application using RTP/RTCP
- Build a new transport protocol from scratch (i.e. runs on top of IP)
  - SCTP
  - DCCP

RTP / RTCP

Two complementary protocols

- Early 90s
- Primary goal: Real time media delivery with a focus on multimedia conferencing

Two complementary protocols

- Actual transportation of real time media
  Real-time Transport Protocol (RTP)
- Control of transportation:
  Real Time Transport Control Protocol (RTCP)

RTP / RTCP

Main characteristics

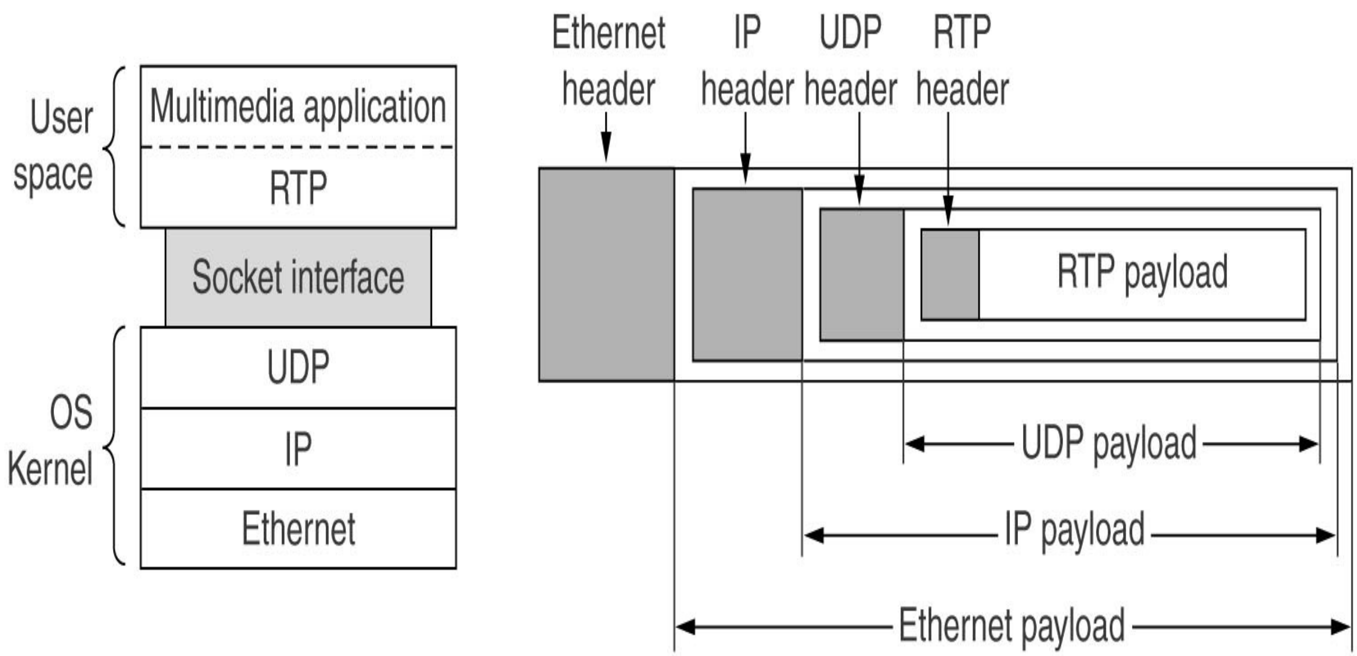
RTP:

- No provision for Quality of service
- No guarantee for out of sequence delivery
- Typically runs on top of UDP but may run on top of other protocols

RTCP:

- Help in providing control by providing information on packets sent, received
  - Information may be used by application to build whatever it thinks is necessary (e.g. reliability, congestion control)

RTP


User space
Multimedia application
RTP
Socket interface
OS Kernel
UDP
IP
Ethernet
Ethernet header
IP header
UDP header
RTP header
RTP payload
UDP payload
IP payload
Ethernet payload

(a) (b)

.

