SC2008 TUT

SC2008-CE3005-CZ3006 (Computer Network)

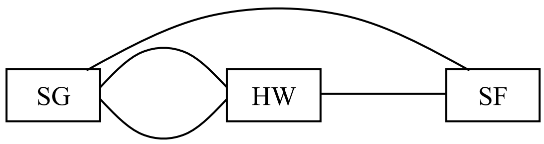
Part I: Tutorial – 1

1. Compare the two layering models – OSI vs. Internet (TCP/IP).

2. Show with the help of a diagram, the various headers that are appended to data at the sending host. Explain what happens to these headers at the switching nodes in a Wide Area Network.

3. In network resilience, the link failure probability can be interpreted as the percentage of the time that the link goes down during a time window. In a carrier-grade network, it is often required that the network should have 6 9's (i.e, 99.9999%) reliability. Please calculate the duration of allowable downtime per year for this network? (hint: *using the definition of failure probability*)

4. Singapore (SG) is connected to San Francisco (SF), via an intermediate node at Hawaii (HW). Two independent links connect between Singapore and Hawaii, and a long-range link connects between Singapore and San Francisco. Assume that each link fails independently with probability of 0.05. Calculate the probability in which SG is disconnected from SF.



$a = 10 \times 10^{-3} = 0.01$

$N \leq 8$

$N = 32 \times 10^3$

$1 + 2(0.01)$

1.002

$N = 32 \times 1002$

SC2008-CE3005-CZ2006 (Computer Network)

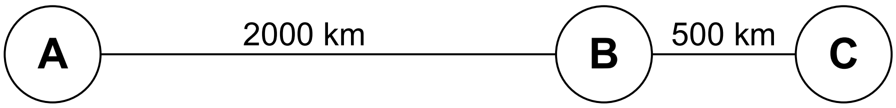
Part I: Tutorial – 2

$U = \frac{N}{1+2a}$

$32000 = \frac{N}{1+2 \cdot 0.01}$

1. A 64kpbs leased line connects two bank branches and is used for transferring secure bank information. A link layer protocol with 3-bit sequence number is deployed. Determine the minimum window size required to ensure that the throughput is at least 32 Kbps. Assume that the average packet size is 80 bytes, the signal propagation delay between the two sites is 10 *ms*, and the probability of error is negligible. (Hint: *flow control*)

2. In the figure below, frames are generated at node A and sent to node C through node B.



Determine the minimum transmission rate required between nodes B and C so that the buffers of node B are not flooded, based on the following:

$a = 10 \times 10^{-8}$

(a) The data rate between A and B is 100 *Kbps*.
(b) The propagation delay is 10 *μsec/km* for both links.
(c) The lines are full duplex between the nodes.
(d) All data frames are 1000 bit long; ACK frames are separate frames of negligible length.
$N_{ab}\ 3$
(e) Between A and B, a sliding-window protocol with a window size of 3 is used, and each frame is acknowledged individually.
(f) Between B and C, a stop and wait is used.
(g) There is no error, and the processing delay at the nodes is negligible.

(Hint: *outgoing rate from B should be at least the same as its incoming rate*)

3. Consider a communication link between two cities of City *S* and City *D*. The frame transmission rate on the link is 100 *kbps* and the frame length is 25 bytes. The distance between City *S* and City *D* is 5 *km* and the propagation delay 3 *ms/km*. The communication link suffers from an average frame error probability of 0.2 and we adopt a Selective-Reject ARQ mechanism between City *S* and City *D* for reliable communication.

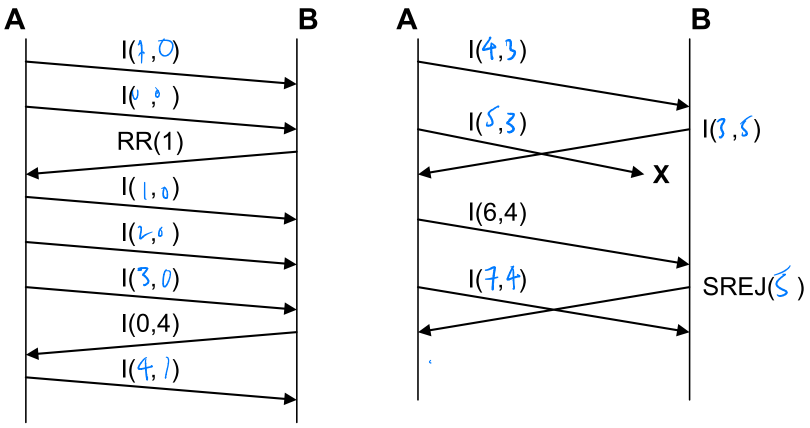
   a) If the window size is 10, compute the link utilization from City S to City D.

   b) Compute the minimum window size and the corresponding number of bits reserved for frame sequence in the header, in order to maximize the link utilization between City S and City D.

(Hint: *ARQ scheme has window size limit*)

4. Let us define a specific link layer protocol with three types of frames: 1) Information Frame I(*i*,*j*) with *i* indicating the current sender sequence and *j* acknowledging receiver sequence (i.e., ready for receiving the *j*-th frame from the receiver), 2) Receiver Ready (or ACK) frame RR(*j*), and Selective Reject (or NACK) frame SREJ (*j*). Complete the time sequence diagram by adding the send and receive sequence numbers to the frames. Also include the next few frames that are sent by A and B to completely recover from the error if any. Assume that 3 bits are allocated for sequence numbering. (Hint: *using a divide-&-conquer strategy*)

$2^3 = \{0..7\}$



(a) (b)

SC2008-CE3005-CZ3006 (Computer Network)

Part I: Tutorial – 3

1. The CSMA/CD specification for a medium is given below:
   (a) Maximum cable length per segment is 200 meters and the propagation delay is 1 μsec per 100 meters.
   (b) Maximum number of stations per segment is 20.
   (c) Maximum number of repeaters between any two segments (or say, between any two stations) of a LAN is 4. Delay through each repeater is 2 μsec.

   Determine the minimum frame size required for correct operation of the CSMA/CD protocol if the network is run at 20 Mbps.

2. Consider an Ethernet 10BASE-T single segment LAN where three stations are connected to an Ethernet hub, and the distance between each station and the Ethernet hub is the same. Assume that each of the three stations transmits a new frame at exactly the same time resulting in a collision, what is the probability that the next event on the channel is also a collision?

