

# Data Center Networking: Challenges and Traditional Protocols

**(ENCS 691K – Chapter 6)** 

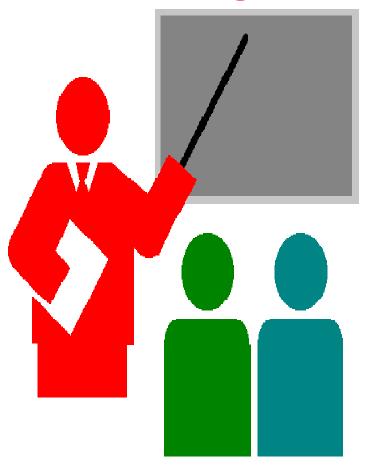
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# Data Center Networking: Challenges and Traditional Protocols



- Data Center Networking Challenges
- On Networking
- On Transport Layer
- Traditional Transport Protocols (Beyond TCP / UDP)
- Traditional Transport Protocols vs. Challenges









## References

- 1. K. Kant, Towards a Virtualized Data Center Transport Protocol, Infocom Workshop, 2008
- 2. M Alizadeh, Data Center TCP, ACM Sigcom 2011



Why is it necessary to re-think networking in cloud data center settings?

- Very high data rates (e.g. 100 Gb/sec Ethernet)
  - TCP can hardly cope with 10 GB/sec
    - New techniques are needed to make TCP cope, e.g.
      - Hardware acceleration
  - Need for QoS mechanisms
    - A single MAC pipe can carry data with different QoS requirements



Why is it necessary to re-think networking in cloud data center settings?

- Wide range of physical layer
  - Wired
  - Wireless
  - Optical
- Emerging PHY/MAC layers, e.g.
  - Ultra Wide Band
    - Huge amount of data over a short distance



Why is it necessary to re-think networking in cloud data center settings?

- Multiple level virtualization and cluster enabled applications
  - Real time applications / soft real time applications vs. other applications



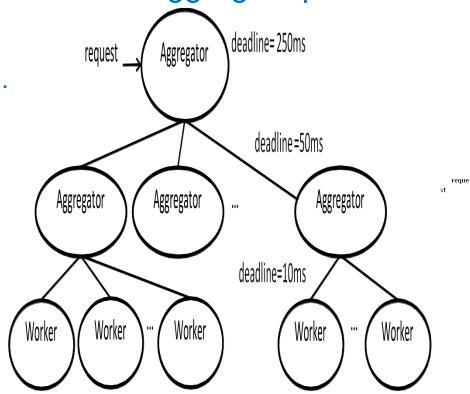
An illustration:

Soft real time applications, e.g.

- Web search
- Advertisement
- Retail



### Partition / Aggregate pattern





### An illustration:

### Examples of requirements:

- Low latency
- High burst tolerance

Important: Many other applications with conflicting requirements reside in the same data center



Let us focus on transport layer protocols requirements

- High data rate support (Up to 100 GB/s)
- User Level Protocol Indicator Support
- QoS friendly
- Virtual cluster support
- Data center flow / cong. Control
- High availability
- Compatibility with TCP/IP base
- Protection against DoS







# **On Networking**





## References

- 1. A. Tanenbaum, Computer Networks, Fourth Edition, Prentice Hall, 2003 (Introduction)
- 2. V. Strivasta and M. Montani, Cross Layer Design: A Survey and The Road Ahead, IEEE Communications Magazine, December 2005, Vol. 43, Issue 12, pp. 112 119



## **Layered Architectures**

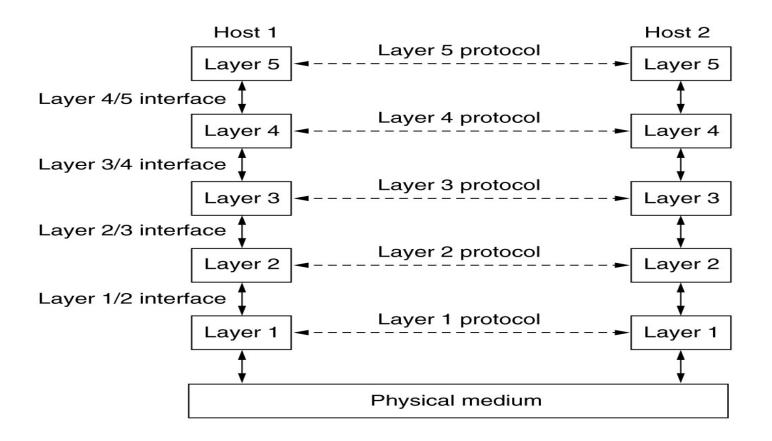


Figure 1.13 (Reference [1])