RTP

Mixers / translators

- Intermediate systems
- Connect 2 or more transport level clouds
  - End systems
  - Mixers / translators
- Use cases
  - Centralized conference bridges
  - Heterogeneous conferences
    - Low speed connection
    - High speed connection
    - Different encoding schemes
  - Some participants behind firewalls

RTP

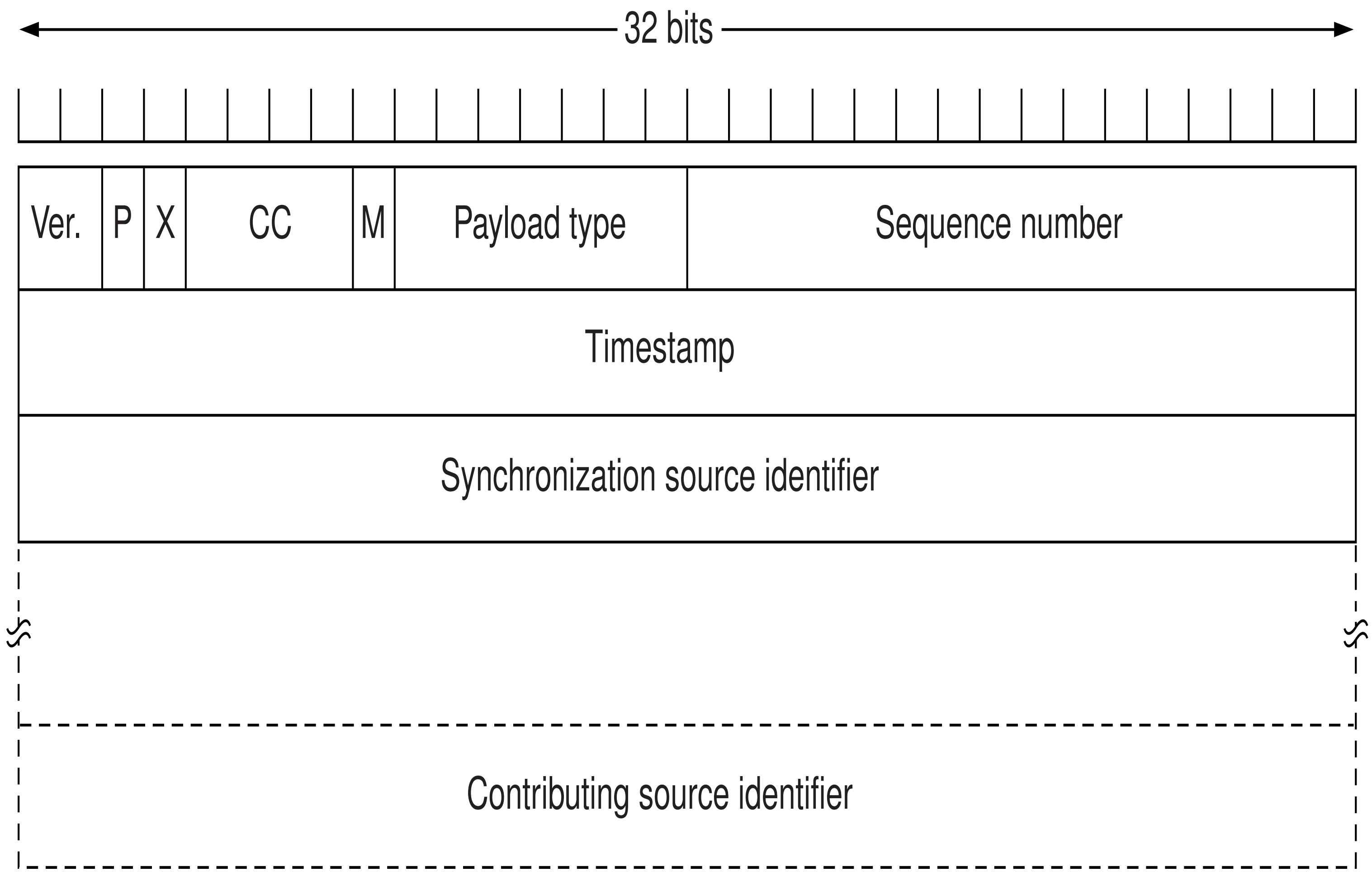
Synchronization source (SSRC)