   (Hint: *if nothing happens on the first retrial time slot, it does not count as an event.*)

3. In a local area network using the CSMA/CD protocol, a modified Binary Exponential Backoff scheme is used if a collision is detected in the channel. Assume that two stations (A and B) are transmitting and their frames collide in one time slot. Each of them will retransmit its data frame over a window of size 2 slots. Station A retransmits in slot 0 with probability of p and station B retransmits in slot 0 with probability of q.

   (i) If $p = 1/3$ and $q = 2/3$, what is the probability that the first event in the channel will be a success?

   (ii) How would you maximize the probability that the first event in the channel will be a success, by choosing proper values for p and q?

4. Suppose that an 11-Mbps 802.11b WLAN is transmitting 64-byte frames back-to-back over a radio channel with a bit error rate of 10^{-7}. How many frames per second will be damaged on average?

5. You are commissioned to design an experimental wireless network for SCE to support its 100 wireless devices. You have decided to adopt a multi-access reservation protocol (MARP) for frame transmission in the data link layer. Specifically, each transmission cycle consists of two phases: a reservation phase and a transmission phase. In the reservation phase, a chosen MAC protocol is used for transmission stations to reserve the channel; and in the transmission phase, the station that successfully reserves the channel transmits one frame. The data rate in

the wireless channel is 1 *Mbps*. The length of the data frame is 1000 *bits*, among which the reservation frame carries 10 information *bits*.

(i) If the MAC protocol used in the reservation phase has a utilization of 0.8, what will be the throughput of the MARP?

(ii) Assume that the slotted Aloha protocol is used in the reservation phase. The utilization for the slotted Aloha protocol is Ge^{-G}, where $G = np$, n is the number of stations, and p is transmission probability. Calculate an optimal transmission probability to maximize the throughput of the MARP, and the corresponding maximum throughput?

In a local area network using the CSMA/CD protocol, a modified Binary Exponential Backoff scheme is used if a collision is detected in the channel. Assume that two stations (A and B) are transmitting and their frames collide in one time slot. Each of them will retransmit its data frame over a window of size 2 slots. Station A retransmits in slot 0 with probability of p and station B retransmits in slot 0 with probability of q.

If $p = 0.25$ and $q = 0.75$, what is the probability that the first event in the channel will be a success?

1. Binary Exponential Backoff (BEB)

- ○ 0.125
- ○ 0.1875
- ● 0.625
- ○ 0.5625

2. Which of the following MAC addresses is a multicast address?

B is odd number
↳ multicast

0%	0%	0%	0%
4A:30:10:21:10:1A	46:20:1B:2E:08:EE	4B:16:7E:28:28:6F	FF:FF:FF:FF:FF:FF

Computer Networks

Part I - Tutorial 2

Question 1

A 64 kbps leased line connects two bank branches and is used for transferring secure bank information. A link layer protocol with 3-bit sequence number is deployed. Determine the minimum window size required to ensure that the throughput is at least 32 Kbps. Assume that the average packet size is 80 bytes, the signal propagation delay between the two sites is 10 ms, and the probability of error is negligible

Q1 answer

Given:

Tx_rate = 64 Kbps

Propagation delay (P_delay) = 10 millisecond

Packet size = 80 bytes = 640 bits

3 bit used for frame sequencing

Calculate Normalised propagation delay

a = Propagation delay /average frame transmission time

Ave frame tx_time = ave. frame size /Tx_rate

= 640 bit /64 K

= 10 msec.

a = 10 msec. /10 msec = 1

Q1 answer

The link utilisation for 32 Kbps is 32K/64K which is 0.5

Utilisation of the link is given by

U = N/(1 +2a) where N is the window size

$0.5 \leq N /(1+ 2*1)$

$N \geq 3/2 = 1.5$

Round up the window size , i.e., **2 (< 23=8)**

Question 2

In the figure below, frames are generated at node A and sent to node C through node B. Determine the minimum transmission rate required between nodes B and C so that the buffers of node B are not flooded, based on the following:

(a) The data rate between A and B is 100 *Kbps*.

(b) The propagation delay is 10 $\mu sec/km$ for both links.

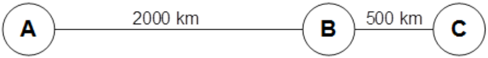
(c) The lines are full duplex between the nodes.

(d) All data frames are 1000 bit long; ACK frames are separate frames of negligible length.

(e) Between A and B, a sliding-window protocol with a window size of 3 is used, and each frame is acknowledged individually.

(f) Between B and C, a stop and wait is used.

(g) There is no error, and the processing delay at the nodes is negligible.



Question 2: Transmission Rate Calculation

A ▶ B: Propagation Time = 2000 *10 μsec = 20 msec

Transmission Time = 1000/(100*103) = 10 msec

B ▶ C: Propagation Time = 500 *10 μsec = 5 msec

Transmission Time = x = 1000/R msec

R = Data rate between B and C in kbps

Start sending 1st packet 00

Start sending 2nd packet 10

Start sending 3rd packet 20

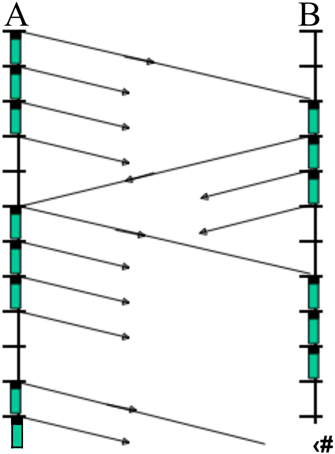
Ack recd for 1st packet, start sending 4th packet 50

Ack recd for 2nd packet, start sending 5th packet 60

Ack recd for 3rd packet, start sending 6th packet 70

Ack recd for 4th packet, start sending 7th packet 100

Ack recd for 5th packet, start sending 8th packet 110



Question 2: Transmission Rate Calculation

A → B: Takes 50 msec to transmit 3 frames.

B → C: Takes 10 + x msec to transmit one frame

Therefore,

$50 = 3\,(10 + x)$ → $x = 20/3$

$x = 1000/R$ → $R = 1000/x = 150$ kbps

Another approach to solve this question

The required condition:

$$\frac{3}{\ } \times 100\ kbps <$$

Because

$$3$$

and

$$\frac{1}{\ }\times R\ kbps =$$

the required condition becomes

which leads to

Question 3

Consider a communication link between two cities of City *S* and City *D*. The frame transmission rate on the link is 100 *kbps* and the frame length is 25 bytes. The distance between City *S* and City *D* is 5 *km* and the propagation delay 3 *ms*/*km*. The communication link suffers from an average frame error probability of 0.2 and we adopt a Selective-Reject ARQ mechanism between City *S* and City *D* for reliable communication.

a) If the window size is 10, compute the link utilization from City S to City D.

b) Compute the minimum window size and the corresponding number of bits reserved for frame sequence in the header, in order to maximize the link utilization between City S and City D.

