Computer Networks(2015 Pattern) Unit V - Transport Layer

By Prof. A.R. Jain

PVG's COE, Nashik

Note: Material for this presentations are taken from Internet and books and only being used for student reference

Services,		
Addressing		
Berkley Sockets,		
Multiplexing,		
ТСР		
Connection establishment,		
Connection release,		
Flow control and buffering,		
TCPTimer management,		
TCP Congestion Control,		
Real Time Transport protocol(RTP),		
Stream Control Transmission Protocol (SCTP),		
Quality of Service (QoS),		
Differentiated services,		
TCP and UDP for Wireless.		

Transport Layer Services

Process to Process delivery

Connection less as well as connection oriented data delivery

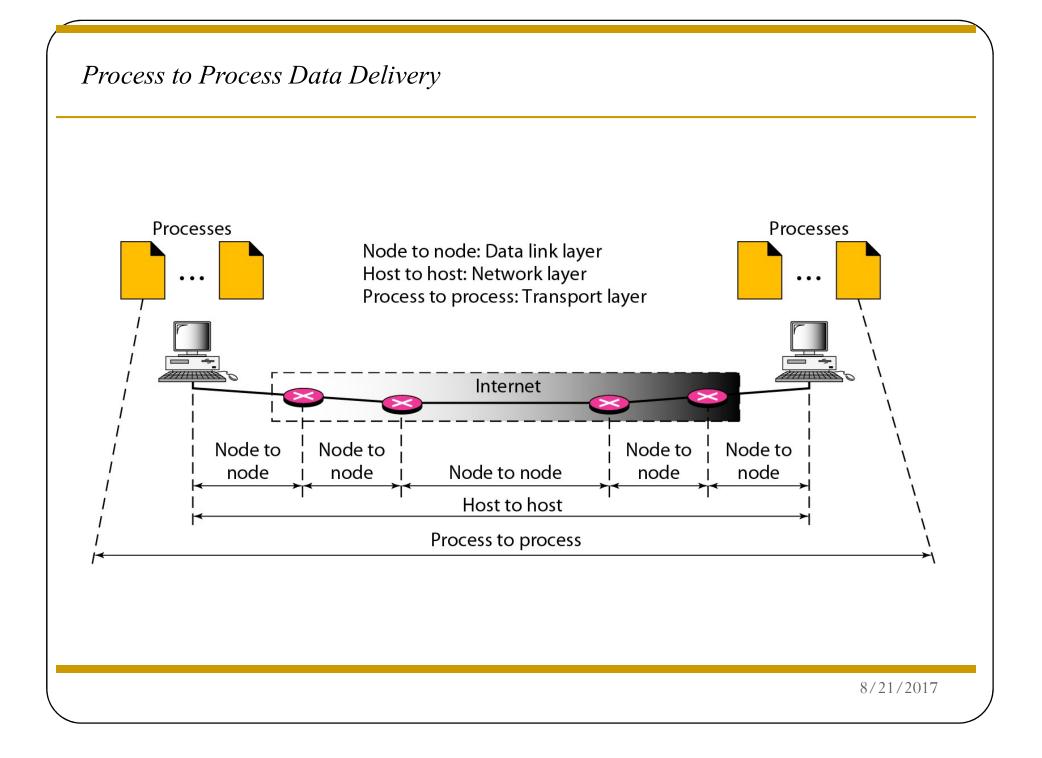
Error control

multiplexing/demultiplexing

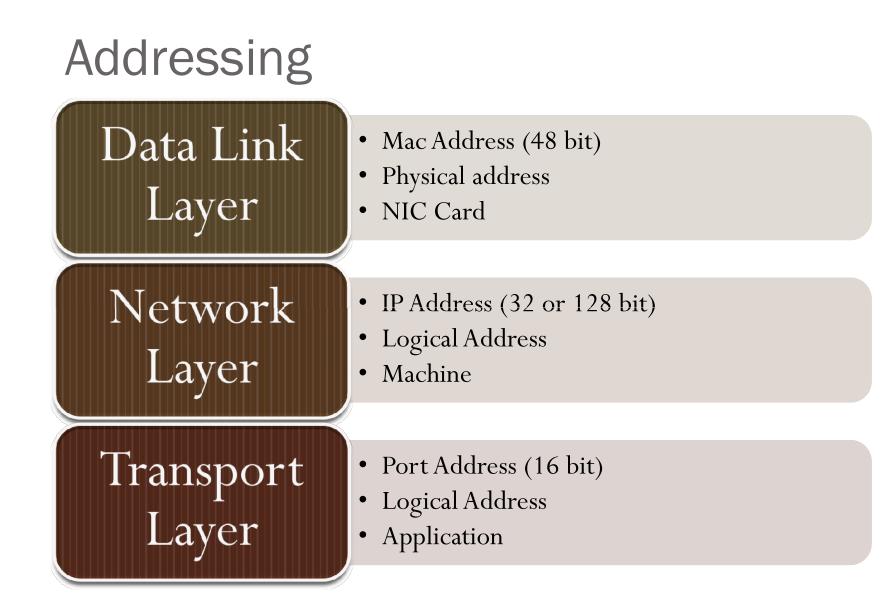
reliable data transfer

flow control

congestion control



Services,		
Addressing		
Berkley Sockets,		
Multiplexing,		
ТСР		
Connection establishment,		
Connection release,		
Flow control and buffering,		
TCPTimer management,		
TCP Congestion Control,		
Real Time Transport protocol(RTP),		
Stream Control Transmission Protocol (SCTP),		
Quality of Service (QoS),		
Differentiated services,		
TCP and UDP for Wireless.		



Transport services and protocols

provide *logical communication* between app processes running on different hosts

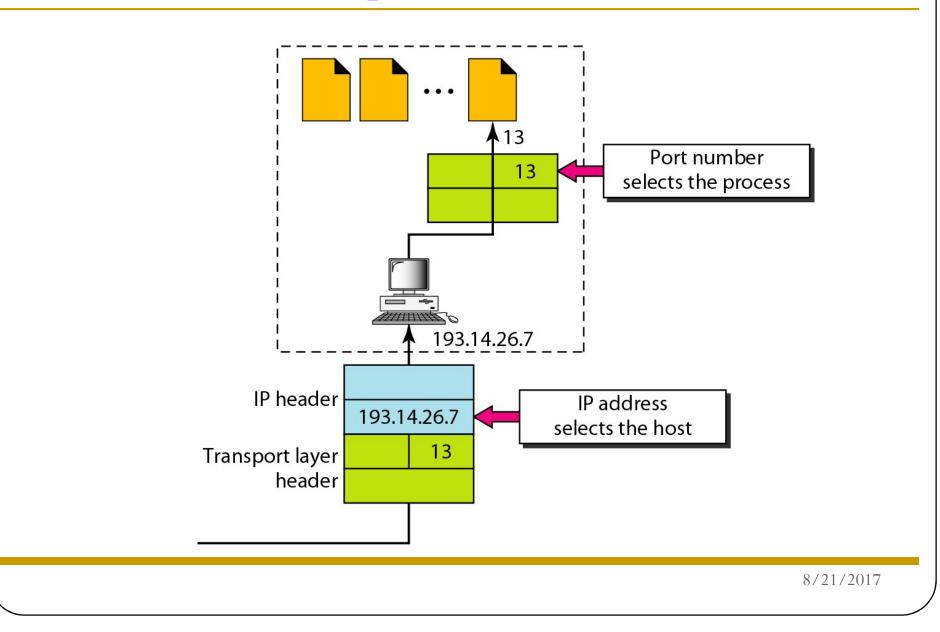
transport protocols run in end systems

- send side: breaks app messages into segments, passes to network layer
- rcv side: reassembles segments into messages, passes to app layer

more than one transport protocol available to apps

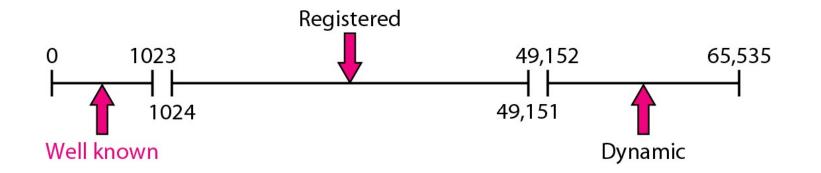
• Internet: TCP and UDP and SCTP

IP addresses versus port numbers

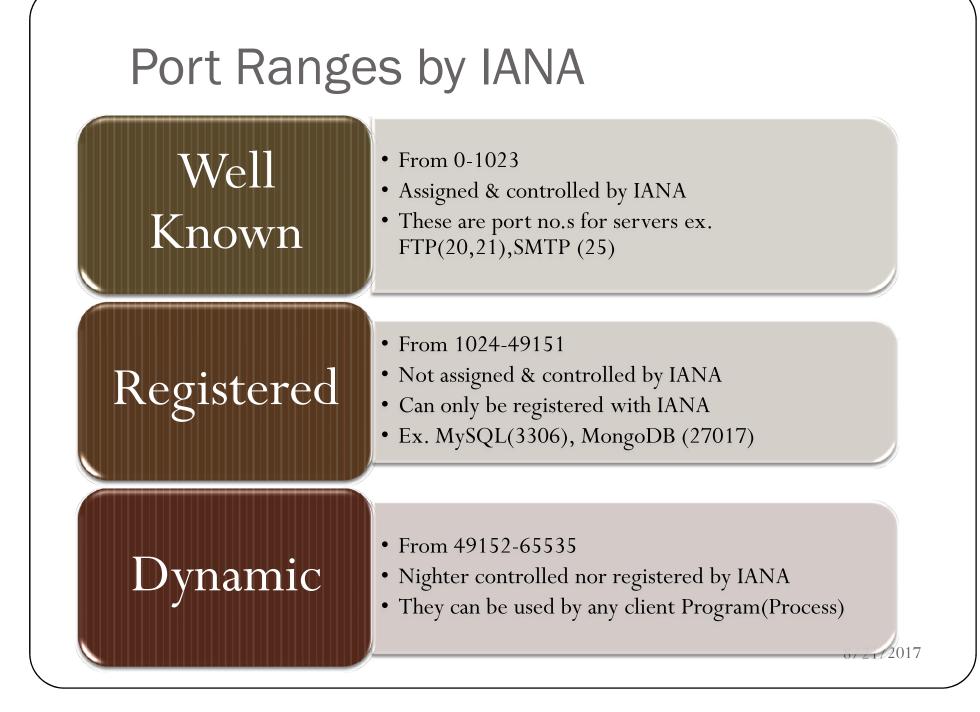


PORT ranges by IANA (Internet Assigned

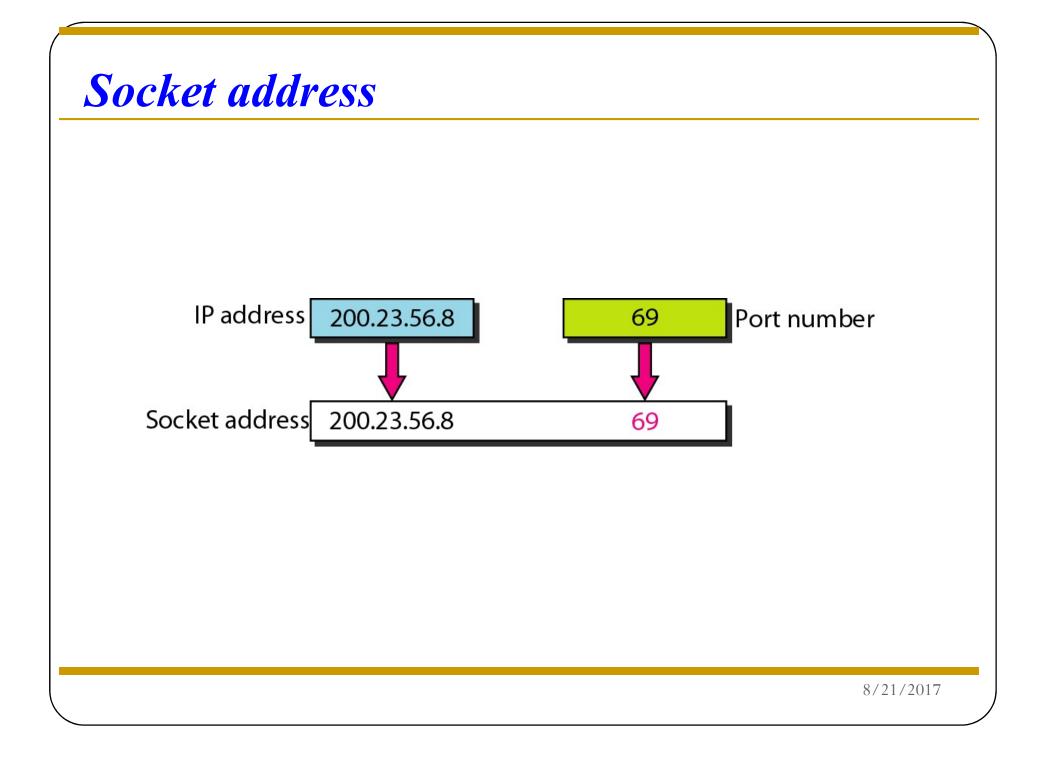
Number Authority)







Services,		
Addressing		
Berkley Sockets,		
Multiplexing,		
ТСР		
Connection establishment,		
Connection release,		
Flow control and buffering,		
TCPTimer management,		
TCP Congestion Control,		
Real Time Transport protocol(RTP),		
Stream Control Transmission Protocol (SCTP),		
Quality of Service (QoS),		
Differentiated services,		
TCP and UDP for Wireless.		

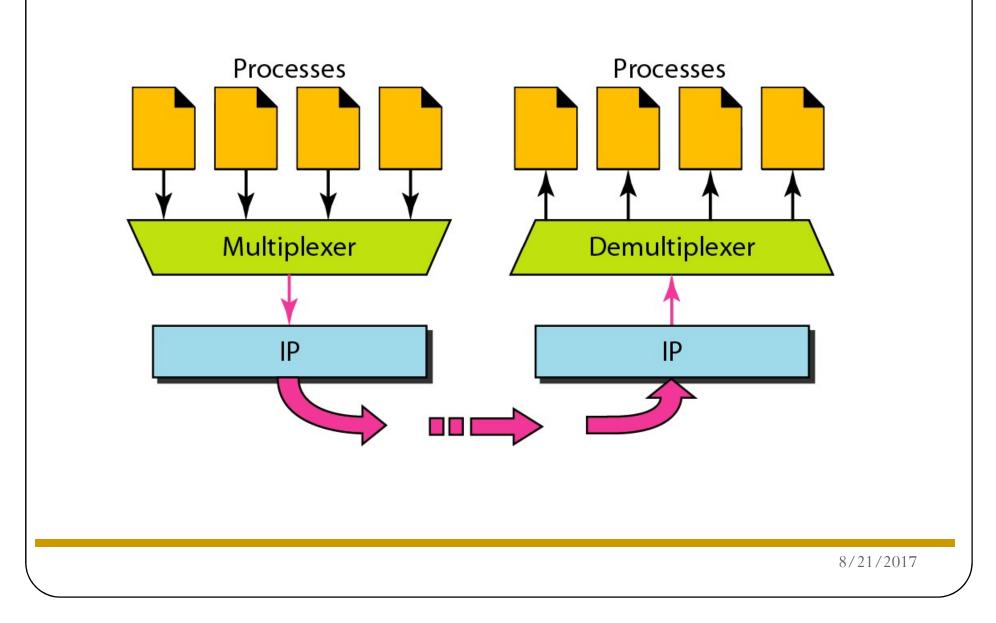


Transport Service Primitives Berkeley Sockets

Primitive	Meaning	
SOCKET	Create a new communication end point	
BIND	Attach a local address to a socket	
LISTEN	Announce willingness to accept connections; give queue size	
ACCEPT	Block the caller until a connection attempt arrives	
CONNECT	Actively attempt to establish a connection	
SEND	Send some data over the connection	
RECEIVE	Receive some data from the connection	
CLOSE	Release the connection	

Services,
Addressing
Berkley Sockets,
Multiplexing,
ТСР
Connection establishment,
Connection release,
Flow control and buffering,
TCPTimer management,
TCP Congestion Control,
Real Time Transport protocol(RTP),
Stream Control Transmission Protocol (SCTP),
Quality of Service (QoS),
Differentiated services,
TCP and UDP for Wireless.