- Grouping of data sources for playing back purpose (e.g. voice vs. video)
- An end system can act as several synchronization sources (e.g. IP phone with video capabilities)
- Translators forward RTP packets with their synchronization source intact

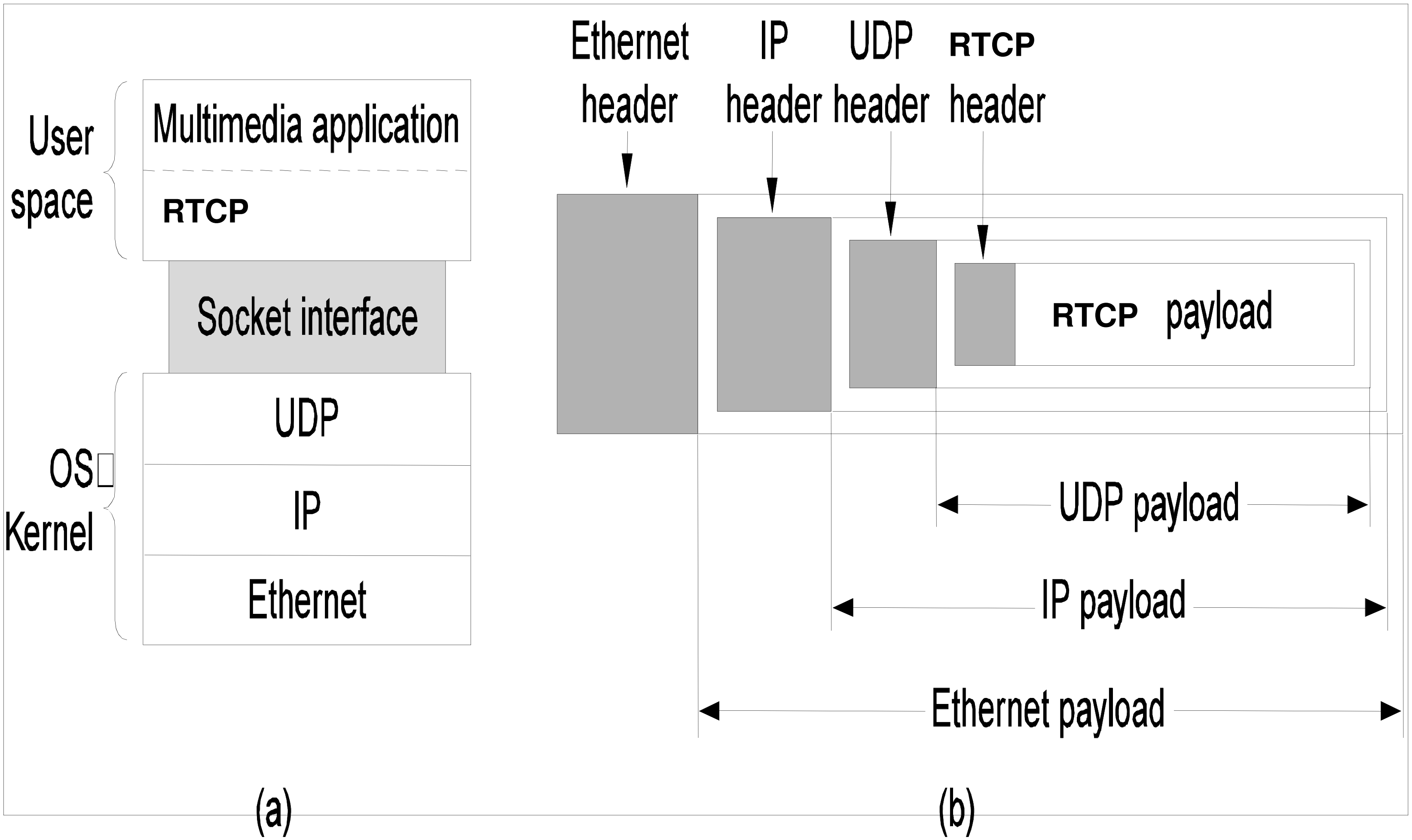
Contributing source (CSRC)