Q3(a): Selective-Reject ARQ Utilization

- Frame transmission time

$$T_f = 25bytes \times 8bits / bytes \div 100kbps = 2ms$$

- Propagation delay

$$T_p = 5km \times 3ms / km = 15ms$$

- Normalized propagation delay

$$a = {}^{T_p}\!/_{T_f} = 15/2 = 7.5$$

- Utilization of the ARQ

$$U = \frac{W(1-P)}{1+2a} = \frac{10(1-0.2)}{1+2\times 7.5} = \frac{8}{16} = 50\%$$

Q3(b): Minimum Window Size

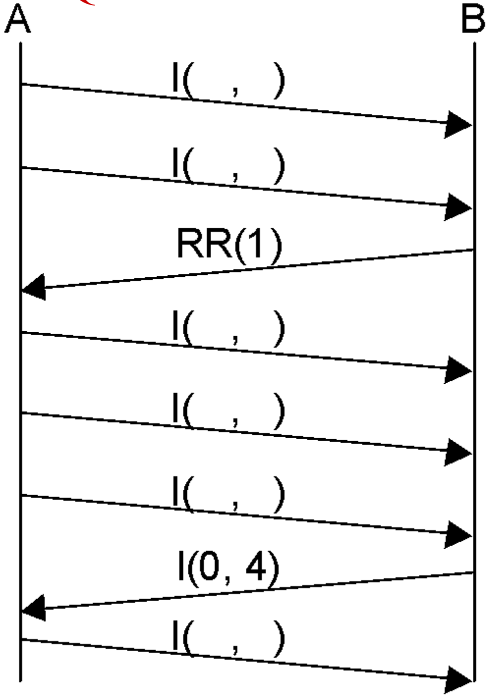
$$W \geq 1 + 2a = 1 + 2x7.5 = 16$$

- Minimum window size to maximize the throughput is **16**
- In Select-Repeat ARQ, for a k-bit sequence, the maximum window size is

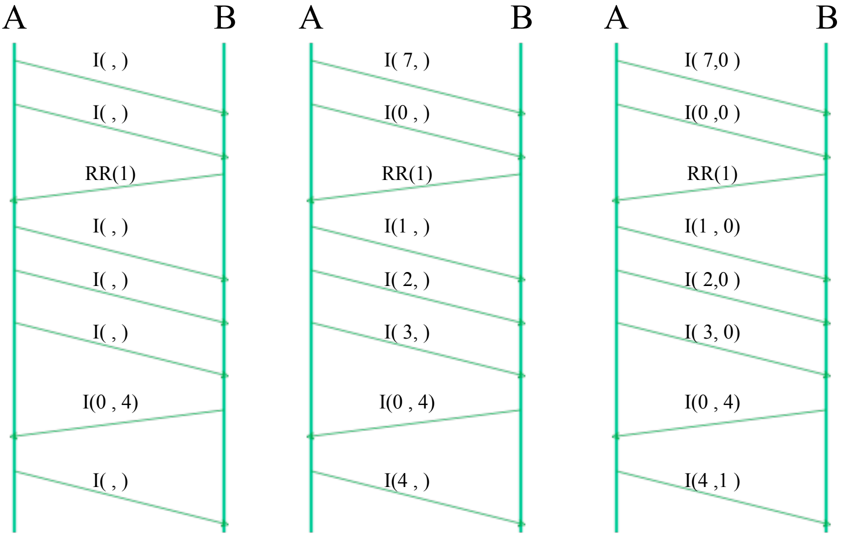
$$16 = W \leq 2^{k-1}$$

- Therefore, the minimum number of bit for sequencing is **5**

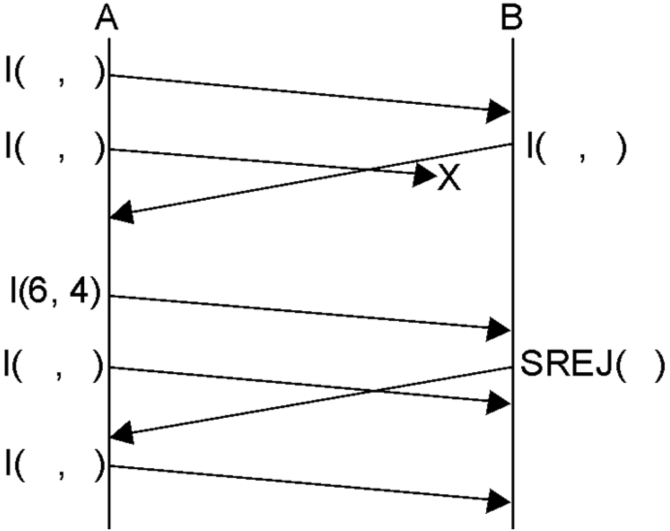
Question 4a: HDLC



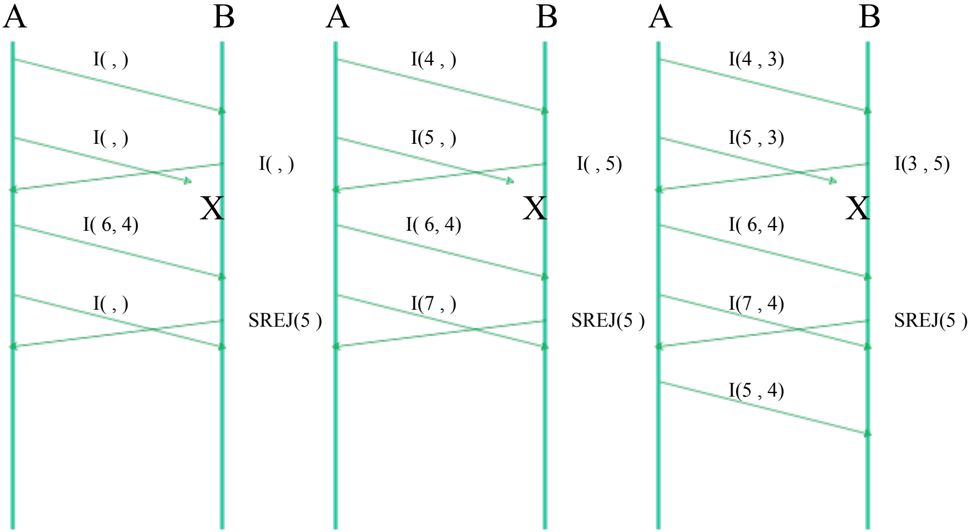
Question 4(a)