# **Layered Architectures**

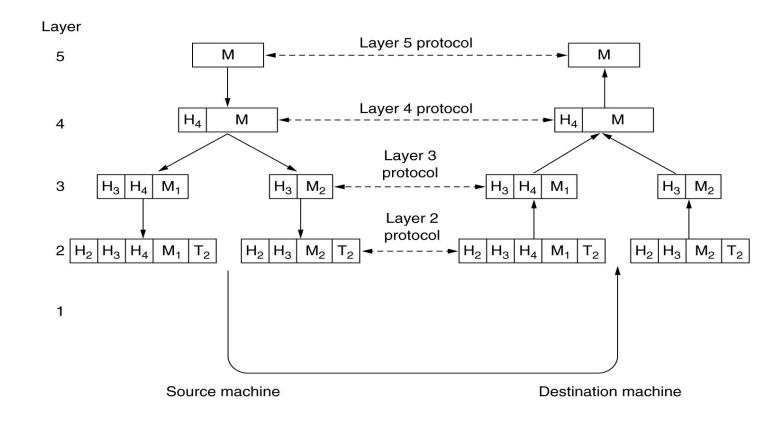


Figure 1.15 (Reference [1])



## **Cross Layered Architecture**

- Definition of cross layer design
  - Violation of the principles of layered protocol architectures
    - Examples
      - Allowing communications between non adjacent layers
      - Sharing variables between layers
      - Designing protocols that span several layers



## **Cross Layered Architectures**

### Main motivation for cross layer design

- Performance improvements, especially in wireless environments
  - An example
    - TCP sender assumes packet errors are indicators of networks congestion and slow down sending rates
      - Case of wired links: true
        - » Need to slow down



## **Cross Layered Architectures**

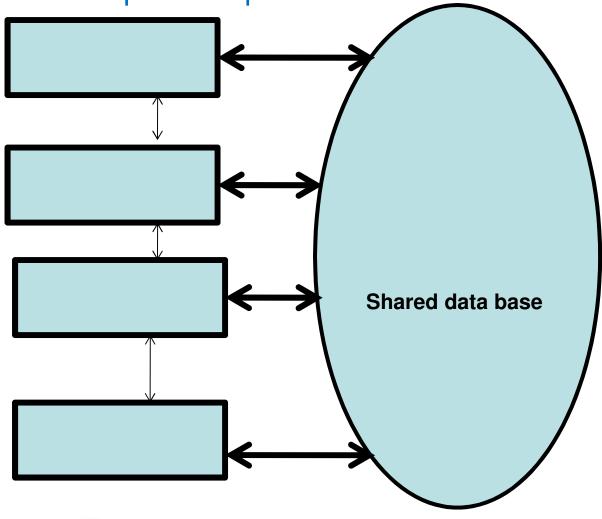
### Main motivation for cross layer design

- Performance improvements, especially in wireless environments
  - An example
    - Case of wireless links
      - » Not always true
      - » May be indicators of errors on physical and data link layers
      - » Information from physical and data link layers to transport layer (i.e. TCP) needed to make correct decision (i.e. slow down or speed up)



# **Cross Layered Architectures**

A shared data base example of implementation





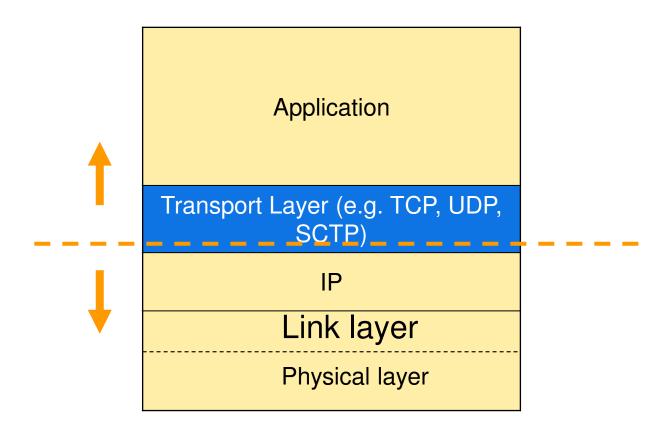


# **On Transport Layer**





# On The Transport Layer





## On The Transport Layer

- Provide service to application layer by using the service provided by network layer
- Hide physical network
  - Hide processing complexity
  - Hide different network technologies and architectures
- Provides host-to-host transport



## On The Transport Layer

- Addressing
- Connection Establishment
- Connection Release
- Flow Control
- Error Detection and Crash Recovery