Multiplexing and demultiplexing



Multiplexing/demultiplexing

•

<u>Multiplexing at</u> <u>send host:</u>

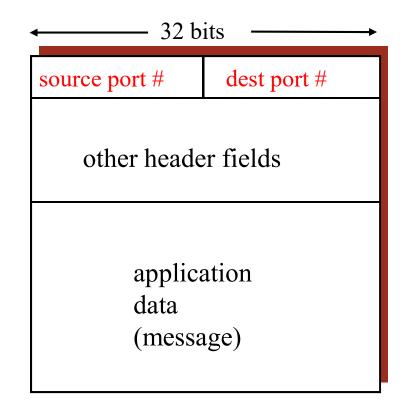
Gathering data from multiple sockets, enveloping data with header (later used for demultiplexing

Demultiplexing at rcv host:

• Delivering received segments to correct socket

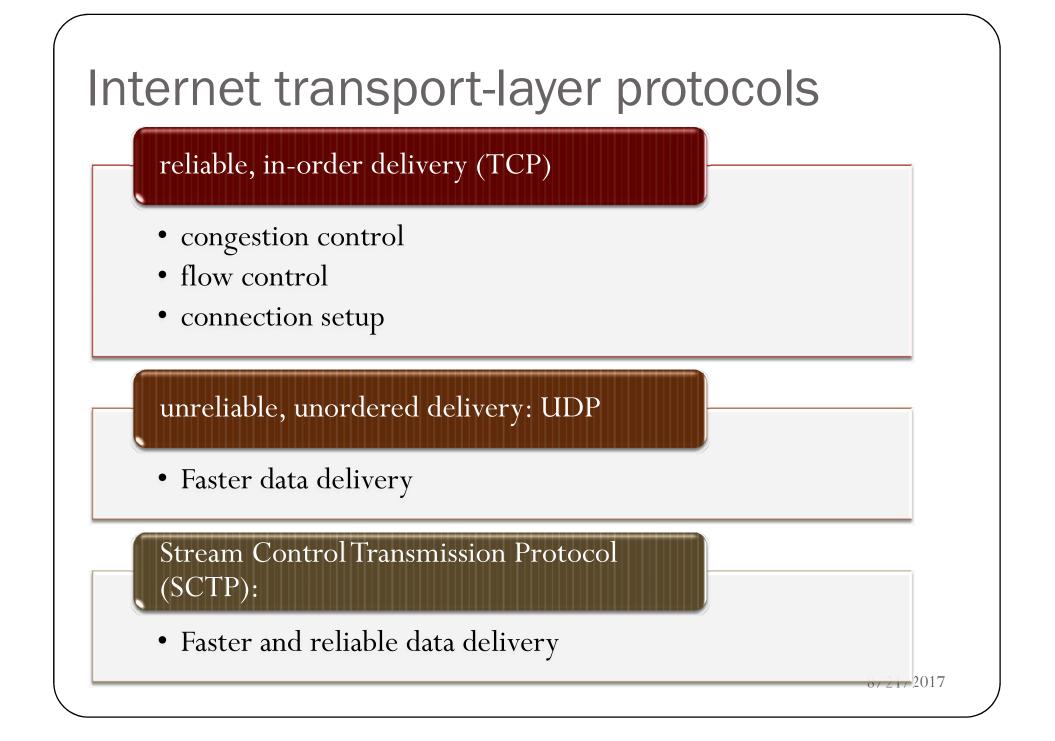
How demultiplexing works

- host receives IP datagrams
 - each datagram has source IP address, destination IP address
 - each datagram carries 1 transport-layer segment
 - each segment has source, destination port number
- host uses IP addresses & port numbers to direct segment to appropriate socket



TCP/UDP segment format

Services,
Addressing
Berkley Sockets,
Multiplexing,
ТСР
Connection establishment,
Connection release,
Flow control and buffering,
TCPTimer management,
TCP Congestion Control,
Real Time Transport protocol(RTP),
Stream Control Transmission Protocol (SCTP),
Quality of Service (QoS),
Differentiated services,
TCP and UDP for Wireless.

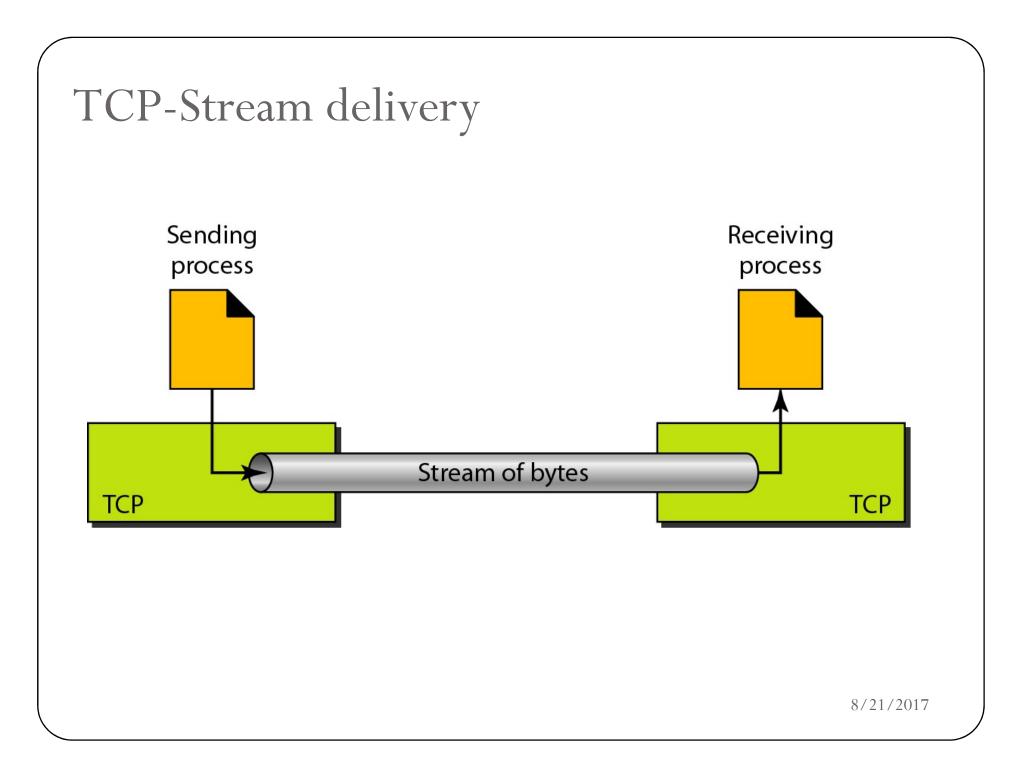


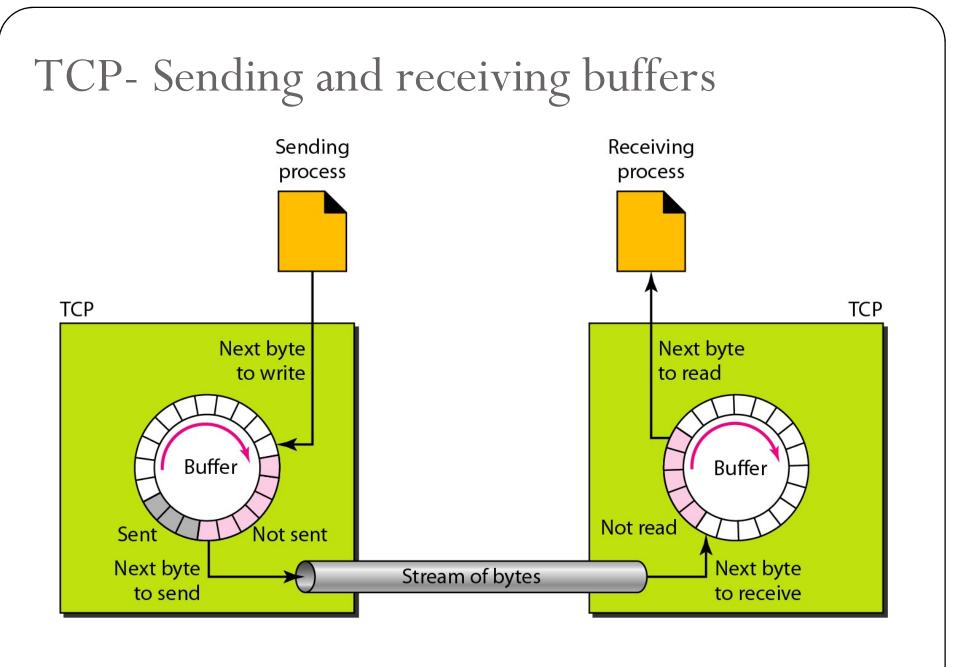
TCP (Transmission control protocol)

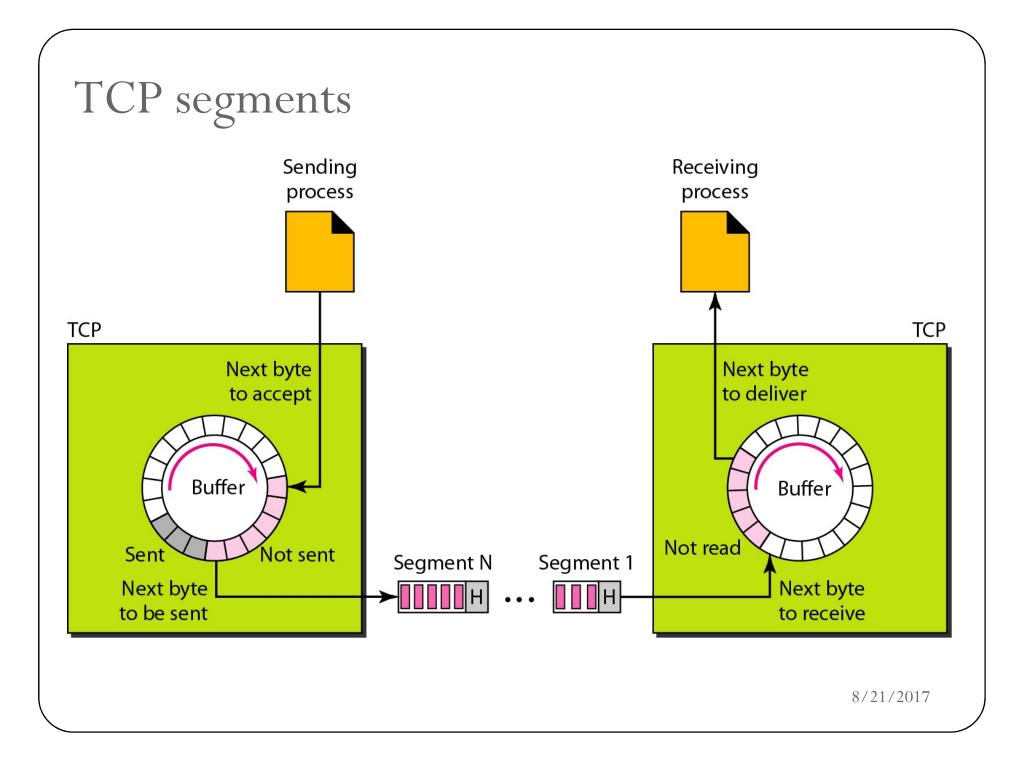
TCP is a connection-oriented protocol; it creates a virtual connection between two TCPs to send data.
 In addition, TCP uses flow and error control mechanisms at the transport level.

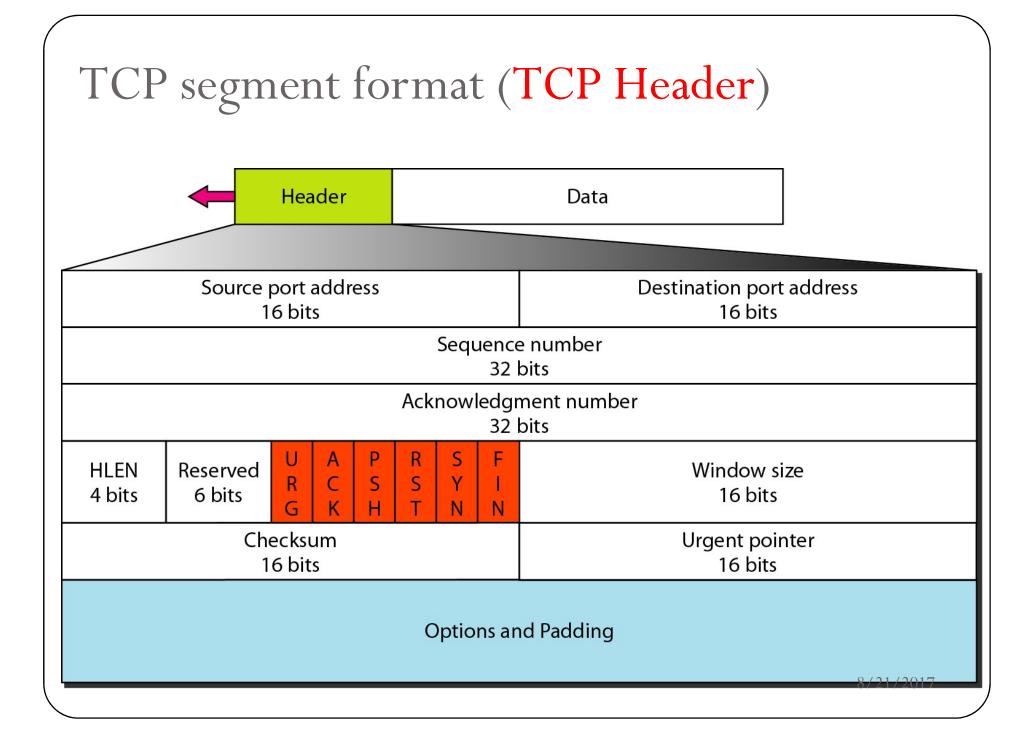
The TCP Service Model-Some assigned ports.