A
B
I(,)
I(,)
RR(1)
I(,)
I(,)
I(,)
I(0 , 4)
I(,)
A
B
I(7,)
I(0 ,)
RR(1)
I(1 ,)
I(2,)
I(3,)
I(0 , 4)
I(4 ,)
A
B
I(7,0)
I(0 ,0)
RR(1)
I(1 , 0)
I(2,0)
I(3, 0)
I(0 , 4)
I(4 ,1)

Question 4b: HDLC



Question 4(b)



A
B
I(,)
I(,)
I(,)
X
I(6, 4)
I(,)
SREJ(5)
A
B
I(4 ,)
I(5 ,)
I(, 5)
X
I(6, 4)
I(7 ,)
SREJ(5)
A
B
I(4 , 3)
I(5 , 3)
I(3 , 5)
X
I(6, 4)
I(7 , 4)
SREJ(5)
I(5 , 4)

In addition to the office hours listed in Lecture Note 1, please feel free to contact Assistant Professor Jun ZHAO as follows to schedule appointments to ask questions. Thanks!

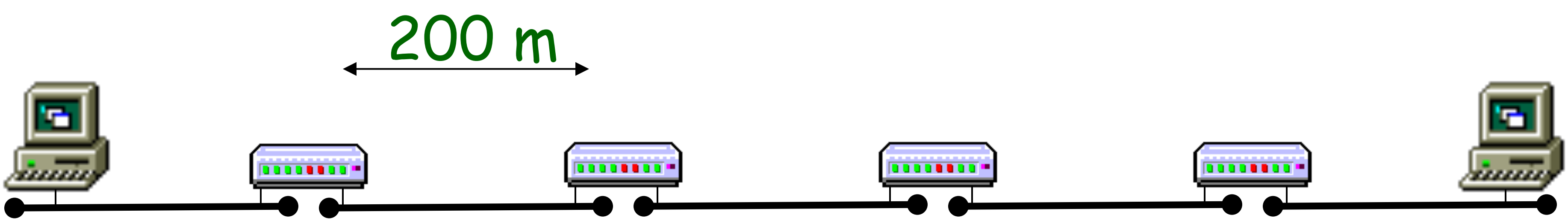
WhatsApp: http://personal.ntu.edu.sg/JunZhao/whatsapp.png
WeChat: https://personal.ntu.edu.sg/JunZhao/wechat.png
Singapore Phone Number (WhatsApp): 8648 3534
Email or Microsoft Teams: JunZhao@ntu.edu.sg
Skype ID: live:junzhaocmu
Office: Block N4, Room 02C-111, 50 Nanyang Ave, Singapore 639798
Homepage: http://personal.ntu.edu.sg/JunZhao/

CZ3006

Part I - Tutorial 3

Q1: Ethernet: Minimum Frame Size Requirement

A round trip signal propagation time + the processing time is the minimum transmission requirement for a frame transmission to ensure a proper detection of a collision.

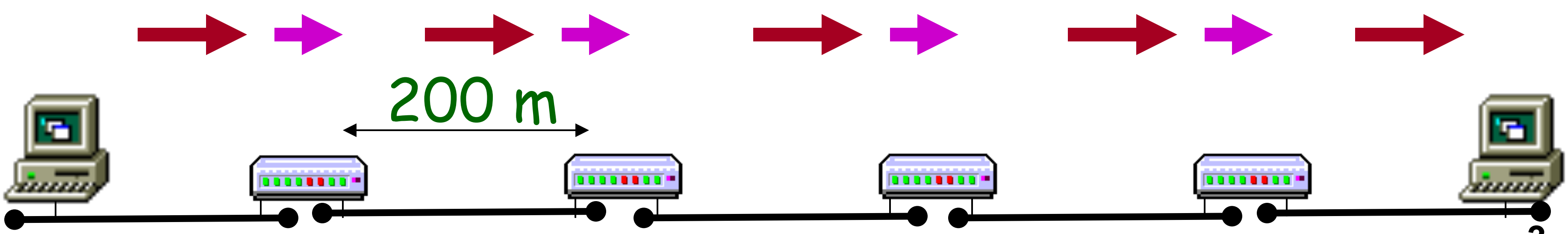


We must consider the "worst" case.

Q1: Minimum Frame Size

Minimum frame size (tx. time)
= 2 x end-to-end signal propagation time
+ processing time (negligible)

End-to-end signal propagation time
= 5 x delay in segment + 4 x delay in each repeater



Q1: Minimum Frame Size

Minimum frame size (tx. time) 36us

= 2 x end-to-end signal propagation time

+ processing time (negligible)

End-to-end signal propagation time

$$= \underbrace{5 \times \text{delay in segment}}_{10\text{us}} + \underbrace{4 \times \text{delay in each repeater}}_{8\text{us}}$$

2us 2us

- Given that signal prop. time is 1us for every 100m
  Delay per segment = 200 / 100 = 2 us
- Given that the delay in each repeater = 2 us

Q1: Minimum Frame Size

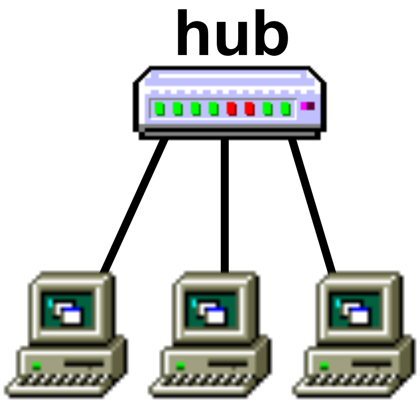
Minimum frame size (tx. time) = 36 us

at 20Mb/s, for a time period of 36 us, the number of bits can be transmitted is:

36us x 20Mb/s = 720 bits

Q2:Binary Exponential Backoff (BEB)