- A source of a stream of RTP packets that has contributed to the combined stream produced by an RTP mixer
- Mixers insert the list of contributing sources in the packets they generate

RTP


32 bits
Ver.
P
X
CC
M
Payload type
Sequence number
Timestamp
Synchronization source identifier
Contributing source identifier

RTCP


User space
Multimedia application
RTCP
Socket interface
OS Kernel
UDP
IP
Ethernet
Ethernet header
IP header
UDP header
RTCP header
RTCP payload
UDP payload
IP payload
Ethernet payload
(a)
(b)

RTCP concepts

Monitor:

- Application that receives RTCP packets sent by participants in an RTP session

Reports

- Reception quality feedback
- Sent by RTP packets receivers (which may also be senders)
  - May be used to build reliability, congestion control or whatever the application deems necessary

RTCP packets

Receiver report
Version
Time stamp
Sender's packet count
Reception report blocks

RTCP packets

```
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|V=2|P|    RC   |   PT=SR=200   |             length            |
header
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                         SSRC of sender                        |
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
|              NTP timestamp, most significant word             | sender
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+ info
|             NTP timestamp, least significant word             |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                         RTP timestamp                         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                     sender's packet count                     |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                      sender's octet count                     |
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
|                 SSRC\_1 (SSRC of first source)                | report
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+ block
| fraction lost |       cumulative number of packets lost       |   1
-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|           extended highest sequence number received           |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                      interarrival jitter                      |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                         last SR (LSR)                         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                   delay since last SR (DLSR)                  |
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
|                 SSRC\_2 (SSRC of second source)               | report
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+ block
:                               ...                             :   2