Port	Protocol	Use	
21	FTP	File transfer	
23	Telnet	Remote login	
25	SMTP	E-mail	
69	TFTP	Trivial File Transfer Protocol	
79	Finger	Lookup info about a user	
80	HTTP	World Wide Web	
110	POP-3	Remote e-mail access	
119	NNTP	USENET news	





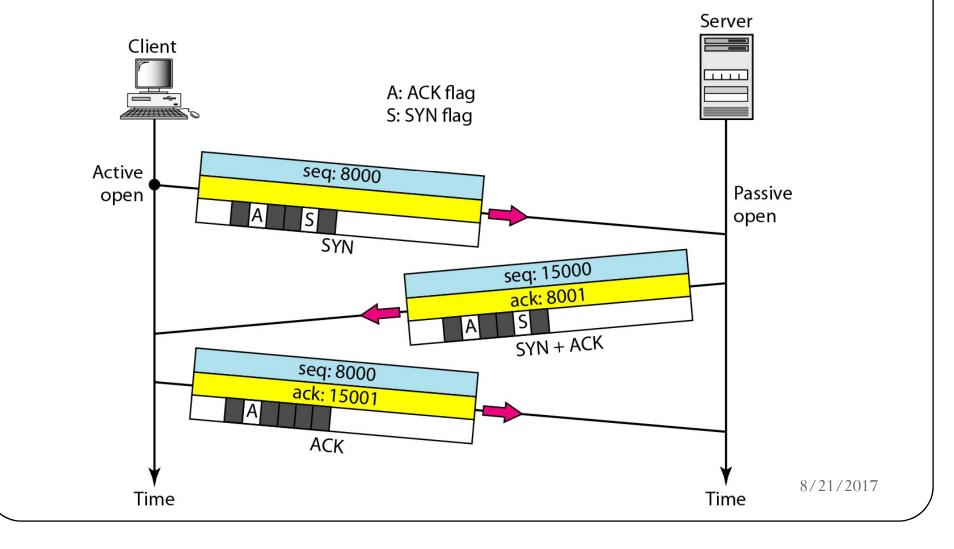




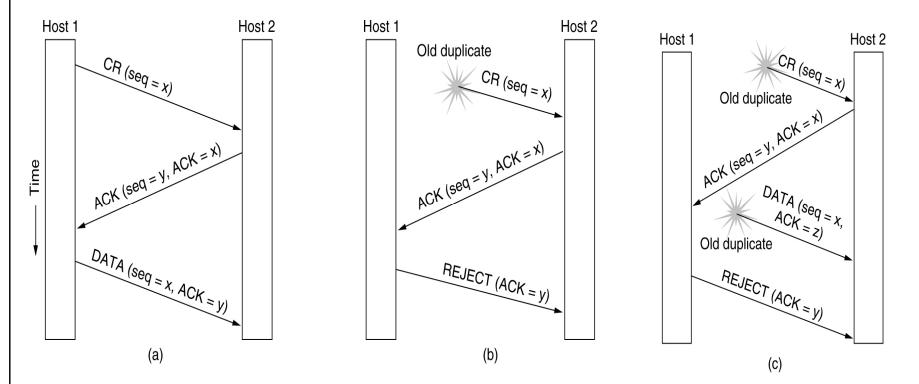
	TCP -C	Control	field			
URG: Urgent pointer is valid ACK: Acknowledgment is valid PSH: Request for push		valid	RST: Reset the connection SYN: Synchronize sequence numbers FIN: Terminate the connection			
	URG	ACK	PSH	RST	SYN	FIN
L						
						8/21/2017

Services,
Addressing
Berkley Sockets,
Multiplexing,
ТСР
Connection establishment,
Connection release,
Flow control and buffering,
TCPTimer management,
TCP Congestion Control,
Real Time Transport protocol(RTP),
Stream Control Transmission Protocol (SCTP),
Quality of Service (QoS),
Differentiated services,
TCP and UDP for Wireless.

TCP Connection establishment using three-way handshaking



Connection Establishment -3 Scenarios



Three protocol scenarios for establishing a connection using a three-way handshake. CR denotes CONNECTION REQUEST.

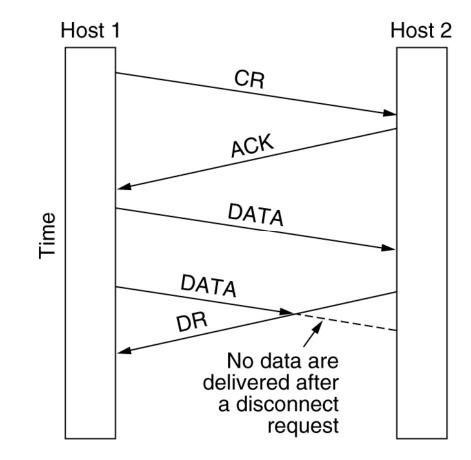
(a) Normal operation,

(b) Old CONNECTION REQUEST appearing out of nowhere.

(c) Duplicate CONNECTION REQUEST and duplicate ACK.

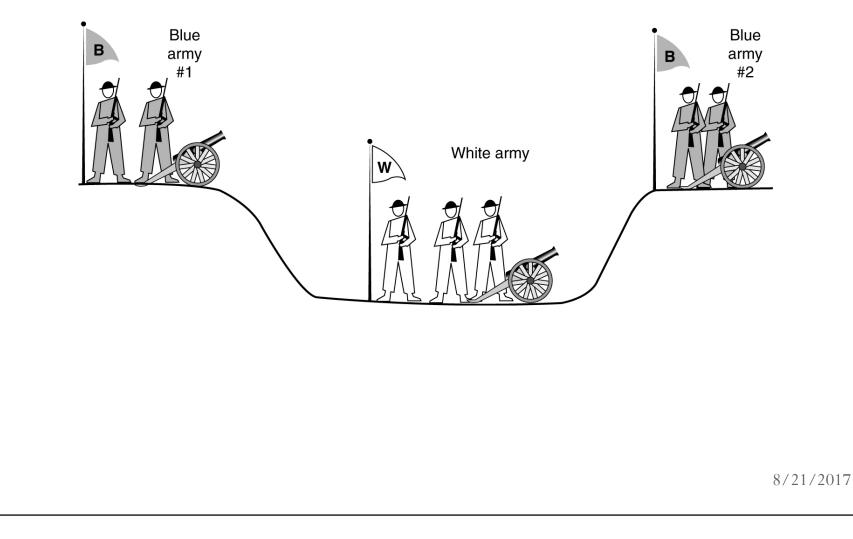
Services,
Addressing
Berkley Sockets,
Multiplexing,
ТСР
Connection establishment,
Connection release,
Flow control and buffering,
TCPTimer management,
TCP Congestion Control,
Real Time Transport protocol(RTP),
Stream Control Transmission Protocol (SCTP),
Quality of Service (QoS),
Differentiated services,
TCP and UDP for Wireless.

TCP- Connection Release

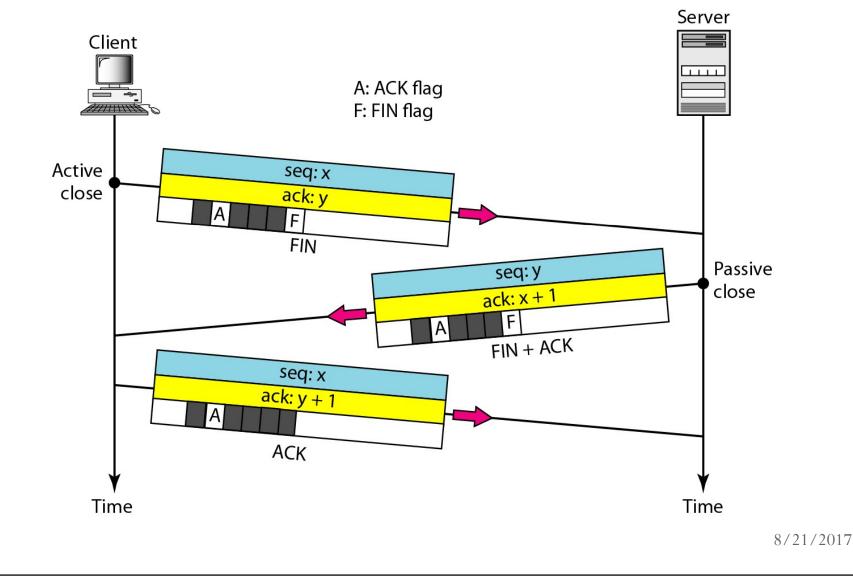


Abrupt disconnection with loss of data.

TCP- Connection Release-The two-army problem.

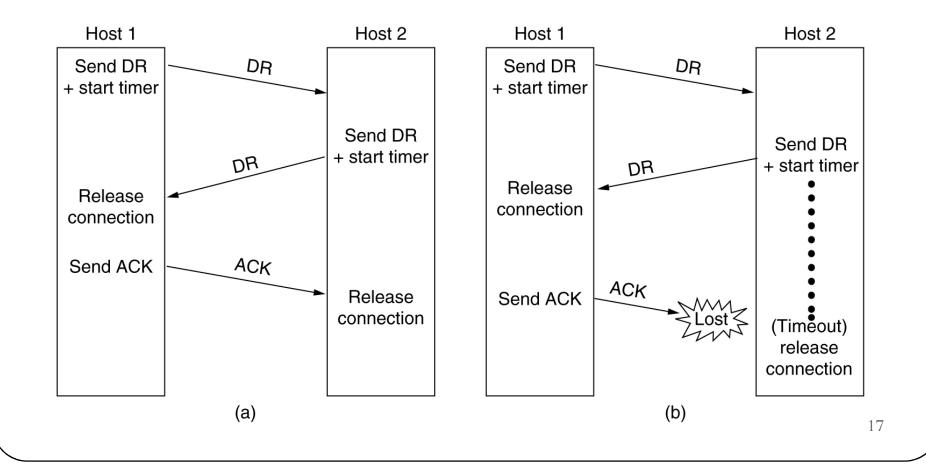


TCP Connection termination using three-way handshaking



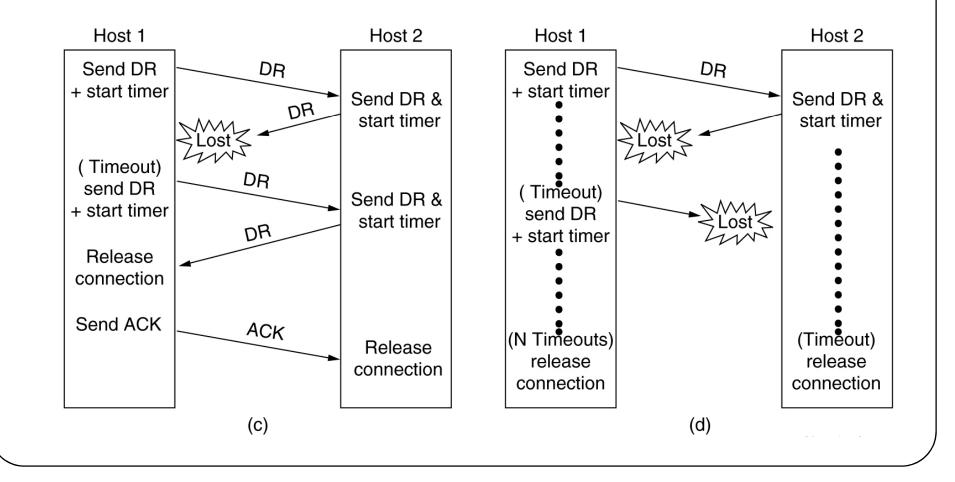
Connection Release Scenarios

Four protocol scenarios for releasing a connection. (a) Normal case of a three-way handshake. (b) final ACK lost.



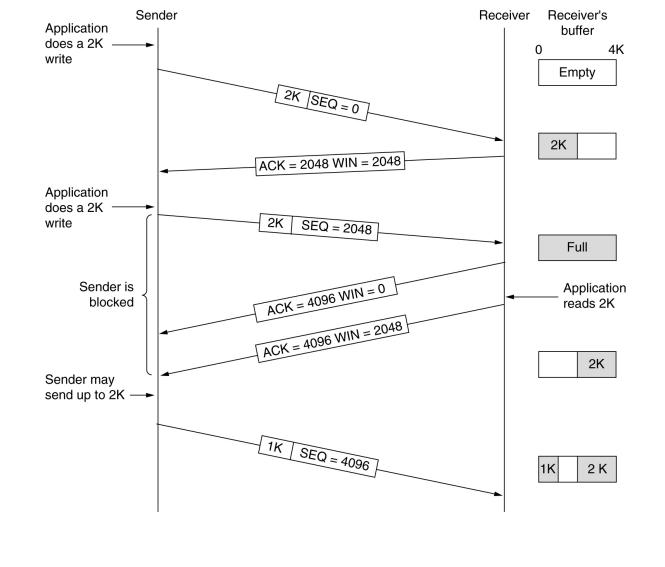
Connection Release Scenarios...

(c) Response lost. (d) Response lost and subsequent DRs lost.

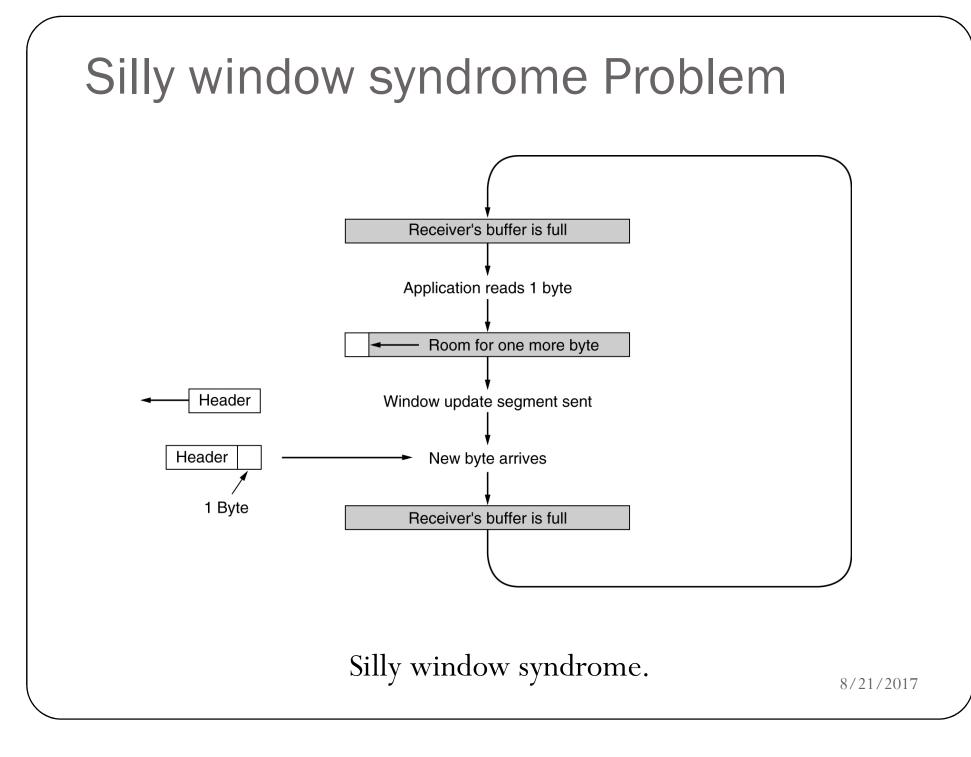


Services,
Addressing
Berkley Sockets,
Multiplexing,
ТСР
Connection establishment,
Connection release,
Flow control and buffering,
TCPTimer management,
TCP Congestion Control,
Real Time Transport protocol(RTP),
Stream Control Transmission Protocol (SCTP),
Quality of Service (QoS),
Differentiated services,
TCP and UDP for Wireless.

TCP Transmission Policy(Flow control)



Window management in TCP.



Solution to Silly window syndrome Problem

There are two solutions

Nagle's solutionClark's solution

Nagle's algorithm

Purpose is to allow the **sender** TCP to make efficient use of the network, while still being responsive to the sender applications.

Idea:

If application data comes in byte by byte, send first byte only. Then *buffer all application data till until ACK for first byte comes in*.

If network is slow and application is fast, the second segment will contain a lot of data.

Send second segment and buffer all data till ACK for second segment comes in.

An exception to this rule is to always send (not wait for ACK) if enough data for half the receiver window or MSS(Maximum segment size) is accumulated.

Clark's algorithm

Purpose is to prevent the **receiver** from sending a window update for 1byte.

Idea:

Receiver is forced to wait until it has a decent amount of space available

The receiver should not send a window update until it can handle the maximum segment size it declared when the connection was established or until its buffer is half empty, whichever is smaller

Outline

Services,
Addressing
Berkley Sockets,
Multiplexing,
ТСР
Connection establishment,
Connection release,
Flow control and buffering,
TCPTimer management,
TCP Congestion Control,
Real Time Transport protocol(RTP),
Stream Control Transmission Protocol (SCTP),
Quality of Service (QoS),
Differentiated services,
TCP and UDP for Wireless.

TCP timers- TCP uses 4 timers to do its work

The most important of these is the RTO (Retransmission TimeOut). When a segment is sent, a retransmission timer is started. If the segment is acknowledged before the timer expires, the timer is stopped. If, on the other hand, the timer goes off before the acknowledgement comes in, the segment is retransmitted (and the timer os started again).