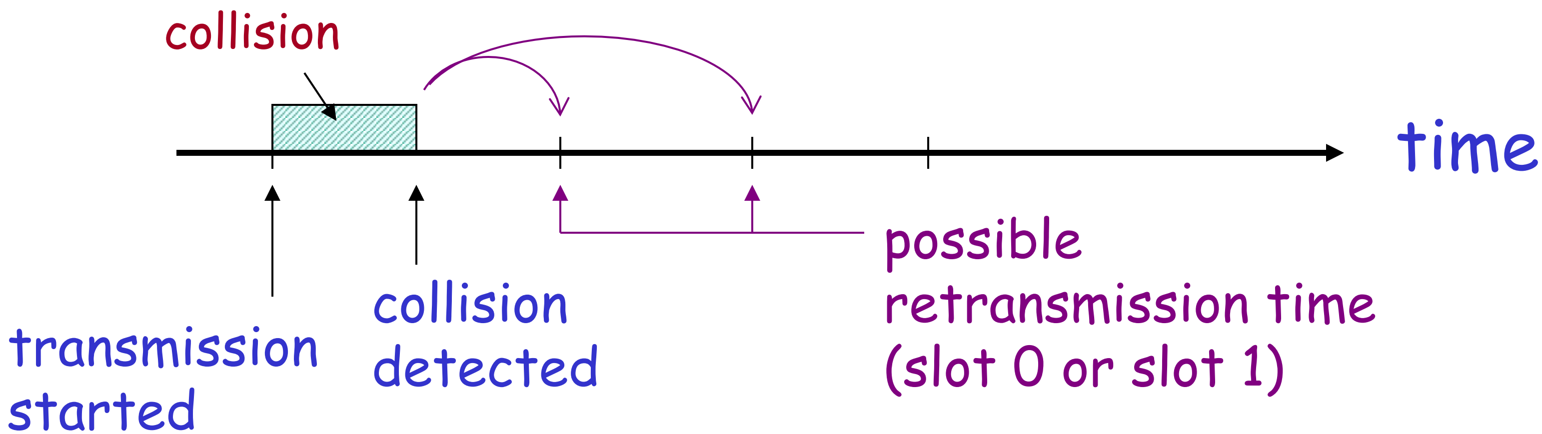
PROBLEM:
Three stations involved in a collision in an attempt to access an Ethernet LAN. Calculate the probability that the next event on the channel is also a collision.



Next event can be either:
- a successful transmission

or
- a collision.

Q2:Binary Exponential Backoff (BEB)



Probability that the next event is also a collision:

= Prob (All choose slot 0 or 1) + Prob (Two choose slot 0)

= Prob (A=0, B=0, C=0) + Prob (A=1, B=1, C=1)

+ Prob (A=1, B=0, C=0) + Prob (A=0, B=1, C=0)

+ Prob (A=0, B=0, C=1)

= (0.5 x 0.5 x 0.5) x 5 (since A, B, & C are independent)

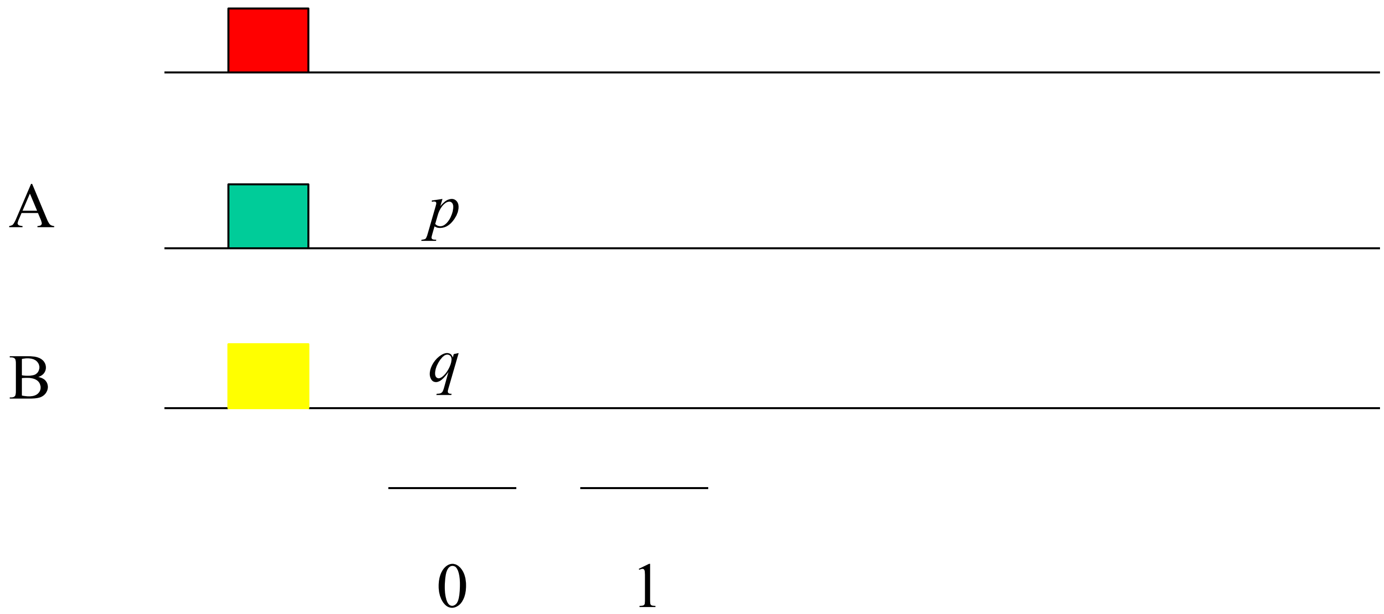
= 0.625

Question 3

In a local area network using the CSMA/CD protocol, a modified Binary Exponential Backoff scheme is used if a collision is detected in the channel. Assume that two stations (A and B) are transmitting and their frames collide in one time slot. Each of them will retransmit its data frame over a window of size 2 slots. Station A retransmits in slot 0 with probability of p and station B retransmits in slot 0 with probability of q.

- If $p = 1/3$ and $q = 2/3$, what is the probability that the first event in the channel will be a success?
- How would you maximize the probability that the first event in the channel will be a success, by choosing proper values for p and q?