# Traditional Transport Layers (Beyond TCP / UDP)



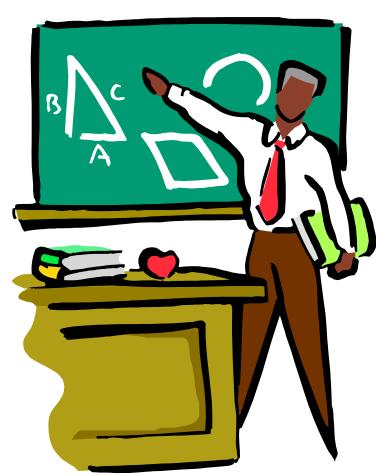


## References

- 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



## **The Other Transport Protocols**



1 - Motivations and taxonomy

2 - Building on UDP: RTP / RTCP

3 - Building from scratch: SCTP

4 - Building from scratch: DCCP



### Key characteristics of TCP

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



## Key characteristics of UDP

- No reliability
- No congestion control
- Uni-homing



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



### 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

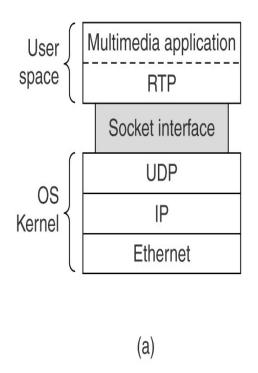
Typically runs on top of UDP but may run on top of other protocols

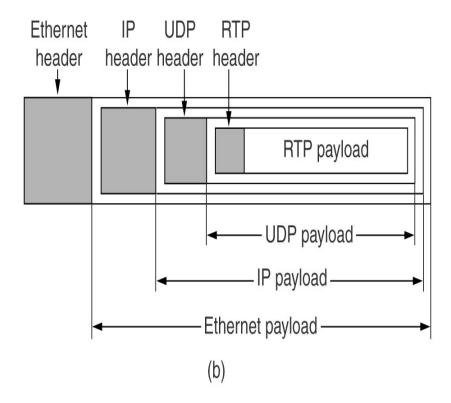
### RTCP:

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)









### 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



### **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)**

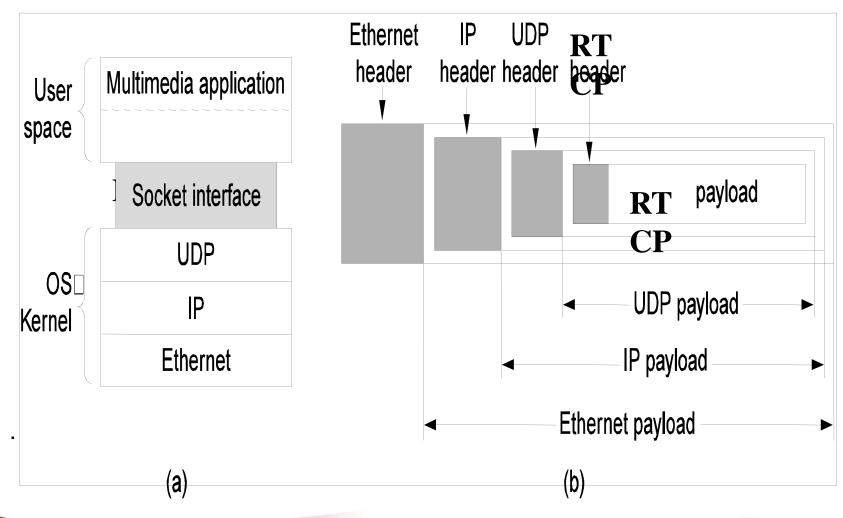
- A source of a stream of RTP packets that has contributed to the combined stream produced by an RTP mixer
- Mixers insert the list of contributing sources in the packets they generate



→ 32 bits —						
Ver.	Р	Χ	CC	М	Payload type	Sequence number
Timestamp						
Synchronization source identifier						
						ĕ
Contributing source identifier						



#### **RTCP**





### 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 |
header
SSRC of sender
NTP timestamp, most significant word | sender
  NTP timestamp, least significant word
RTP timestamp
     +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-
          sender's packet count
          sender's octet count
    | report
        SSRC\_1 (SSRC of first source)
| fraction lost | cumulative number of packets lost
    extended highest sequence number received
          interarrival jitter
 last SR (LSR)
  ------
         delay since last SR (DLSR)
SSRC\_2 (SSRC of second source)
                             report
                          -+-+-+ block
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 <u>Protocol</u>