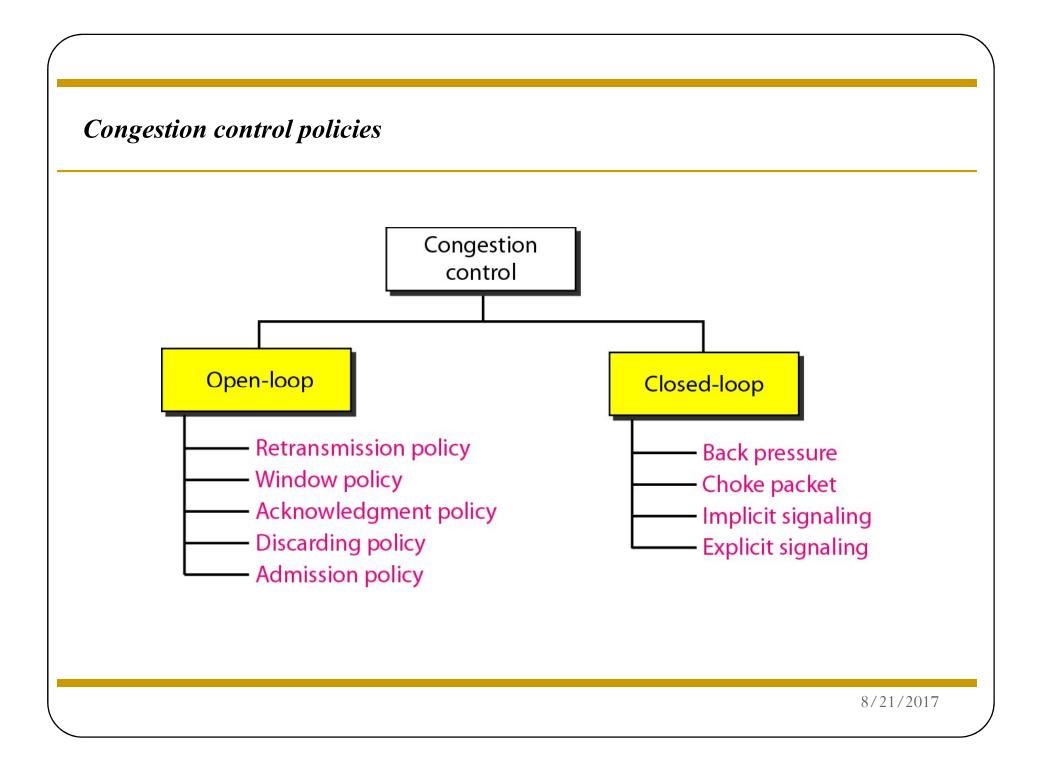
Persistence timer- It is designed to prevent the following deadlock.

A third timer that some implementations use is the keepalive timer. When a connection has been idle for a long time, the keepalive timer may go off to cause one side to check whether the other side is still there.

The last timer used on each TCP connection is the one used in the TIME WAIT state while closing. It runs for twice the maximum packet lifetime to make sure that when a connection is closed; all packets created by it have died off.

Outline

Services,
Addressing
Berkley Sockets,
Multiplexing,
ТСР
Connection establishment,
Connection release,
Flow control and buffering,
TCPTimer management,
TCP Congestion Control,
Real Time Transport protocol(RTP),
Stream Control Transmission Protocol (SCTP),
Quality of Service (QoS),
Differentiated services,
TCP and UDP for Wireless.



TCP congestion control

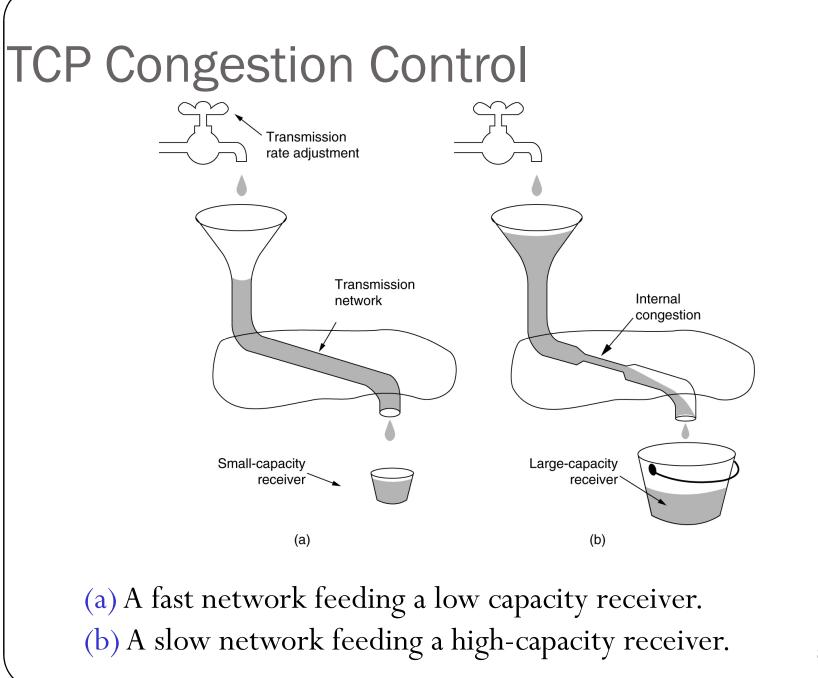
We looked at how TCP handles flow control. In addition we know the congestion happens. The only real way to handle congestion is for the sender to reduce sending rate.

So how does on detect congestion ?

In old days, packets were lost due to transmission errors and congestion. But nowadays, transmission errors are very rare (except for wireless). So, TCP assumes a lost packet as an indicator of congestion.

So does TCP deal with congestion ?

It maintains an indicator of network capacity, called the *congestion window*



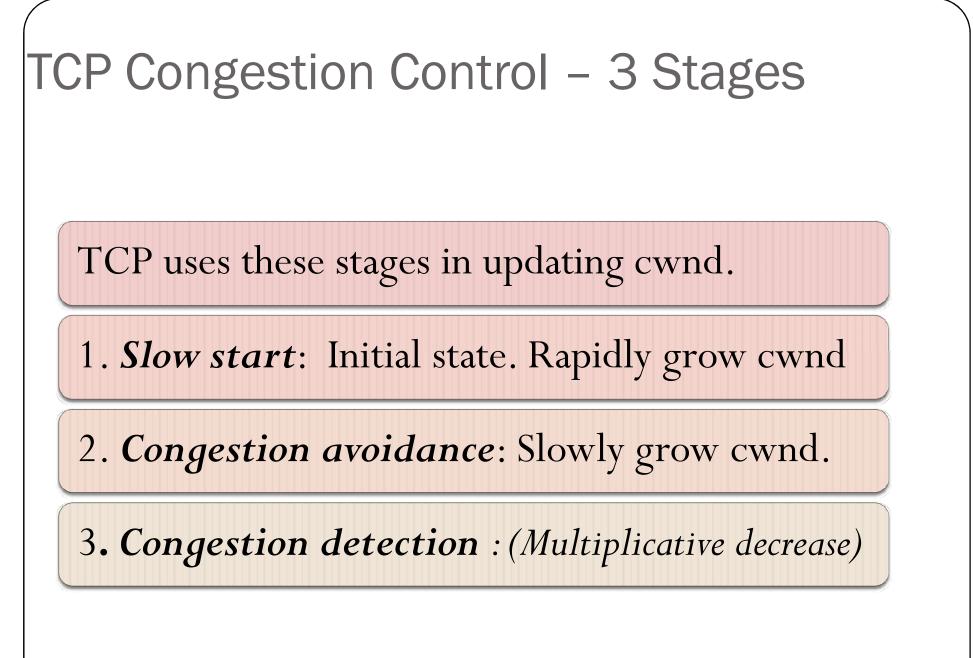
TCP congestion control

In essence TCP deals with two potential problems separately:

Problem **Receiver capacity** \longrightarrow **Receiver window** (rwnd)

Solution **Network capacity** — **Congestion window** (cwnd)

Each window reflect the number of bytes the sender may transmit. The sender sends the minimum of these two sizes. This size is the *effective window*.



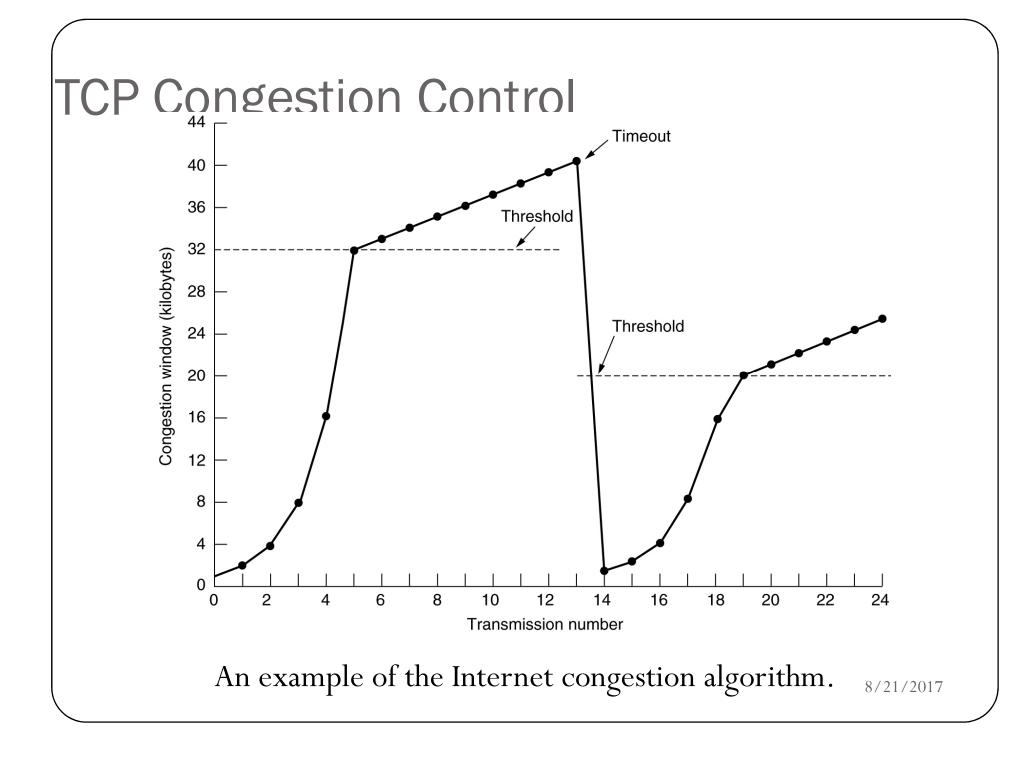
TCP Congestion Control – Slow start

When connection is established , the sender initializes the congestion window to the size of the maximum segment in use on the connection.

It then sends the one maximum segment

If this segment is acknowledged before timeout occurs then it doubles the segment size

This is continued until the timeout occurs or receivers window size is reached



TCP Congestion Control-Congestion Avoidance

When the size of congestion window reaches the slow start threshold, the slow start phase stops and the additive phase begins.

TCP Congestion Control-Congestion Detection

If congestion occurs the congestion window size must be decreased.

That means when a timer time outs or when 3 Acks are received the size of the threshold is dropped to ½ (multiplicative decrease)

Outline

Services,

Addressing

Berkley Sockets,

Multiplexing,

TCP

Connection establishment,

Connection release,

Flow control and buffering,

TCPTimer management,

TCP Congestion Control,

UDP – (Gap Analysis) content beyond syllabus

Real Time Transport protocol(RTP),

Stream Control Transmission Protocol (SCTP),

Quality of Service (QoS),

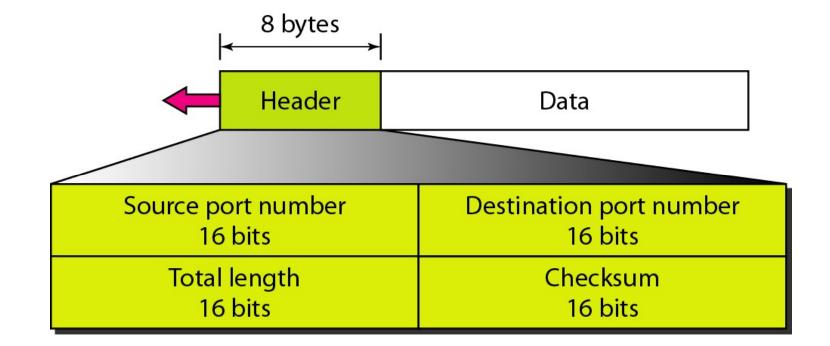
Differentiated services,

TCP and UDP for Wireless.

USER DATAGRAM PROTOCOL (UDP)

• The User Datagram Protocol (UDP) is called a connectionless, unreliable transport protocol. It does not add anything to the services of IP except to provide process-to-process communication instead of host-to-host communication.





UDP Pseudo Header

0	4	8	12	<mark>1</mark> 6	20	24	28	32
	Source Address (from IP Header) Destination Address (from IP Header)							
	Reserved	served Protocol (from IP Header)				Length om UDP Head	der)	

Figure : UDP Pseudo Header Format

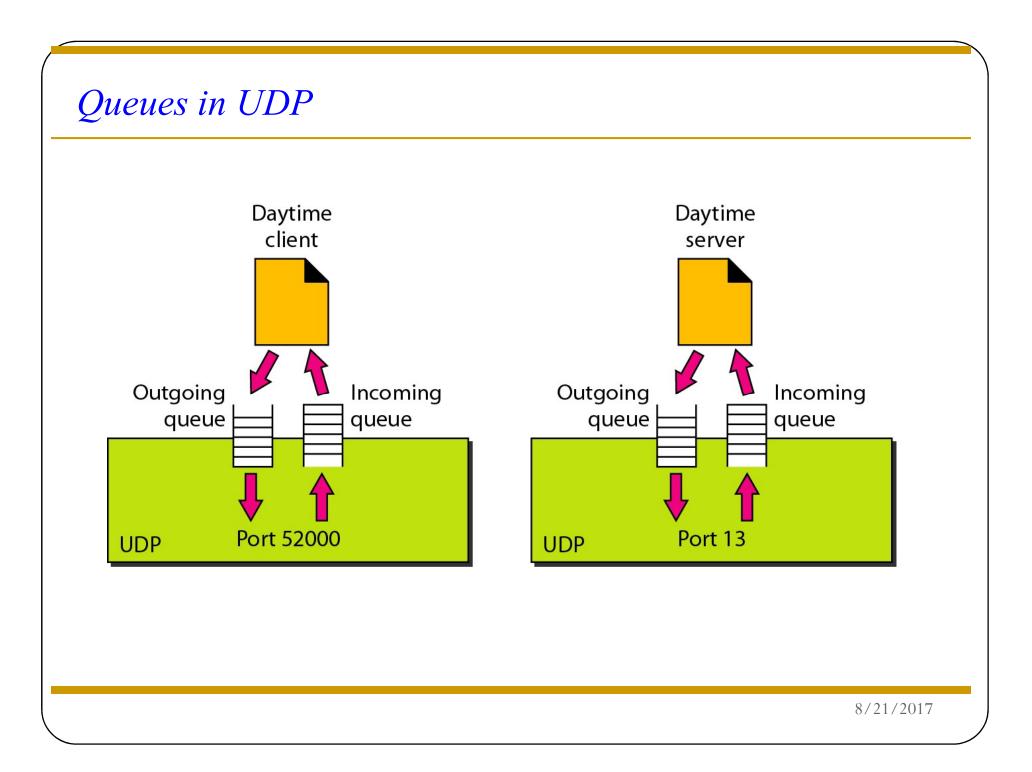
UDP Operations

Connectionless service

No Flow and error control except checksum

Encapsulation and Decapsulation of messages in IP datagram

Queing



Uses of UDP

Simple Request reply communication

Suitable for process with internal flow and control mechanisms. Eg. TFTP

The Real-Time Transport Protocol

Used in route updating protocol like Routing Information Protocol(RIP)

Remote Procedure Call(RPC)

Suitable for Multicasting. Multicasting capability is inbuilt in UDP software's

Outline

Services,

Addressing

Berkley Sockets,

Multiplexing,

TCP

Connection establishment,

Connection release,

Flow control and buffering,

TCPTimer management,

TCP Congestion Control,

UDP – (Gap Analysis) content beyond syllabus

RealTimeTransport protocol(RTP),

Stream Control Transmission Protocol (SCTP),

Quality of Service (QoS),

Differentiated services,

TCP and UDP for Wireless.