Q3 answer



Q3(i) answer

	A	B	Probability	Success?
	0	0	pq	
	0	1	$p(1\text{-}q)$	Yes
	1	0	$(1\text{-}p)q$	Yes
	1	1	$(1\text{-}p)(1\text{-}q)$	

$p = 1/3$, $q = 2/3$

Pr(first event is success) = p(1-q)+(1-p)q

= 1/3*1/3 + 2/3 *2/3

= 5/9

Q3(ii) answer

	A	B	Probability	Success?
	0	0	pq	
	0	1	$p(1-q)$	Yes
	1	0	$(1-p)q$	Yes
	1	1	$(1-p)(1-q)$	

Pr(first event is success) = p(1-q)+(1-p)q
minimize $pq + (1-p)(1-q)$ is achieved when
$pq = 0$ *and* $(1-p)(1-q) = 0$

To maximize the throughput, one can choose
(i) $p = 0$, $q = 1$, or
(ii) $p = 1$, $q = 0$

Question 4

Suppose that an 11-Mbps 802.11b WLAN is transmitting 64-byte frames back-to-back over a radio channel with a bit error rate of 10^{-7}. How many frames per second will be damaged on average?

Q4 Answer

A frame contains 512 bits. The bit error rate is $p = 10^{-7}$. The probability of all 512 of them surviving correctly is $(1 - p)^{512}$, which is about 0.9999488. The fraction damaged is thus about 5×10^{-5}. The number of frames per second is $11 \times 10^{6} / 512$ or about 21,484. Multiplying these two numbers together, we get about 1 damaged frame per second.

Taylor expansion:

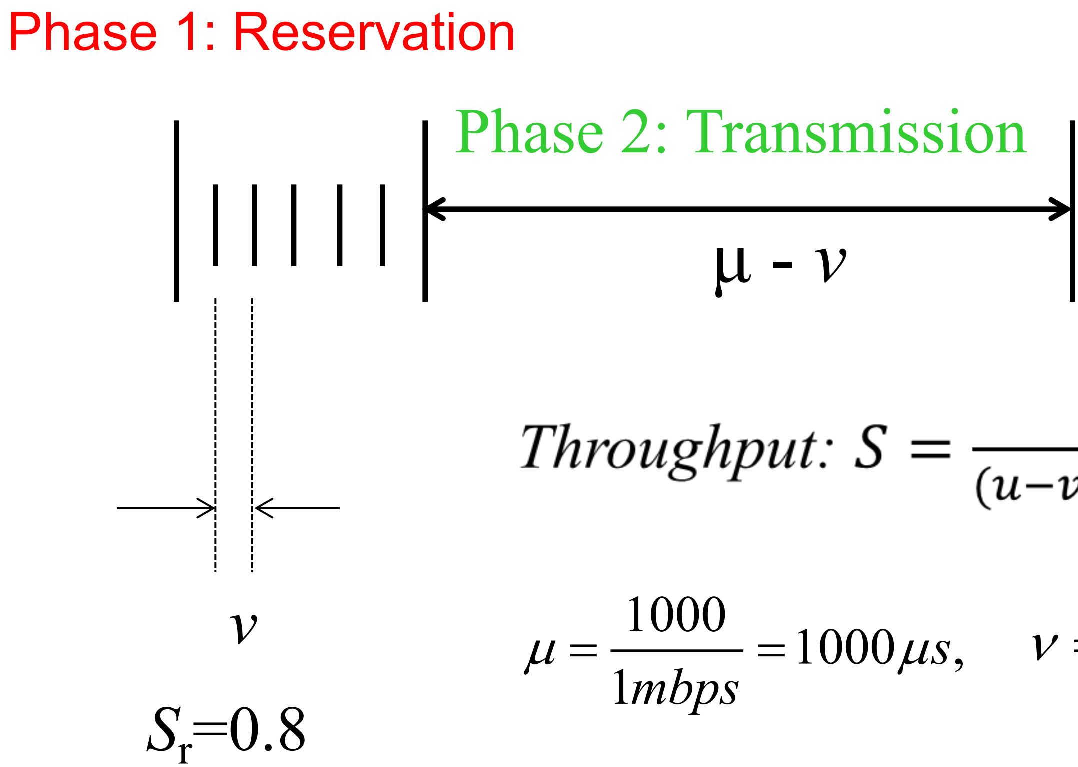
$(1 - p)^{512} \approx 1\text{-}p*512 = 1\text{-}\ 10^{-7} *512 = 0.9999488$

Question 5

You are commissioned to design an experimental wireless network for SCE to support its 100 wireless devices. You have decided to adopt a multi-access reservation protocol (MARP) for frame transmission in the data link layer. Specifically, each transmission cycle consists of two phases: a reservation phase and a transmission phase. In the reservation phase, a chosen MAC protocol is used for transmission stations to reserve the channel; and in the transmission phase, the station that successfully reserves the channel transmits one frame. The data rate in the wireless channel is 1 *Mbps*. The length of the data frame is 1000 *bits*, among which the reservation frame carries 10 information *bits*.

Q5(i) Answer

(i) If the MAC protocol used in the reservation phase has a utilization of 0.8, what will be the throughput of the MARP?



$$Throughput: \; S = \frac{u}{(u-v)+{}^{v}/_{S_r}}$$

$$\mu = \frac{1000}{1mbps} = 1000\,\mu s, \qquad \nu = \frac{10}{1mbps} = 10\,\mu s$$

$$S = \frac{1000}{990 + {}^{10}/_{0.8}} = \frac{1000}{1002.5} = 99.75\%$$

Q5(ii) Answer

(ii) Assume that the slotted Aloha protocol is used in the reservation phase. The utilization for the slotted Aloha protocol is Ge^{-G}, where $G = np$, n is the number of stations, and p is transmission probability. Calculate an optimal transmission probability to maximize the throughput of the MARP, and the corresponding maximum throughput?

In order to maximize the utilization, it is equivalent to maximize $S_r = Ge^{-G}$.

$$\frac{dS_r}{dG} = e^{-G} - Ge^{-G} = 0 \Rightarrow G^* = 1$$

In this case, $p^* = \dfrac{G^*}{n} = 1/100 = 0.01$,

$$S_{r,\max} = 1/e$$

The maximum utilization is derived as

$$S_{\max} = \frac{\mu}{(\mu - v) + {}^{v}\!/\!_{S_{r,\max}}} = \frac{1000}{(1000-10)+10e} = 98.31\%$$

In addition to the office hours listed in the previous slide, please feel free to contact Assistant Professor Jun ZHAO as follows to schedule appointments to ask questions. Thanks!