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
|                  profile-specific extensions                  |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

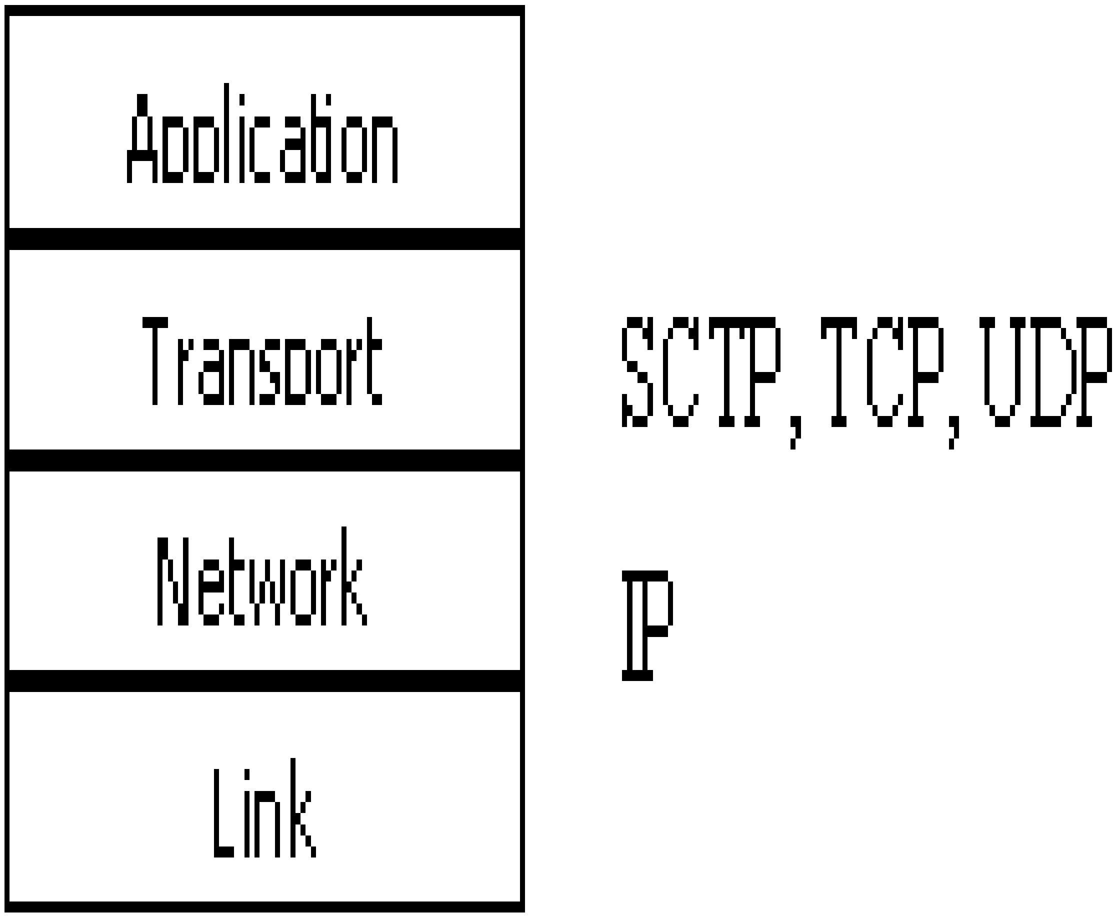
.

Stream Control Transmission Protocol (SCTP)

Designed in early 2000s to carry multimedia session signaling traffic over IP, then subsequently extended to meet the needs of a wider range of application

- Design goals much more stringent than TCP design goals (e.g. redundancy, higher reliability)
- Offer much more than TCP
- A sample of additional features
  - Four way handshake association instead of three way handshake connection
  - Multi-homing instead of uni-homing
  - Multi-streaming instead of uni-streaming

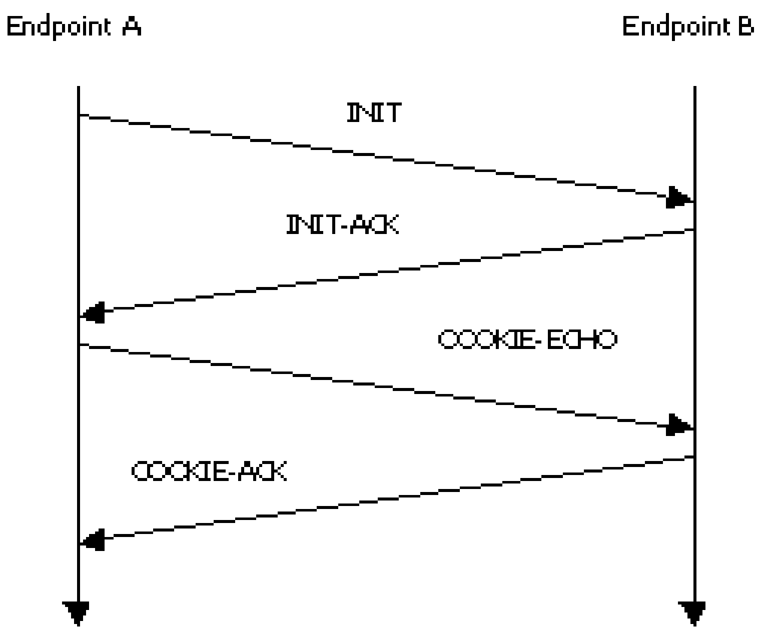
Stream Control Transmission Protocol



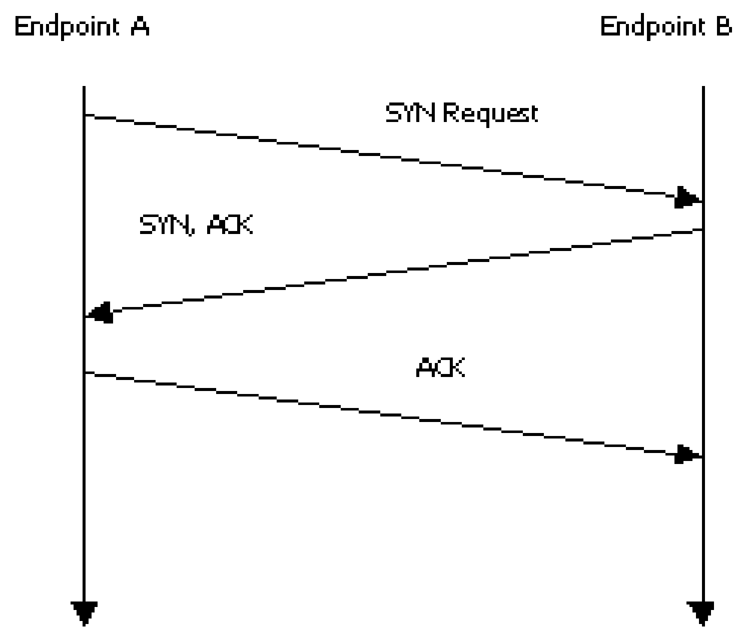
Four way handshake

Why?

- Key reason: Make SCTP resilient to denial of service (DOS) attacks, a feature missing in TCP



SCTP



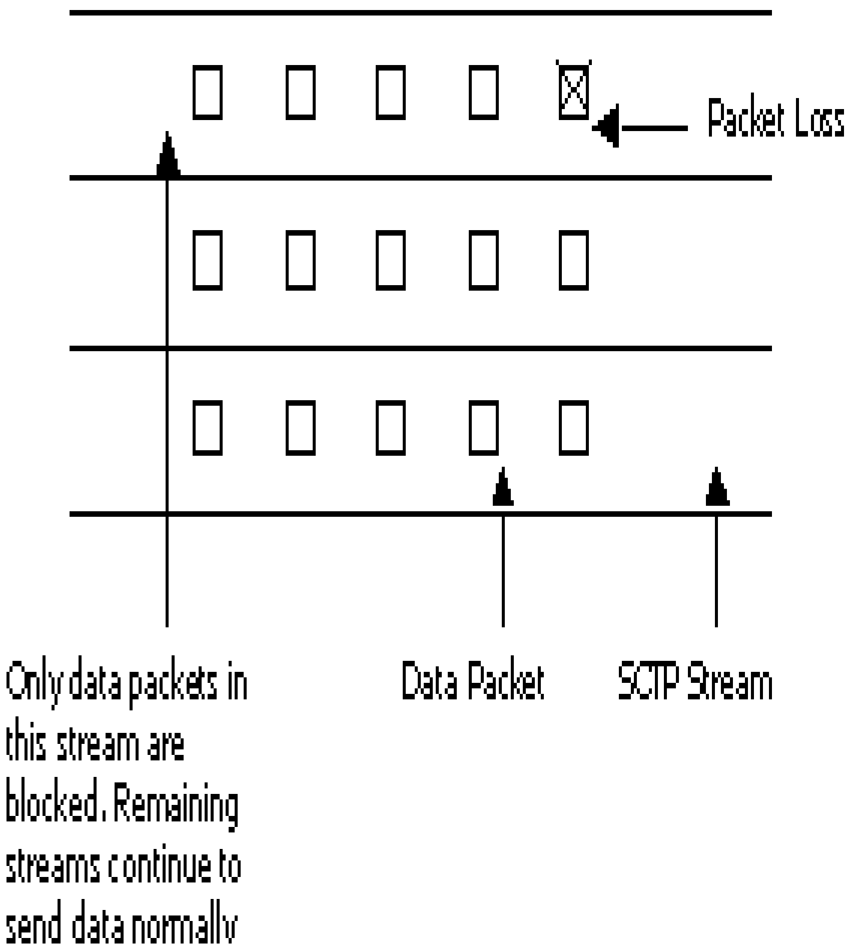
TCP

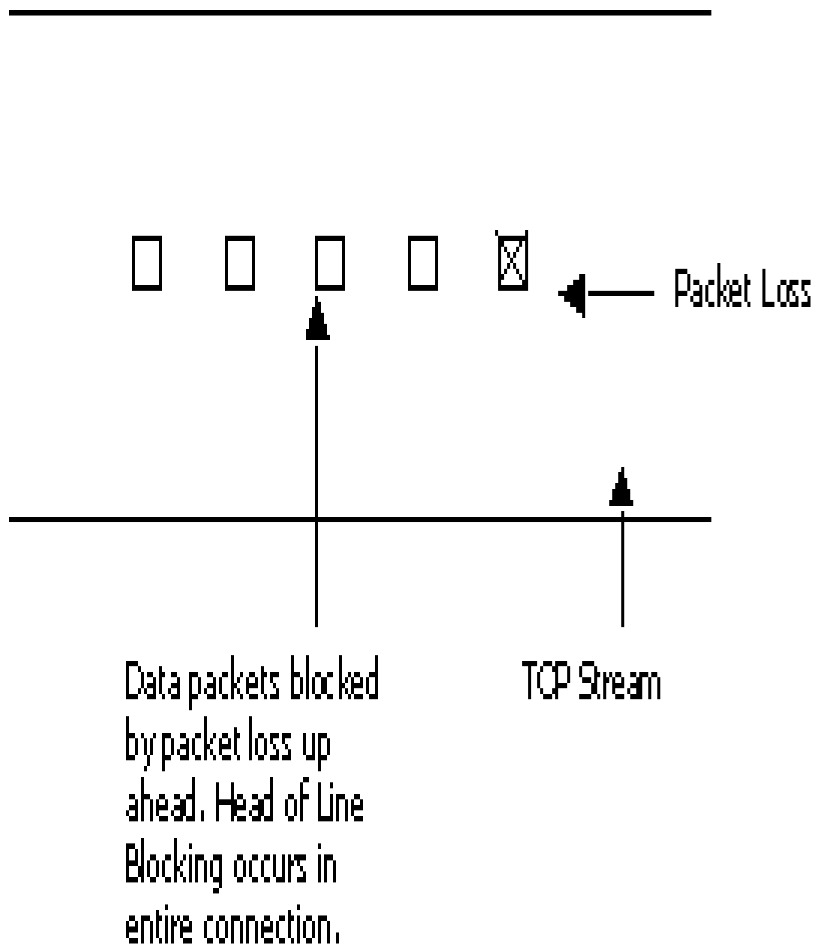
Multi-homing

Why?

- Key reason: Make SCTP resilient in resource failures, a feature missing in TCP
  - Multi-homed host: Host accessible via multiple IP addresses
  - Use cases
    - Subscription to multiple ISP to ensure service continuity when of the ISP fails
    - Mission critical systems relying on redundancy