RTP: A Transport Protocol for Real-Time Applications

- Internet standard for real-time data
 - Interactive audio, video, and simulation data
- Primarily designed for multi-user multimedia conference
 - Session management
 - Scalability considerations
- Provides end-to-end transport functions for real-time applications
 - Payload type identification
 - Sequence numbering
 - Timestamping
 - Delivery monitoring

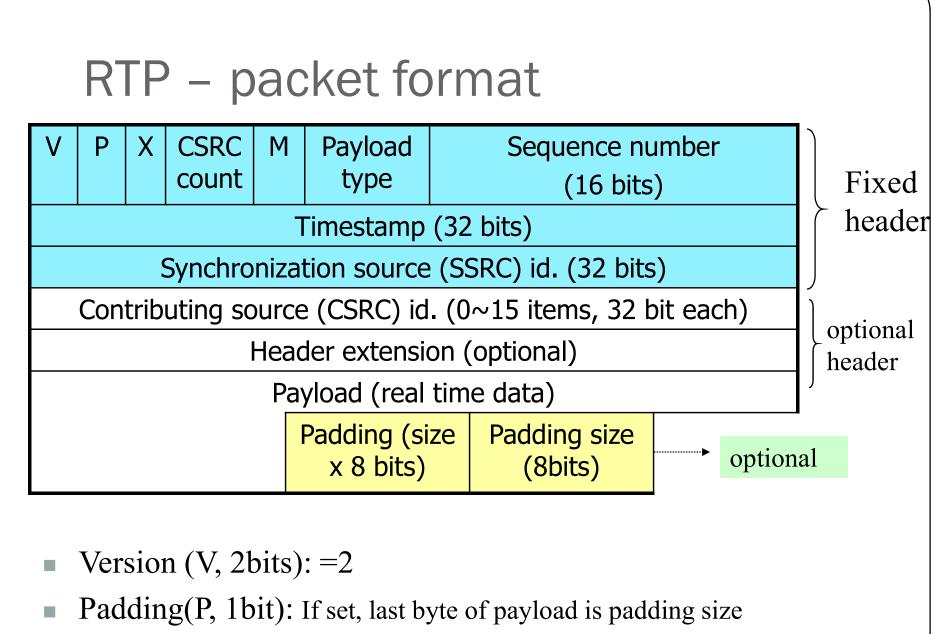
Introduction – cont.

- Containing two closely linked parts: data + control
 - RTP: Real-time transport protocol
 - Carry real-time data

• RTCP: RTP control protocol

- QoS monitoring and feedback
- Session control
- Architecture

Applic	ations			
RTP & RTCP				
Other transport and	UDP			
network protocols	IP			



Extension(X, 1bit): If set, variable size header extension exists

RTP - header

- CSRC count (4 bits): number of Contributors, max 16 can be possible
- Marker (1 bit): defined in *profile*, mark end of data
- Payload type (7 bits): Audio/Video encoding scheme
- Sequence number: random initial value, increase by one for each RTP packet; for loss detection and seq. restoration
- SSRC: identify source; chosen randomly and locally;
 collision needs to be resolved
- CSRC list: id. of contributing sources, inserted by *mixer*

Outline

Services,

Addressing

Berkley Sockets,

Multiplexing,

TCP

Connection establishment,

Connection release,

Flow control and buffering,

TCPTimer management,

TCP Congestion Control,

UDP – (Gap Analysis) content beyond syllabus

RealTimeTransport protocol(RTP),

Stream Control Transmission Protocol (SCTP),

Quality of Service (QoS),

Differentiated services,

TCP and UDP for Wireless.

SCTP

- Stream Control Transmission Protocol (SCTP) is a
 - reliable,
 - message-oriented
 - transport layer protocol.
- SCTP, however, is mostly designed for Internet applications that have recently been introduced.
- These new applications need a more sophisticated service than TCP can provide.

SCTP is a message-oriented, reliable protocol that combines the best features of UDP and TCP.

Comparison

- UDP: Message-oriented, Unreliable
- TCP: Byte-oriented, Reliable
- SCTP
 - -Message-oriented, Reliable
 - Other innovative features
 - Association, Data transfer/Delivery
 - Fragmentation,
 - Error/Congestion Control

Some SCTP applications

Protocol	Port Number	Description
IUA	9990	ISDN over IP
M2UA	2904	SS7 telephony signaling
M3UA	2905	SS7 telephony signaling
H.248	2945	Media gateway control
H.323	1718, 1719, 1720, 11720	IP telephony
SIP	5060	IP telephony

Services of SCTP

Process-to-Process Communication

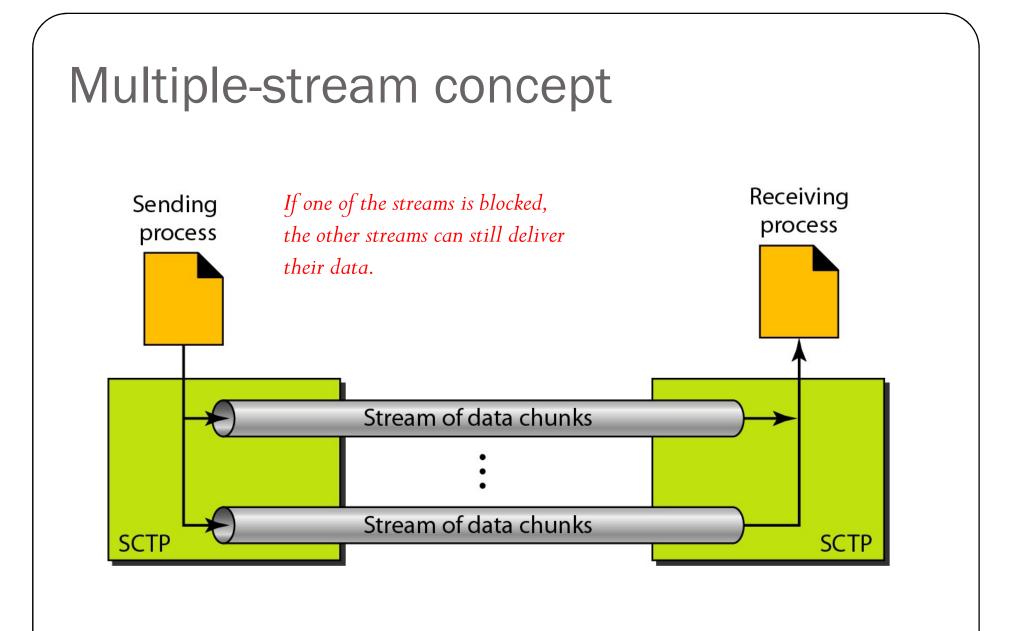
Multiple Streams

Multihoming

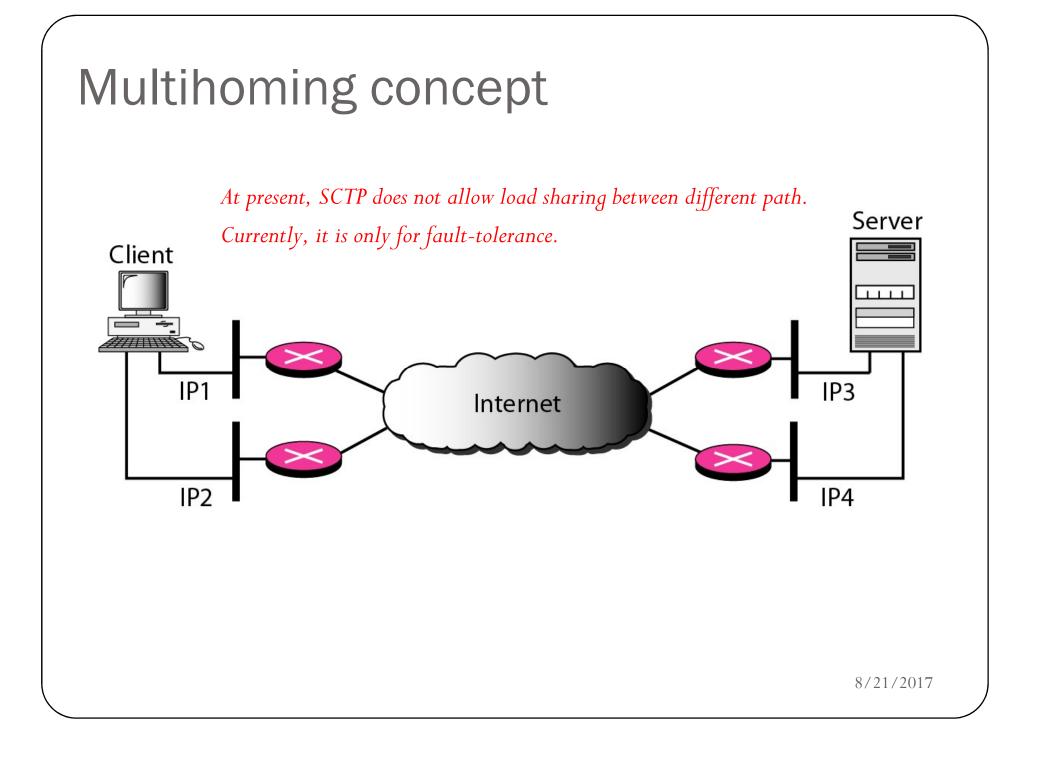
Full-Duplex Communication

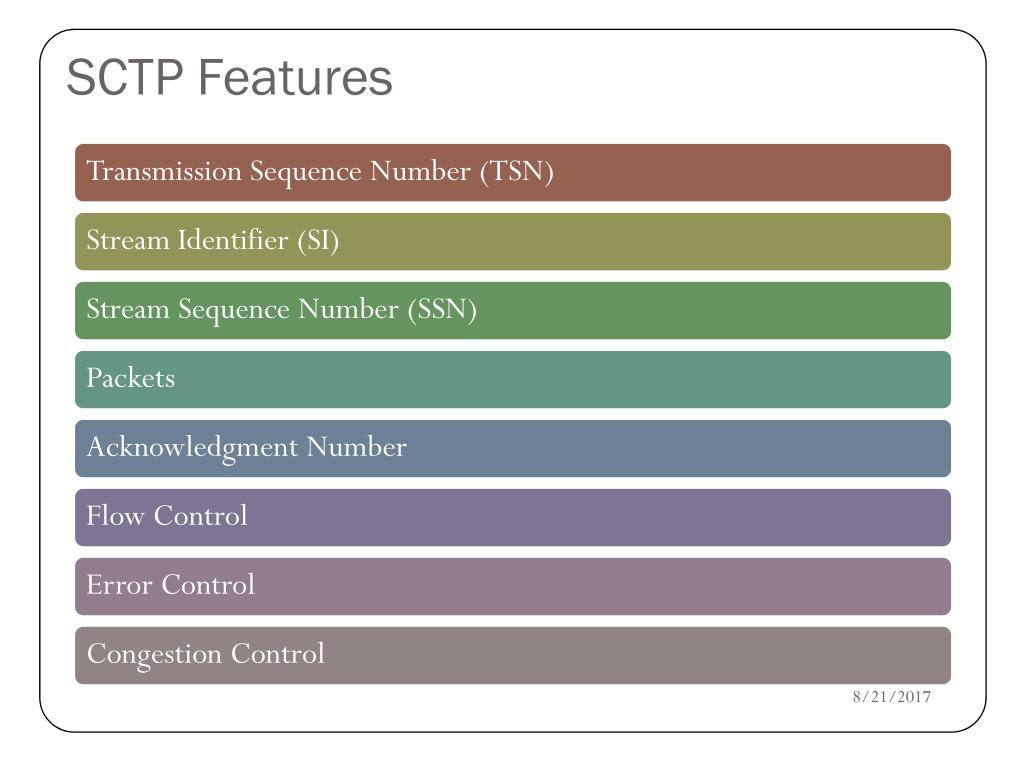
Connection-Oriented Service

Reliable Service



An association in SCTP can involve multiple streams.





In SCTP, a data chunk is numbered using a TSN.

To distinguish between different streams, SCTP uses an SI.

To distinguish between different data chunks belonging to the same stream, SCTP uses SSNs.

Comparison between UDP,TCP and SCTP

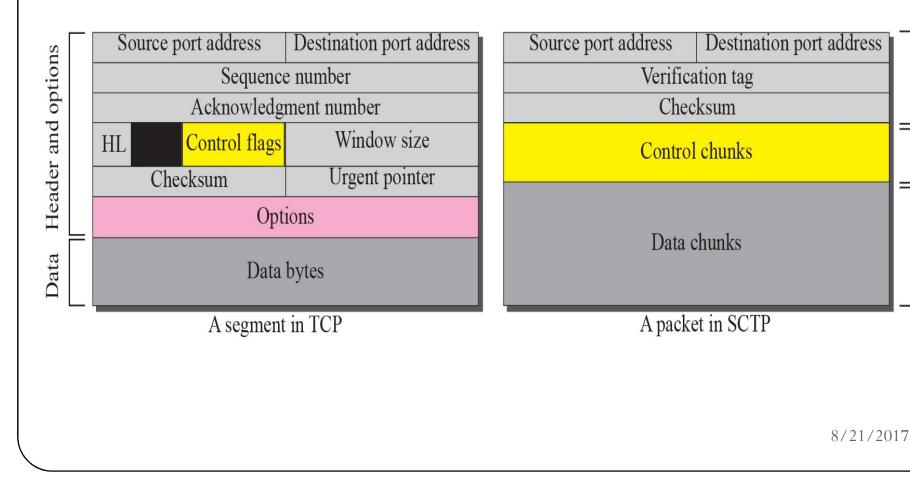
UDP	ТСР	SCTP
Message oriented protocol	Byte oriented protocol	Message oriented protocol
Preserve message boundaries	Does not Preserve message boundaries	Preserve message boundaries
Unreliable	Reliable	Reliable
No congestion and flow control	Have congestion and flow control	Have congestion and flow control
Each message follows different route so no sequencing	Each message follows same route so have in sequence data delivery	have in sequence data delivery
Port no 17	Port no 6	Port no 132

Comparison between a TCP segment and an SCTP packet

TCP has segments; SCTP has packets.

Control Header

Data

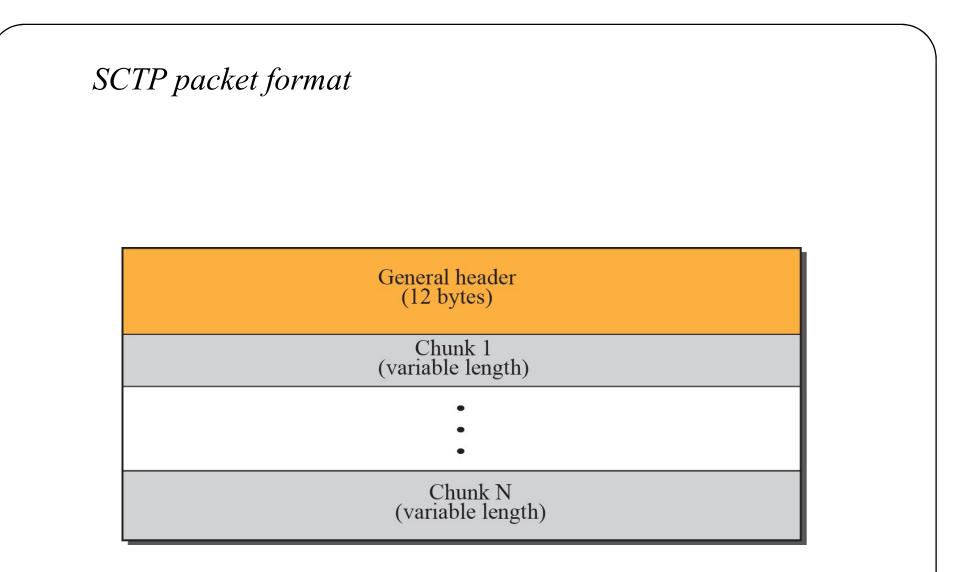


SCTP PACKET FORMAT

- □ In this section, we show the format of a packet and different types of chunks.
- □ An SCTP packet has a mandatory general header and a set of blocks called chunks.