WhatsApp: http://personal.ntu.edu.sg/JunZhao/whatsapp.png
WeChat: https://personal.ntu.edu.sg/JunZhao/wechat.png
Singapore Phone Number (WhatsApp): 8648 3534
Email or Microsoft Teams: JunZhao@ntu.edu.sg
Skype ID: live:junzhaocmu
Office: Block N4, Room 02C-111, 50 Nanyang Ave, Singapore 639798
Homepage: http://personal.ntu.edu.sg/JunZhao/

Tutorial 4

Jun ZHAO

Assistant Professor

School of Computer Science and Engineering

Nanyang Technological University

WhatsApp: http://personal.ntu.edu.sg/JunZhao/whatsapp.png
WeChat: https://personal.ntu.edu.sg/JunZhao/wechat.png
Singapore Phone Number (WhatsApp): 8648 3534
Email or Microsoft Teams: JunZhao@ntu.edu.sg

1. What is the principal difference between connectionless communication and connection-oriented communication?

Question 1's Answer:

- Connection-oriented communication has three phases. In the establishment phase a request is made to set up a connection. Only after this phase has been successfully completed can the data transfer phase be started and data transported. Then comes the release phase.

- Connectionless communication does not have these phases. It just sends the data.

2. *Packet switched networks route each packet as a separate unit, independent of all others. Virtual-circuit networks do not have to do this, since each data packet follows a predetermined route. Does this observation mean that virtual-circuit networks do not need the capability to route isolated packets from an arbitrary source to an arbitrary destination? Explain your answer.*

Knowledge point:

Packet switched: connectionless (just like postmail)

Virtual circuit: connection-oriented (just like phone communication)

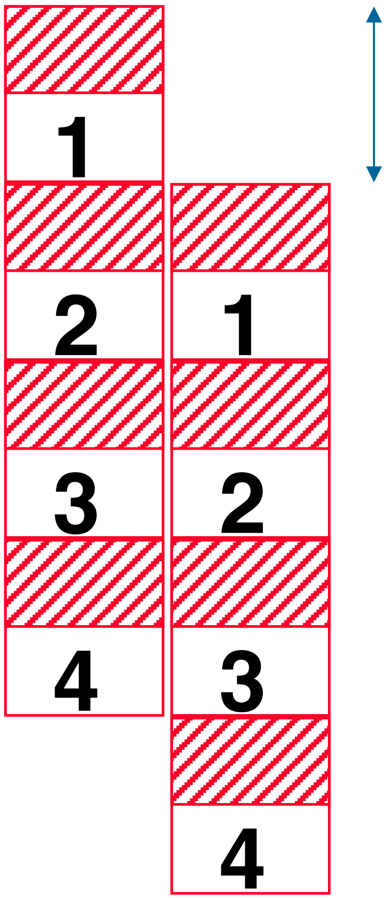
Question 2's Answer:

Virtual circuit subnets most certainly need this capability in order to route **connection setup packets** from an arbitrary source to an arbitrary destination.

Question 3:

Consider a packet switched network. Two nodes, node S and node D, are connected through an intermediate node I. A message of size 1000 bytes is transmitted from node S to node D. The message is fragmented into four packets each with a 50-byte header. All links run the same data rate. If propagation delay is negligible, determine the minimum data rate of the links to achieve 100ms of total transmission delay. (Hint: *pipeline effect*)

Question 3's Answer:



Transmission time, T_f
(frame size = 250 + 50 bytes)

Let:

- **d: the link data rate (to determine)**
- **T_f: transmission time of a packet**
- **T: The total transmission time**

The last bit is sent at time $4T_f$. To get to the destination, the last packet must be retransmitted by the intermediate router I. The retransmission takes time T_f. Then the total delay is $T = 5T_f$

According the question, we know that T<100ms

so, $5T_f$ < 100ms - - - (1)

Since T_f = (250 bytes + 50 bytes) * 8 / d - - - (2)

By (1) & (2), d > 120 kbps

4. A factor in the delay of a store-and-forward packet-switching system is how long it takes to store and forward a packet through a switch.

   If switching time is 10 μsec, is this likely to be a major factor in the response of a client-server system where the client is in New York and the server is in California?

   Assume the propagation speed in copper and fiber to be 2/3 the speed of light in vacuum.

Question 4's Answer:

The speed of propagation is 200,000 km/sec, i.e., 200 meters/μsec. In 10 μsec the signal travels 2 km.

Thus, each switch adds the equivalent of 2 km of extra cable.

So the questions are:

Where is New York and California?

How many switches the signal have to travel?

If the client and server are separated by 4000 km (e.g., from New York to San Francisco), traversing even 50 switches adds only 100 km to the total path, which is only 2.5%.

Thus, switching delay is not a major factor under this circumstance.


≈ 4000 km
San Francisco
New York
0
500.00 km
1,000.00 km
1,500.00 km
2,000.00 km
2,500.00 km
3,000.00 km
3,500.00 km
4,128.94 km
United States
Vancouver
Seattle
Portland
Sacramento
Las Vegas
Los Angeles
San Diego
Phoenix
Tucson
El Paso
Dallas
Austin
San Antonio
Houston
New Orleans
Kansas City
Chicago
Detroit
Toronto
Ottawa
Montreal
Indianapolis
Nashville
Atlanta
Charlotte
Washington
Jacksonville
Orlando
Miami
Havana
Monterrey
WASHINGTON
OREGON
IDAHO
MONTANA
WYOMING
NEVADA
UTAH
COLORADO
CALIFORNIA
ARIZONA
NEW MEXICO
NORTH DAKOTA
SOUTH DAKOTA
MINNESOTA
IOWA
KANSAS
OKLAHOMA
TEXAS
MISSOURI
ARKANSAS
LOUISIANA
MISSISSIPPI
WISCONSIN
MICHIGAN
ILLINOIS
INDIANA
OHIO
KENTUCKY
TENNESSEE
GEORGIA
FLORIDA
VIRGINIA
NORTH CAROLINA
SOUTH CAROLINA
NEW YORK
CT
DE
NEW HAMPSHIRE
BAJA CALIFORNIA
SONORA
CHIHUAHUA
COAHUILA
NUEVO LEON
Gulf of California
Gulf of Mexico

Why assuming 50 switches for the signal to travel from New York to San Francisco?

- Maximal number of hops allowed:
  64 in MacBook’s Operating System (MacOS),
  255 in Windows Operating System.

- Even we assume 100 switches, switching delay is just 5% (i.e., not a major factor) of the path delay.

5. Compare the delay in sending an x-bit message over a k-hop path in a circuit-switched network and in a (lightly loaded) packet-switched network.