    - Load balancing
  - Multi-homing with SCTP (only for redundancy)
    - Multi-homed host binds to several IP addresses during associations unlike TCP which binds to a single IP address
      - Retransmitted data is sent to an alternate IP address
      - Continued failure to reach primary address leads to the conclusion that primary address has failed and all traffic goes to alternate address

Multi-streaming





Data Congestion Control Protocol (DCCP)

One of the most recent transport protocols (Second half of the 2000s)

- Primary goal:
  - Delivery of real time media (somehow similar to the goal assigned to RTP / RTCP)
- Build on the experience acquired in protocol design / deployment since the design of RTP / RTCP (ie. Early 1990s)
  - Some examples of improvements:
    - Congestion control incorporated in the transport protocol (unlike RTP/RTCP)
    - Possibility to avoid DoS

Overall view

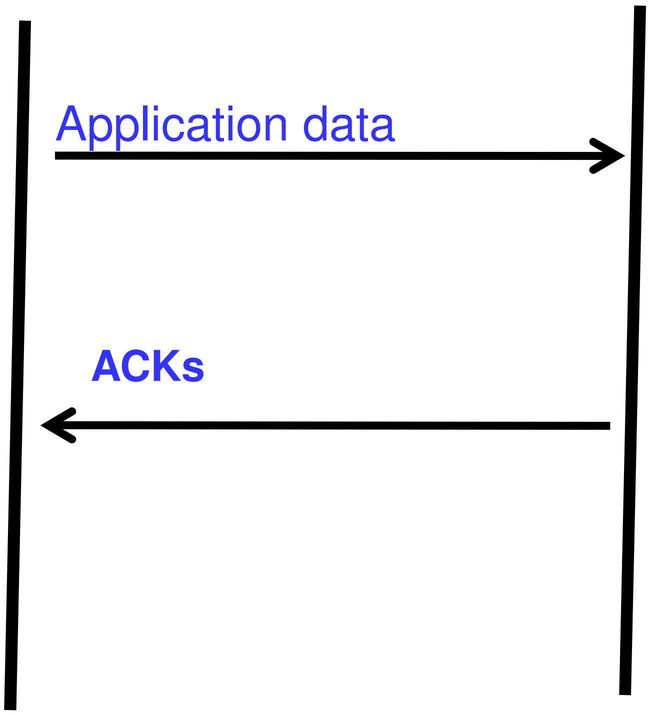
- Three way handshake connection like TCP
  - In-built possibility to use cookies during response phase to avoid DoS
  - A connection can be seen as two half-connections (i.e. uni-directional connections)
    - Possibility for a receiver to send only ACK
- Reliable connection establishment and feature negotiation
- Unreliable data transfer (no retransmission)
- Feature negotiation

The protocol states