□ There are two types of chunks:

- 1. control chunks and
- 2. data chunks.



In an SCTP packet, control chunks come before data chunks.

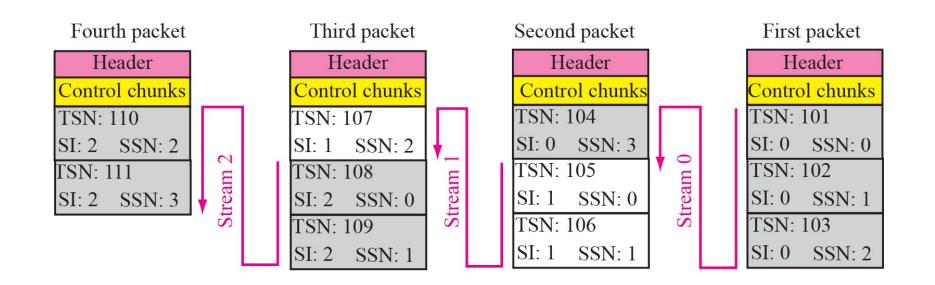
General header (Common layout of a chunk)						
			•			
Source port address 16 bits		Destination port address 16 bits				
Verification tag						
32 bits						
Checksum						
	32					

In SCTP, control information and data information are carried in separate chunks.

Data chunks are identified by three identifiers: TSN, SI, and SSN. TSN is a cumulative number identifying the association; SI defines the stream; SSN defines the chunk in a stream.

In SCTP, acknowledgment numbers are used to acknowledge only data chunks; control chunks are acknowledged by other control chunks if necessary. 8/21/20

Packet, data chunks, and streams



Flow of packets from sender to receiver

Outline

Services,

Addressing

Berkley Sockets,

Multiplexing,

TCP

Connection establishment,

Connection release,

Flow control and buffering,

TCPTimer management,

TCP Congestion Control,

UDP – (Gap Analysis) content beyond syllabus

RealTimeTransport protocol(RTP),

Stream Control Transmission Protocol (SCTP),

Quality of Service (QoS),

Differentiated services,

TCP and UDP for Wireless.

QoS Parameters

Reliability

Jitter

Delay

Bandwidth

Requirements

Reliability- *Reliability* is concerned with the ability of a *network* to carry out a desired operation according to its specifications

Jitter- **Jitter** is defined as a variation in the delay of received packets.

Delay- is the amount of time required to transmit packets.

Bandwidth- amount of information that can be transmitted over a **network** in a given amount of time

017

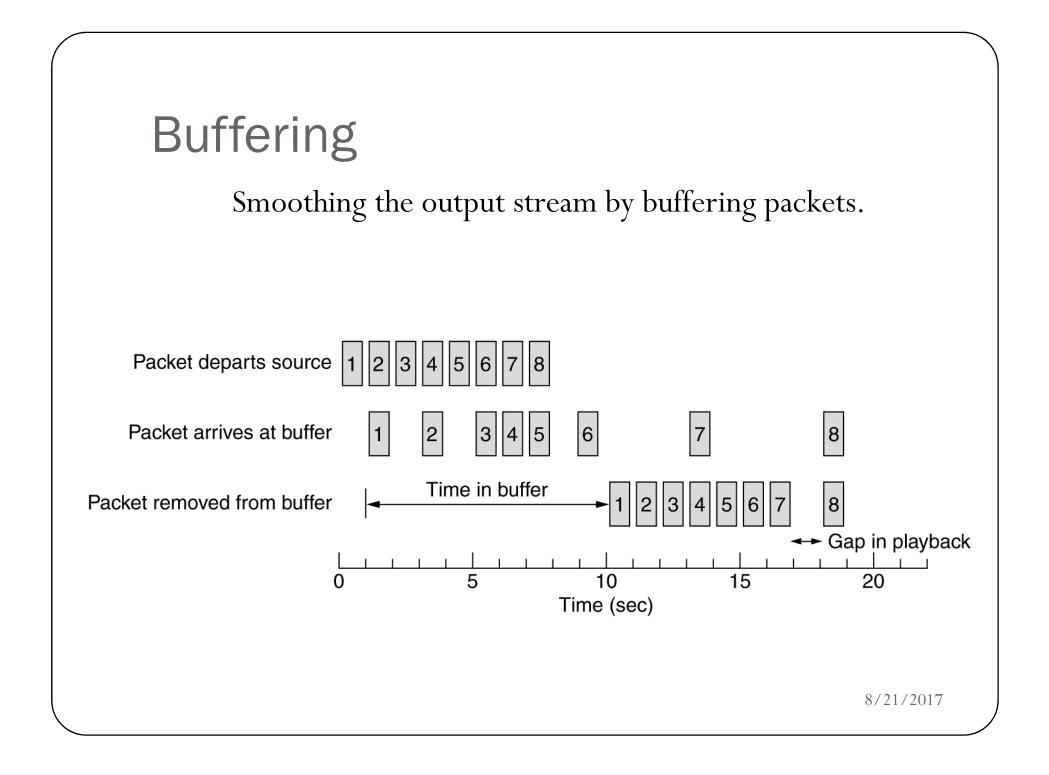
Requirements

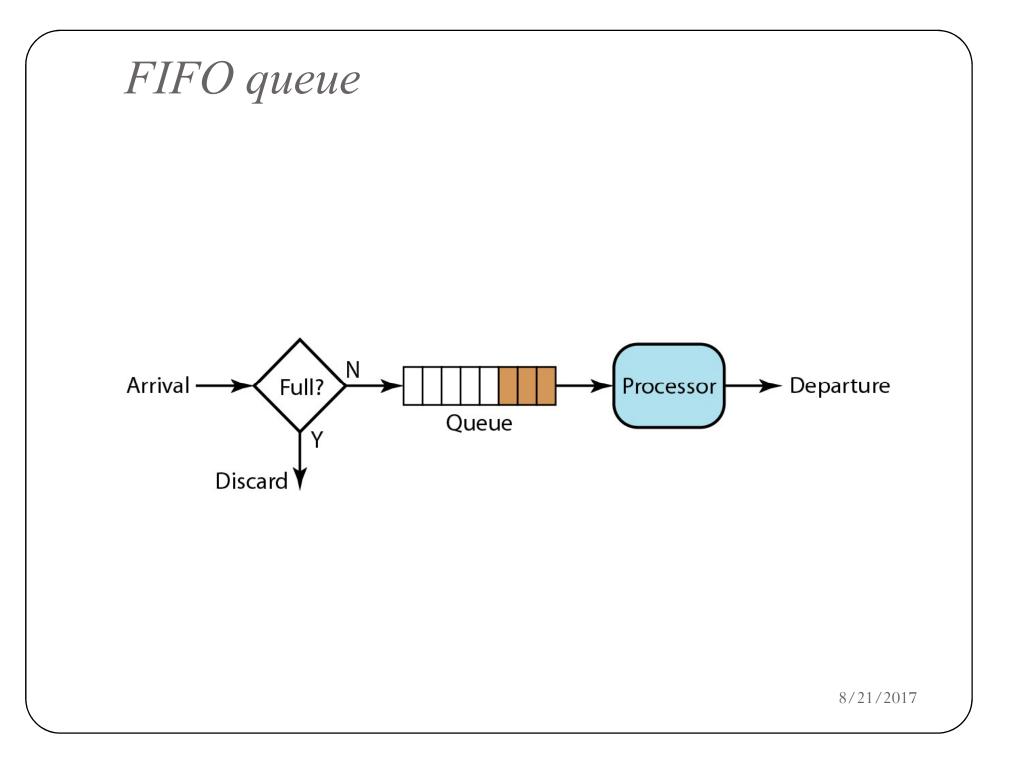
How stringent the quality-of-service requirements are.

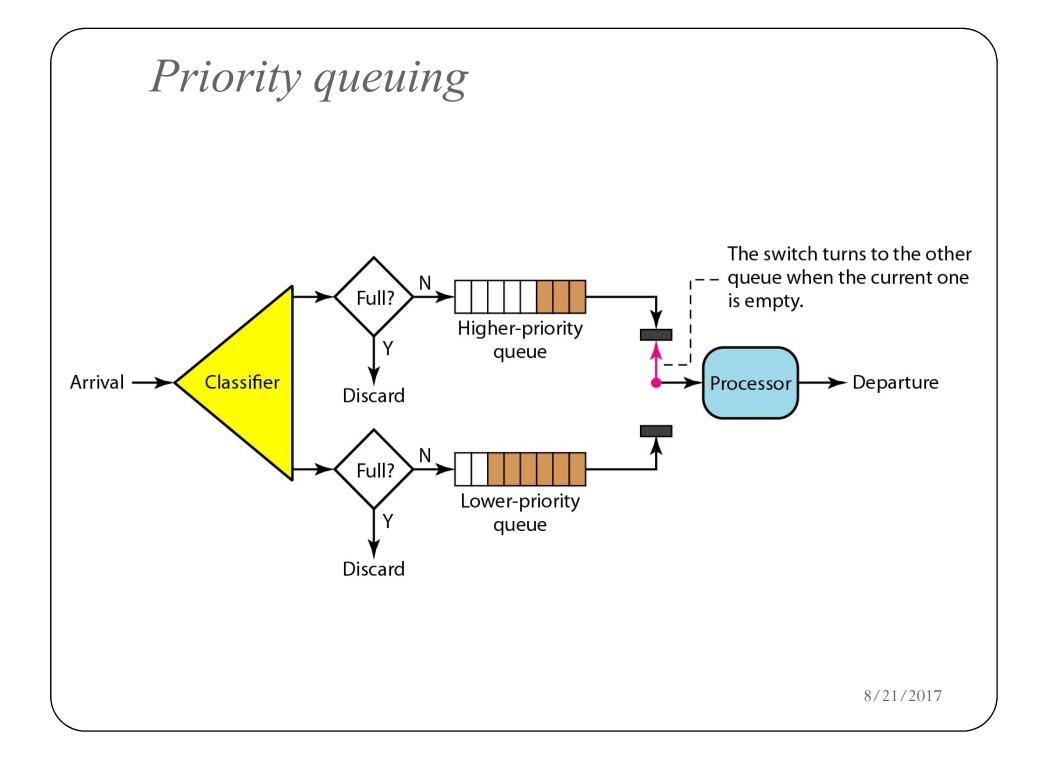
Application	Reliability	Delay	Jitter	Bandwidth
E-mail	High	Low	Low	Low
File transfer	High	Low	Low	Medium
Web access	High	Medium	Low	Medium
Remote login	High	Medium	Medium	Low
Audio on demand	Low	Low	High	Medium
Video on demand	Low	Low	High	High
Telephony	Low	High	High	Low
Videoconferencing	Low	High	High	High

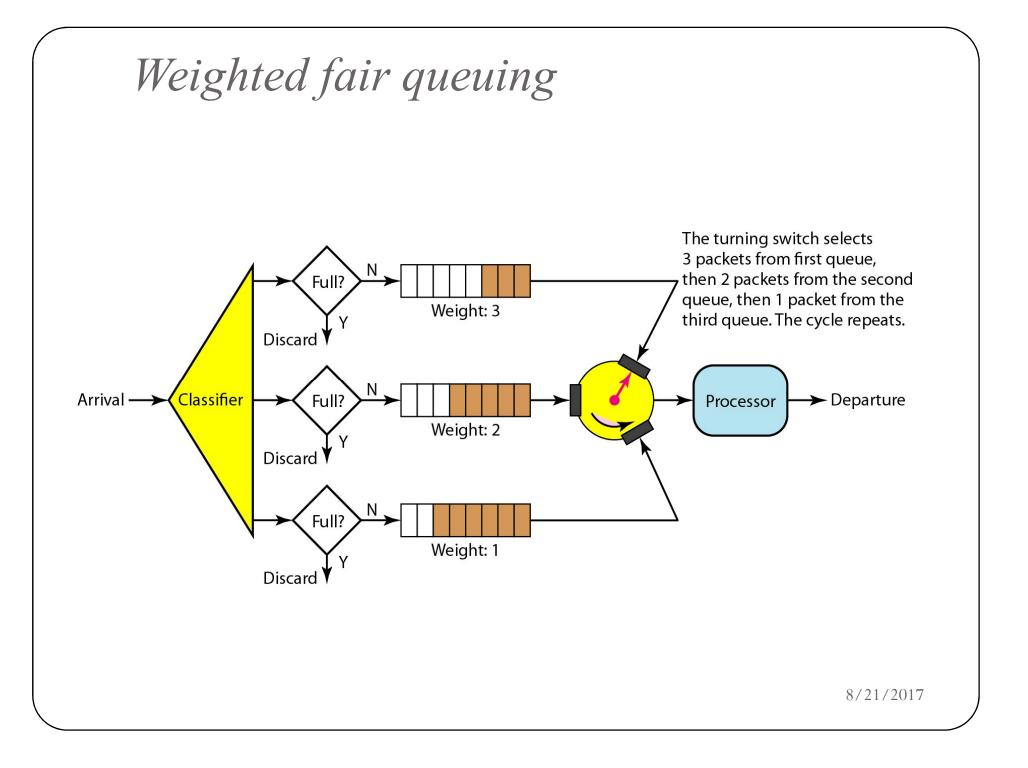
Techniques to achieve Good QoS

- Buffering
- Scheduling
 - FIFO queue
 - Priority queuing
 - Weighted fair queuing
- Traffic Shaping
 - Leaky bucket algorithm
 - Token bucket algorithm
- Resource reservation

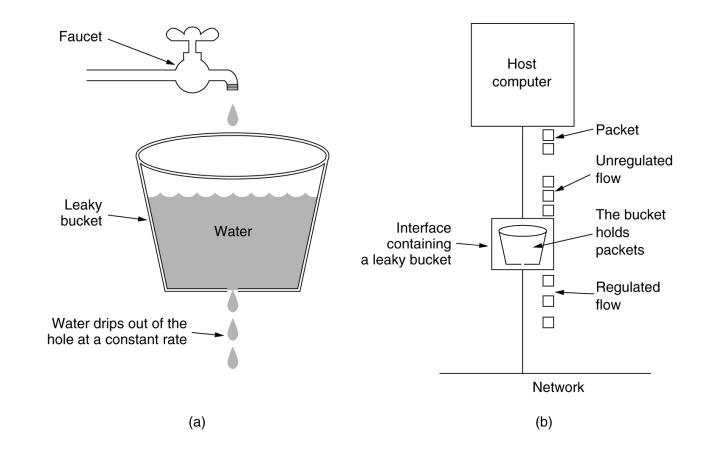






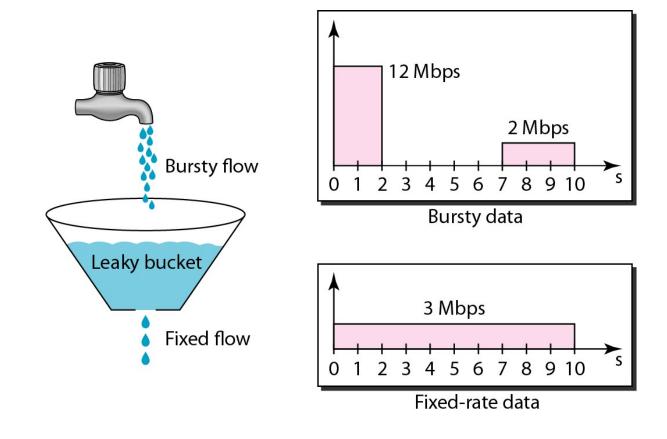


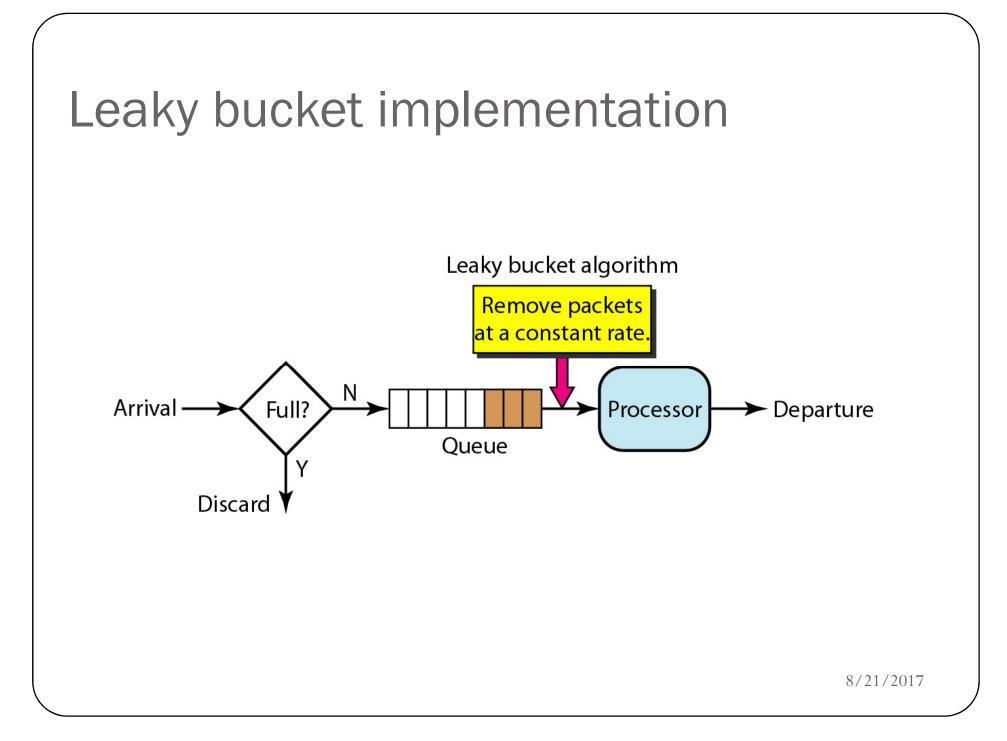




(a) A leaky bucket with water. (b) a leaky bucket with packets.