Application

Transport

Network

Link

SCTP, TCP, UDP

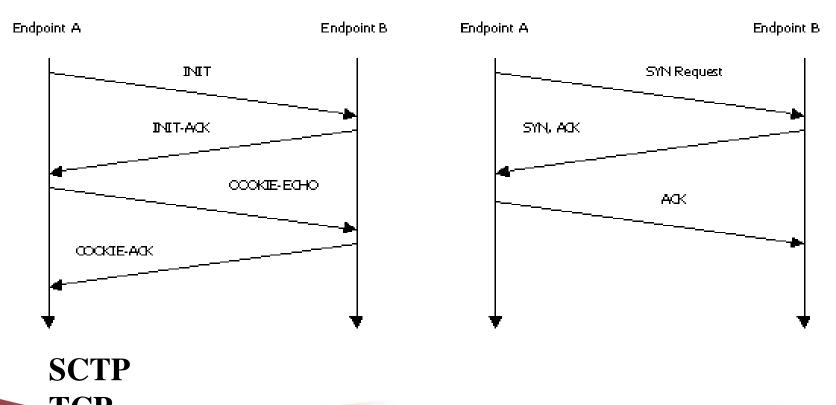
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## Four way handshake

#### Why?

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





### **Multi-homing**

#### Why?

- Key reason: Make SCTP resilient in resource failures, a feature missing in TCP (High availability)
  - 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



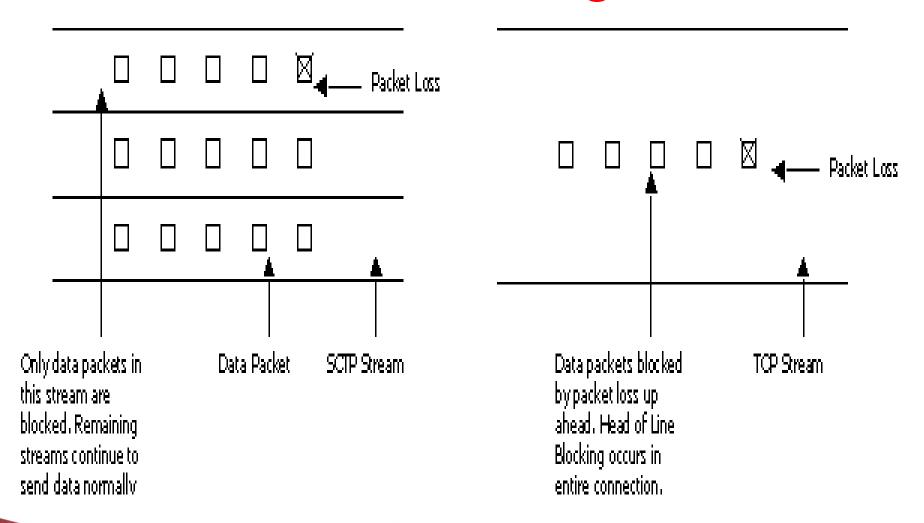
### **Multi-homing**

#### Why?

- Key reason: Make SCTP resilient in resource failures, a feature missing in TCP
  - 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. unidirectional connections)
    - Possibility for a receiver to send only ACK
- Reliable connection establishment and feature negotiation
- Unreliable data transfer (no retransmission)
- Feature negotiation



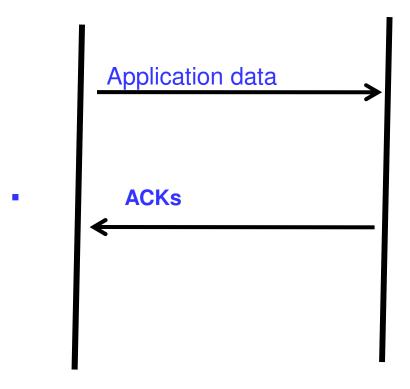
### The protocol states

Client Server (0) No connection **CLOSED** LISTEN (1) Initiation DCCP-Request --> **REQUEST** <-- DCCP-Response **RESPOND** DCCP-Ack or DCCP-DataAck --> **PARTOPEN** (2) Data transfer <-- DCCP-Data, Ack, DataAck --> **OPEN OPEN** (3) Termination <-- DCCP-CloseReq **CLOSEREQ** DCCP-Close --> **CLOSING** <-- 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
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- Acknowledgements report last received packet
- Data drop option
  - Examples
    - Application not listening
    - Receiver buffer
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  - 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





# Traditional Transport Protocols vs. Challenges





#### References

1. K. Kant, Towards a Virtualized Data Center Transport Protocol, Infocom Workshop, 2008



# Traditional Transport Protocols vs. Challenges (Ref. 1.)

Feature	TCP	SCTP	IBA
Scalability to 100 Gb/s	difficult	difficult	Easy?
Msg. based & ULP support	No	Yes	Yes
QoS friendly transport?	No	No	Yes
Virtual cluster support	No	No	limited
DC centric flow/cong. control	No	No	limited
Power aware transmission	Limited	limited	No
High availability features	Poor	Fair	Fair
Compatible w/ TCP/IP base	Yes	Yes	No
Protection against DoS attacks	Poor	Good	No



## The End