```
Client                                      Server
    ------                                   ------
              (0) No connection
    CLOSED                                  LISTEN

              (1) Initiation
    REQUEST     DCCP-Request -->
                   <-- DCCP-Response    RESPOND
    PARTOPEN    DCCP-Ack or DCCP-DataAck -->

              (2) Data transfer
    OPEN        <-- DCCP-Data, Ack, DataAck -->     OPEN

              (3) Termination
                   <-- DCCP-CloseReq    CLOSEREQ
    CLOSING     DCCP-Close -->
                     <-- DCCP-Reset        CLOSED
    TIMEWAIT
    CLOSED
```

Half connection

Use case: Unidirectional streams (e.g. Streaming applications)



Data transfer

- Packets have sequence numbers
    - Client – server and server – client sequence numbers are independent
        - Tracking on both sides is possible
- Acknowledgements report last received packet
- Data drop option
    - Examples
        - Application not listening
        - Receiver buffer
        - Corrupt
    - May help in selecting congestion control mechanism

Data transfer

- Packets have sequence numbers
  - Client – server and server – client sequence numbers are independent
    - Tracking on both sides is possible
- Acknowledgements report last received packet
- Data drop option
  - Examples
    - Application not listening
    - Receiver buffer
    - Corrupt
  - May help in selecting congestion control mechanism

Feature negotiation

- Enable dynamic selection of congestion mechanism
  - Data drop option may help
    - Tracking on both sides is possible
  - TCP congestion control may be used
  - Other mechanisms may also be used

References

1, IETF RFC 3550, RTP / RTCP
2. A. Caro et al., SCTP: A Proposed Standard for Robust Internet Data Transport, IEEE Computer November 2003
3. S. Fu and M. Atiquzzaman, SCTP: State of the Art in Research, Products and Technical Challenges, IEEE Communications Magazine, April 2004
4. P. Natarajan et al., SCTP: What, Why and How? IEEE Internet Computing, September / October 2009
5. Y-C Lai, DCCP: Transport Protocol with Congestion Control and Unreliability, IEEE Internet Computing, September / October 2008