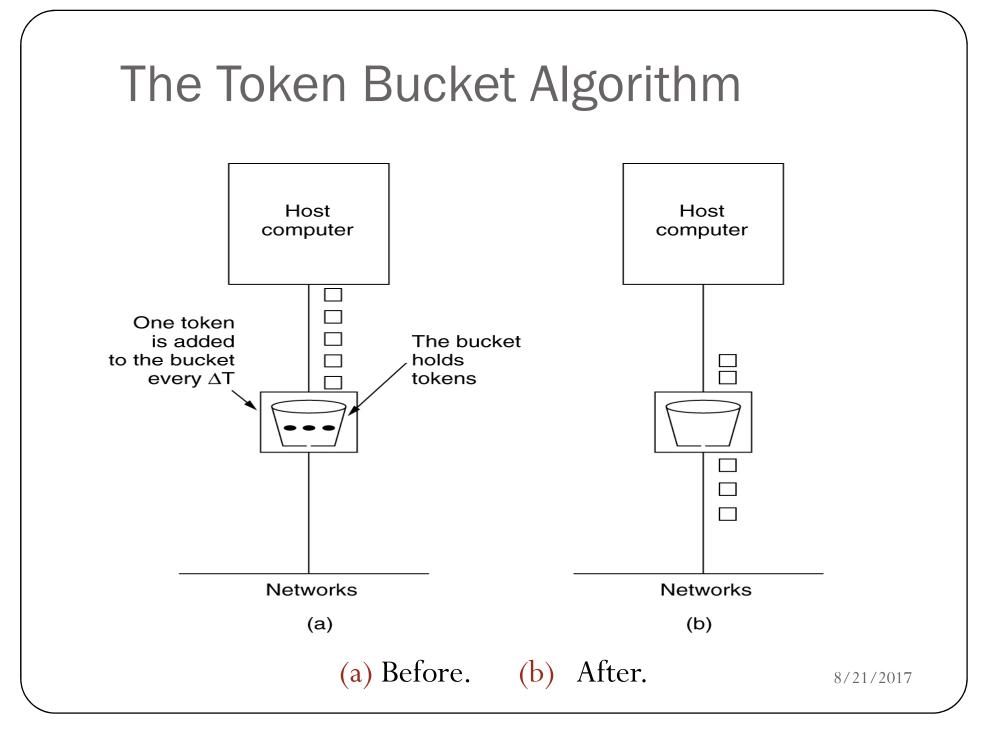


Note

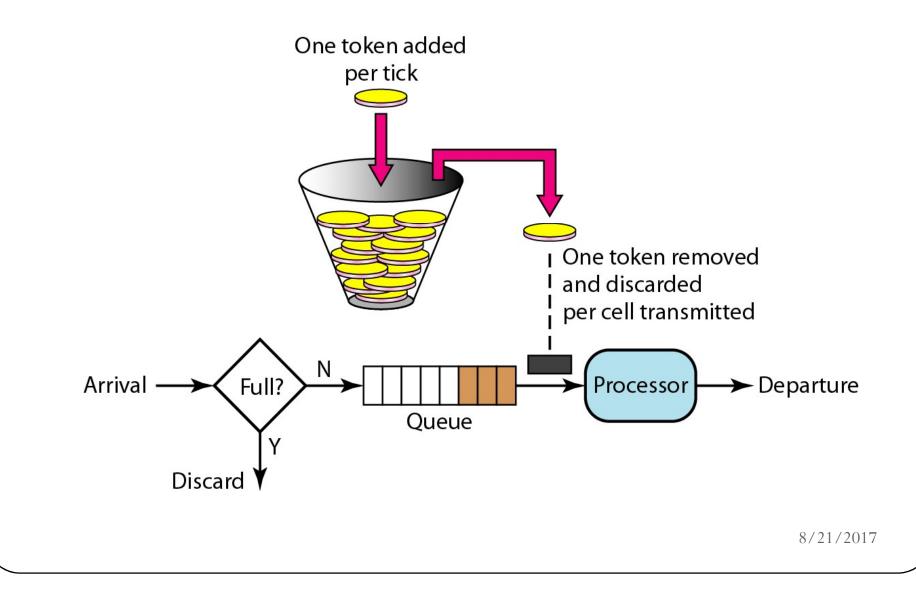
A leaky bucket algorithm shapes bursty traffic into fixed-rate traffic by averaging the data rate. It may drop the packets if the bucket is full.



The token bucket allows bursty traffic at a regulated maximum rate.



Token bucket algorithm implementation



Admission Control

An example of flow specification.

Parameter	Unit
Token bucket rate	Bytes/sec
Token bucket size	Bytes
Peak data rate	Bytes/sec
Minimum packet size	Bytes
Maximum packet size	Bytes

Outline

Services,

Addressing

Berkley Sockets,

Multiplexing,

TCP

Connection establishment,

Connection release,

Flow control and buffering,

TCPTimer management,

TCP Congestion Control,

UDP – (Gap Analysis) content beyond syllabus

RealTimeTransport protocol(RTP),

Stream Control Transmission Protocol (SCTP),

Quality of Service (QoS),

Differentiated services,

TCP and UDP for Wireless.

Two models to provide Quality of Service

Integrated Services

Differentiated Services

Integrated Services

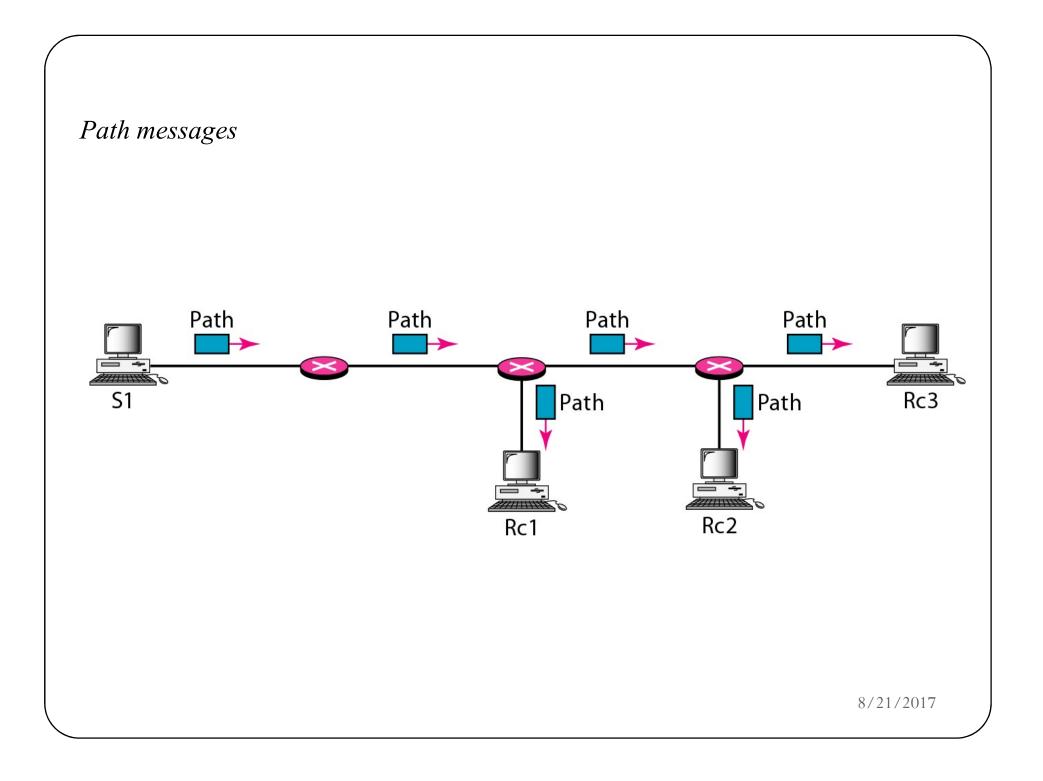
- Flow based QoS model
- Which means used need to create a flow, a kind of virtual circuit from source to destination and inform all routers about the resource requirement
- This kind of reservation of resources is done by a protocol called RSVP(Resource Reservation Protocol)

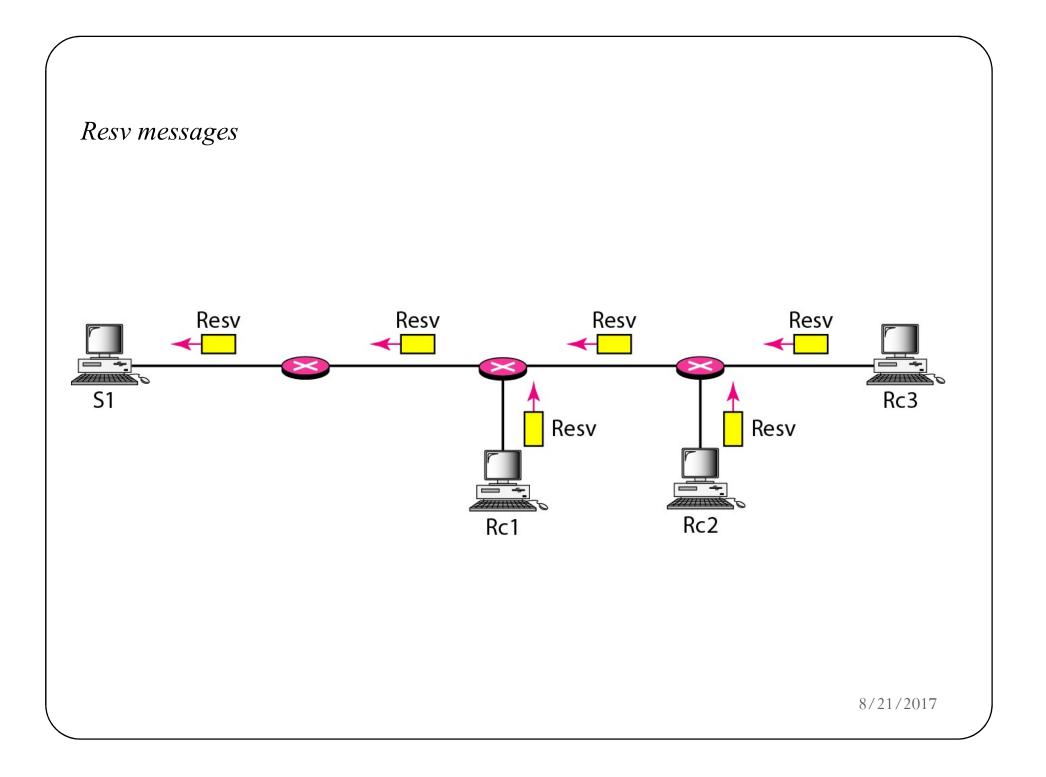
Integrated Services

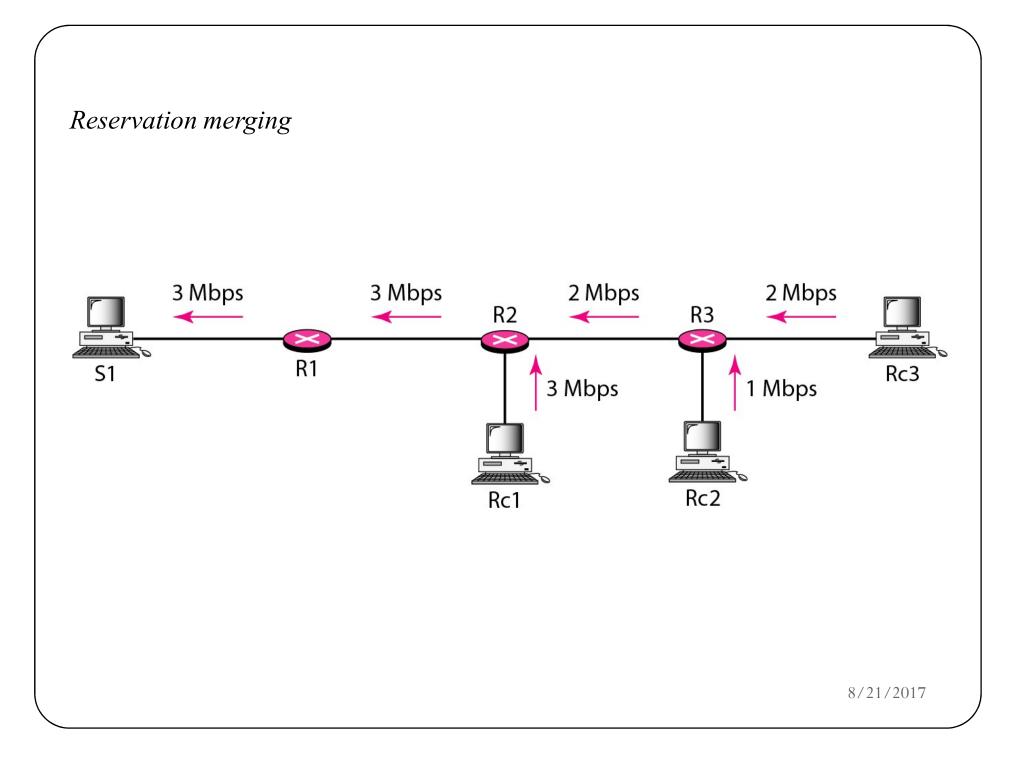
- Resource reservation means reserve how much buffer, bandwidth etc is needed.
- When a router receives flow specification from an application, it decides to admit or deny the service
- Two classes of service is defined for Integrated serviced
- 1. Guaranteed Service Class(For real time application)
- 2. Controlled-load Service(For application require reliablility)

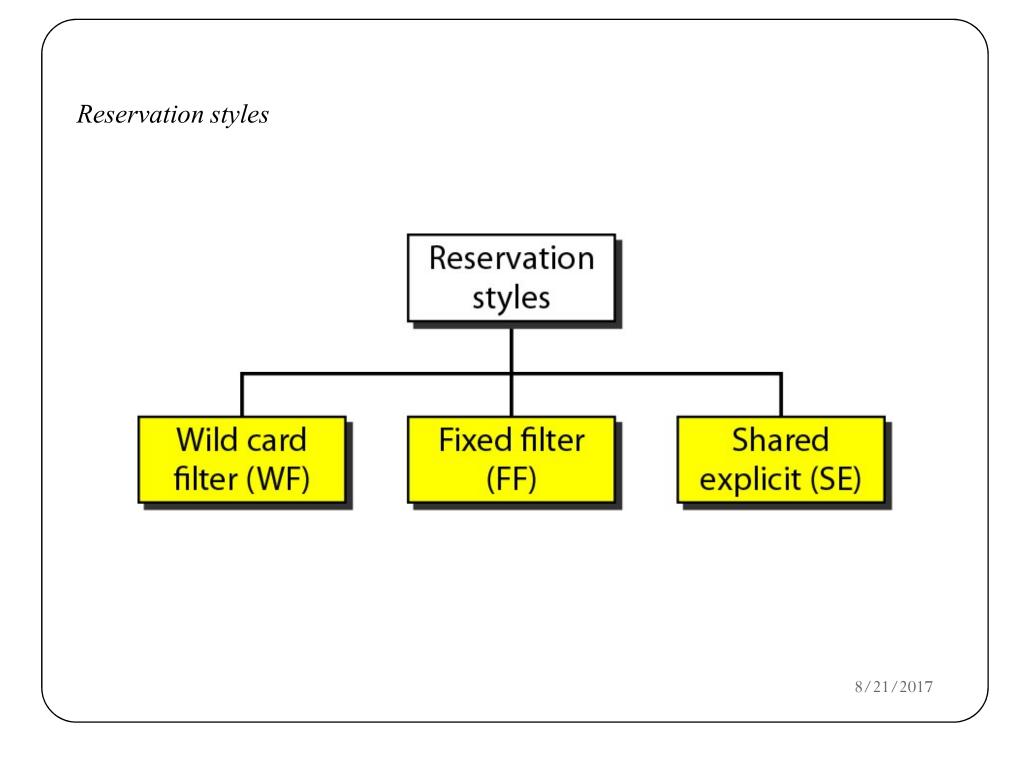
RSVP-The Resource ReSerVation Protocol

- The **Resource Reservation Protocol** (**RSVP**) is a Transport layer protocol designed to reserve resources across a network for an Integrated service network.
- RSVP operates over an IPV4 or IPV6 and provides resource reservations for multicast or unicast data flows
- RSVP can be used by either host or routers to request or deliver specific levels of quality of service (QoS) for application data streams or flows.
- RSVP defines how applications place reservations and how they can give up the reserved resources once the need for them has ended.









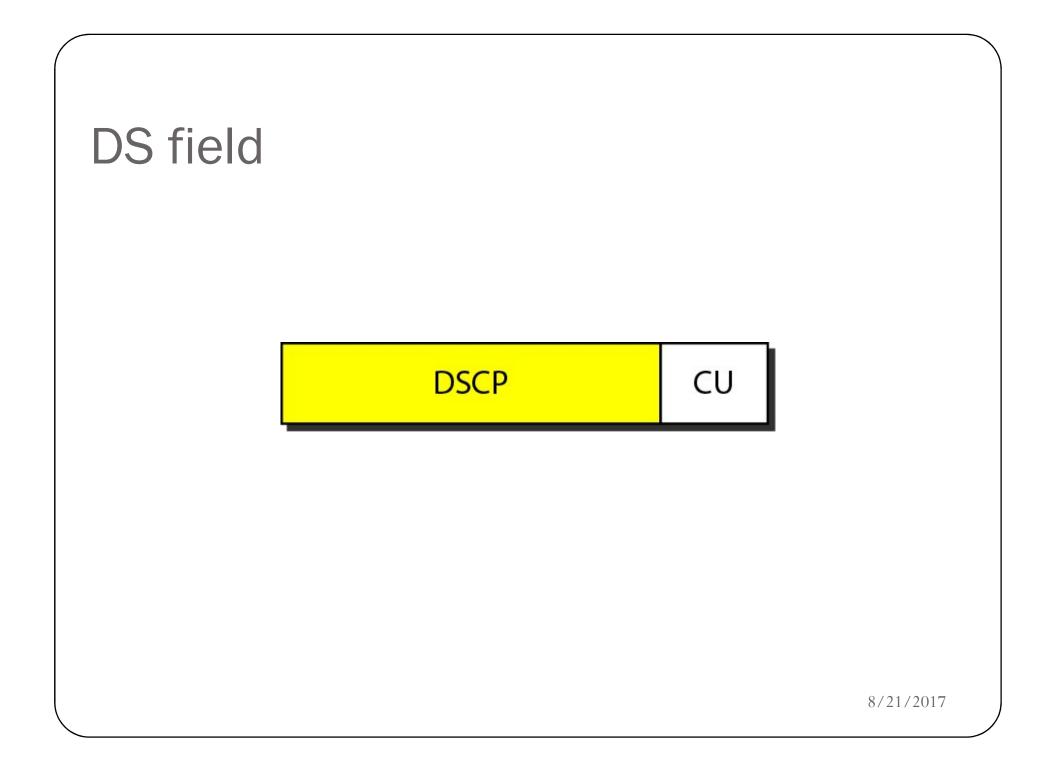
Problems with Integrated Services

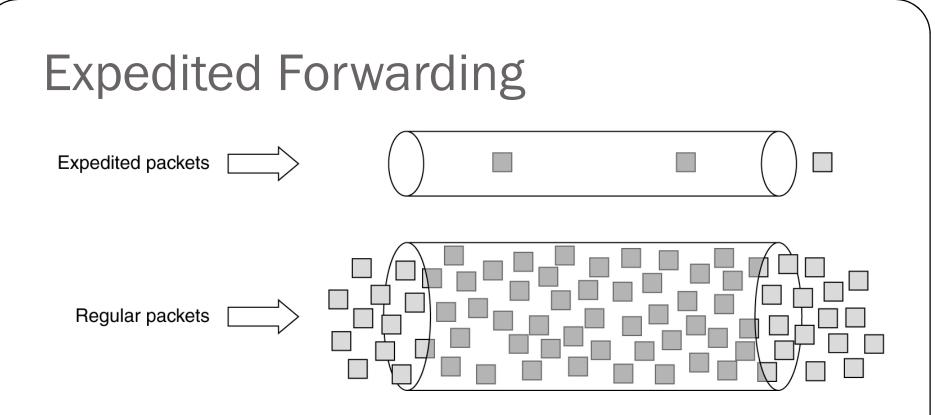
Scalability: Each router keep information for each flow. So does not possible to scale more

Service Type Limitation: Only two types of services are provided guaranteed and control based

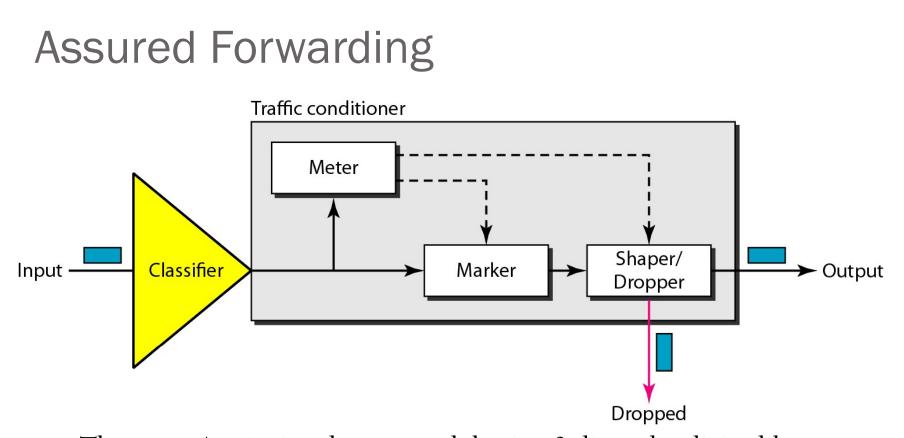
Differentiated Services

- Handles shortcomings of Integrated Services .
- In differentiated model router does not store information about flows.
- No advance reservation is required
- It is a Class based service model
- Each packet contains a field called DS field
- It has two types of models
- 1. Expedited forwarding
- 2. Assured Forwarding





- In this model two classes of service is available:
 - 1>Regular
 - 2> Expedited
- Expedited packets experience a traffic-free network.



- There are 4 priority classes , each having 3 discard policies like low, medium and high.
- Traffic controller have Classifier, Marker and Shaper/Dropper
- Packet is classified according to priority, then marked according to their class .
- Shaper/dropper filter these packet that may drop or delay the _{8/21/2017} packet.

Outline

Services,

Addressing

Berkley Sockets,

Multiplexing,

TCP

Connection establishment,

Connection release,

Flow control and buffering,

TCPTimer management,

TCP Congestion Control,

UDP – (Gap Analysis) content beyond syllabus

RealTimeTransport protocol(RTP),

Stream Control Transmission Protocol (SCTP),

Quality of Service (QoS),

Differentiated services,

TCP and UDP for Wireless.

TCP over Wireless : outline

- TCP over Wireless: Problems
- TCP over Wireless: Solutions/Schemes
 - Split TCP
 - 1.IndirectTCP
 - 2.Selective repeat protocol
 - 3.MobileTCP
 - TCP-aware link layer
 - 1.Snoop
 - 2.WTCP
 - Link layer protocol
 - End-to-end protocol
 - 1.Selective Acknowledgement
 - 2.Explicit Loss Notification

TCP over Wireless: Problems

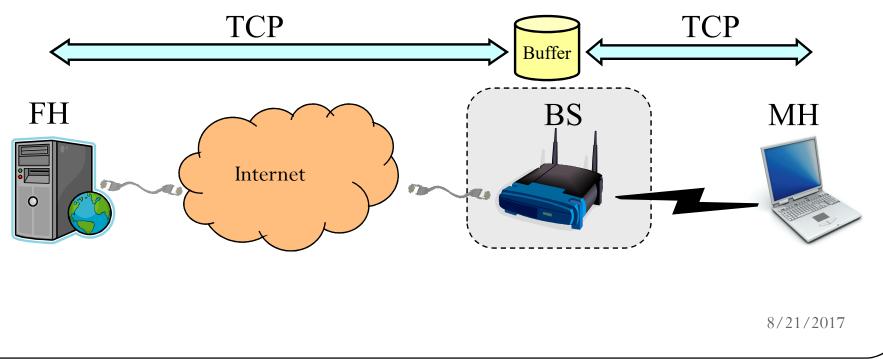
- TCP has been optimized for wired networks.
- Any packet loss is considered to be the result of network congestion and the congestion window size is reduced drastically as a precaution.
- Sources of errors in wireless links:
- 1. Due to hands off between cells
- 2. Packet losses due to futile transmissions
- 3. Packet losses due to transmission errors in wireless links

TCP over Wireless : outline

- TCP over Wireless: Problems
- TCP over Wireless: Solutions (Schemes)
 - Split TCP
 - 1.Indirect TCP
 - 2.Selective repeat protocol
 - 3.MobileTCP
 - TCP-aware link layer
 - 1.Snoop
 - 2.WTCP
 - Link layer protocol
 - End-to-end protocol
 - 1.Selective Acknowledgement
 - 2.Explicit Loss Notification

Split TCP: Indirect TCP

- I-TCP splits end-to-end TCP connection into two connections
 - Fixed host to BS
 - BS to mobile host
- Two TCP connections with independent flow/congestion control contexts
- Packets buffered at BS

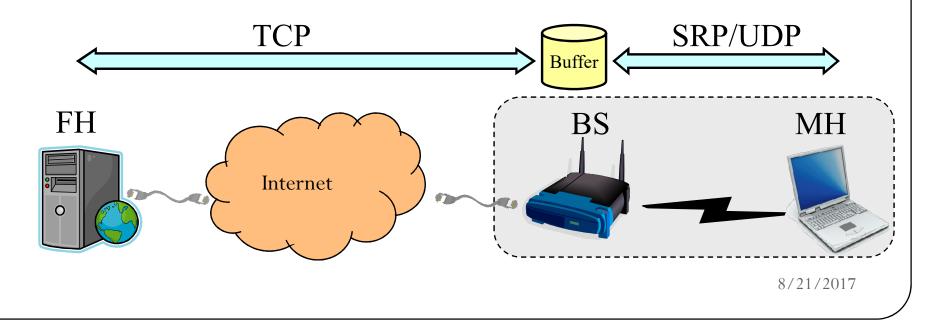


Split TCP: Indirect TCP

- Pros
 - Separates flow and congestion control of wireless and wired
 --higher throughput at sender
- Cons
 - Breaks TCP end-to-end semantics
 - Ack at FH does not mean MH has received the packet
 - BS failure causes loss of data
 - Neither FH nor MH can recover the data
 - On path change, data has to be forwarded to new BS
 - Wireless part is the bottleneck

Split TCP: Selective Repeat Protocol

- Similar to I-TCP but uses SRP/UDP (Selective Repeat Protocol over UDP) over wireless link, Improving End-to-End Performance of TCP over Mobile Internetworks
- Pros
 - Better performance over wireless links
- Cons
 - All cons of I-TCP except last one



Split-TCP: Mobile TCP

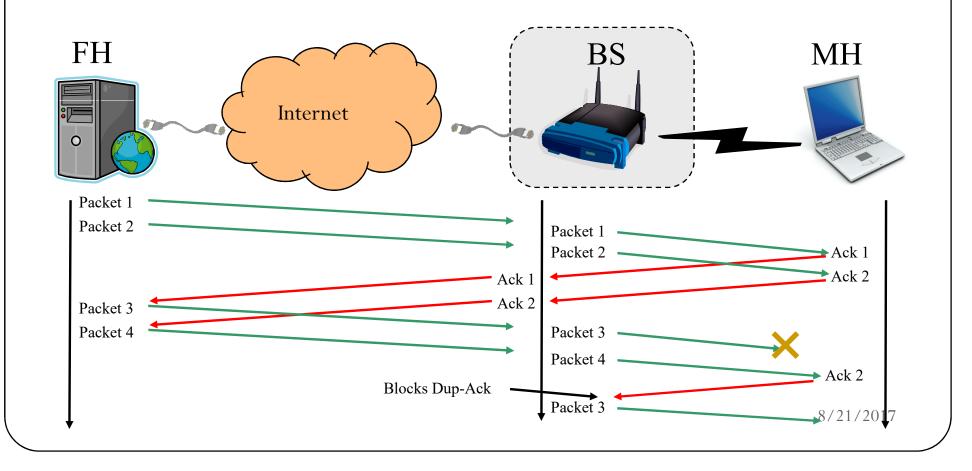
- Similar to I-TCP but tries to keep TCP end-to-end semantics
- No buffering , no retransmission at base station BS.
- BS only monitors all packets and only acks the last packet after it is received by MH
- Pros
 - Data will be recovered eventually after BS failure
 - BS buffer does not overflow
- Cons
 - Worse performance
 - Still not exactly the TCP semantics

TCP over Wireless : outline

- TCP over Wireless: Problems
- TCP over Wireless: Solutions (Schemes)
 - Split TCP
 - 1.Indirect TCP
 - 2.Selective repeat protocol
 - 3.MobileTCP
 - TCP-aware link layer
 - 1.Snoop
 - 2.WTCP
 - Link layer protocol
 - End-to-end protocol
 - 1.Selective Acknowledgement
 - 2.Explicit Loss Notification

TCP-aware Link Layers: Snoop

- Link layer is aware of TCP traffic
- BS caches data and monitors acks. Retransmits on duplicate acks and drops duplicate acks



TCP-aware Link Layers: Snoop

- Pros
 - No modification to FH and MH
 - BS only keeps soft state—BS failure does not break TCP
- Cons
 - Does not work with encrypted packets
 - Does not work if data packets and acks traverse different paths
 - Increases RTT—high timeout

TCP-aware Link Layers: WTCP

- Similar to Snoop
- WTCP corrects RTT by modifying the timestamp in return acks