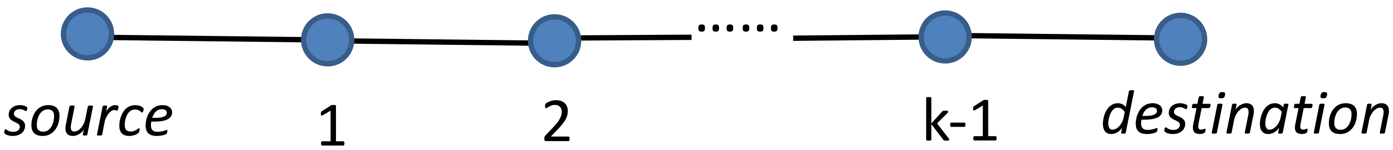
   The circuit set up time is s sec,
   the propagation delay is d sec per hop,
   the packet size is p bits,
   and the data rate is b bps.

   Under what conditions does the packet network have a lower delay?

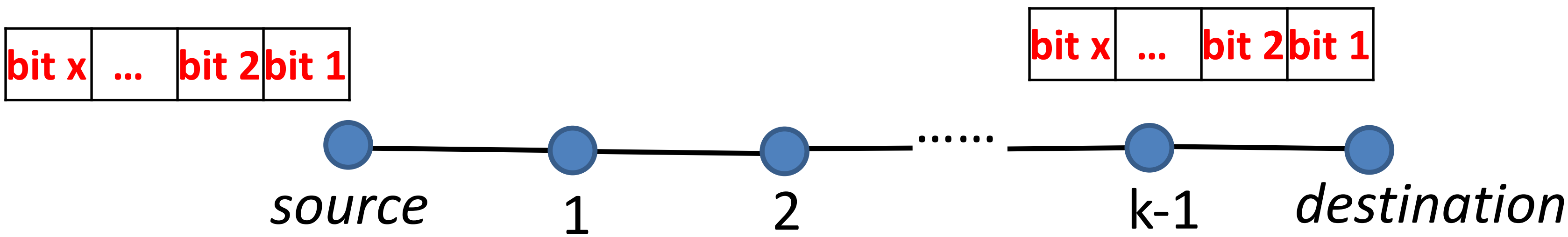
Question 5's Answer —— Slide 1:

With circuit switching,

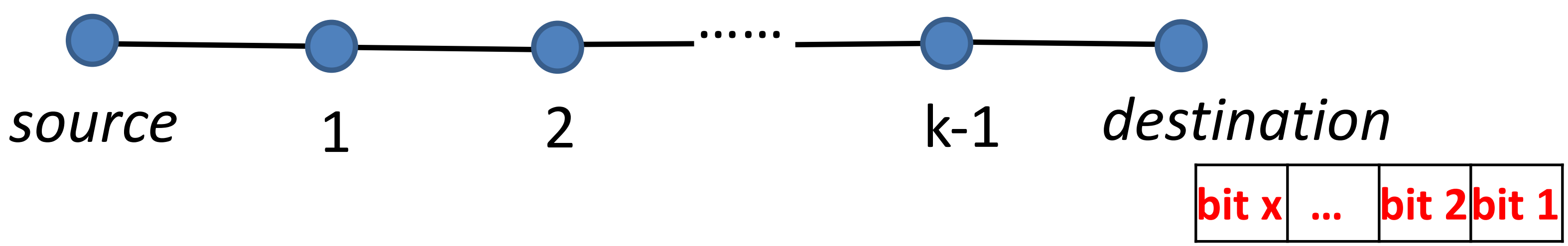
- After *s* time, the circuit is set up.



- After *kd* more time, the first bit is received by destination.



- After *x/b* more time, the last bit is received by destination.

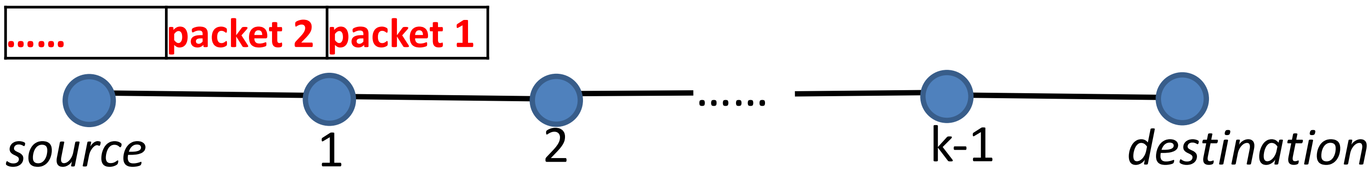


- Total delay: $s + x/b + kd$

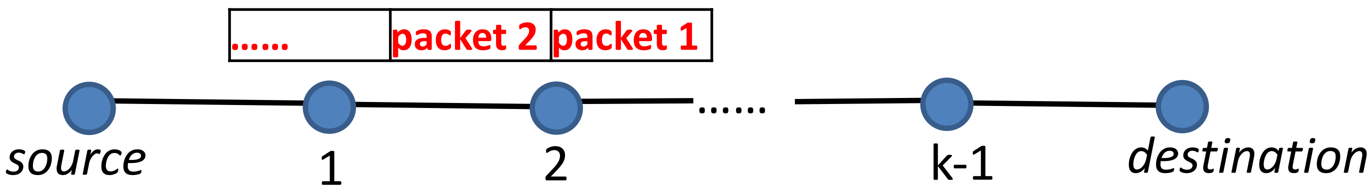
Question 5's Answer —— Slide 2:

With packet switching,

- After *p/b+d* time, the 1st packet is sent to the 1st hop.

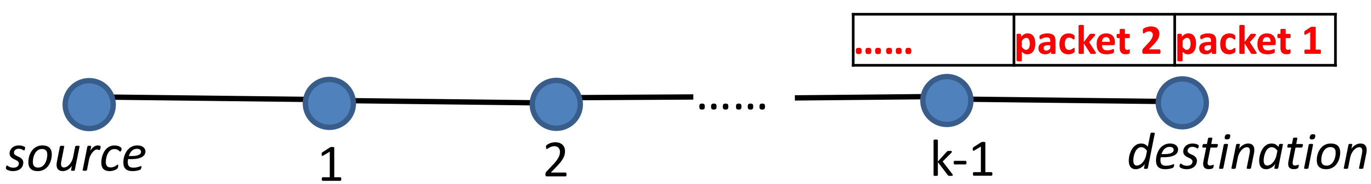


- After *p/b+d* more time, the 1st packet is sent to the 2nd hop.

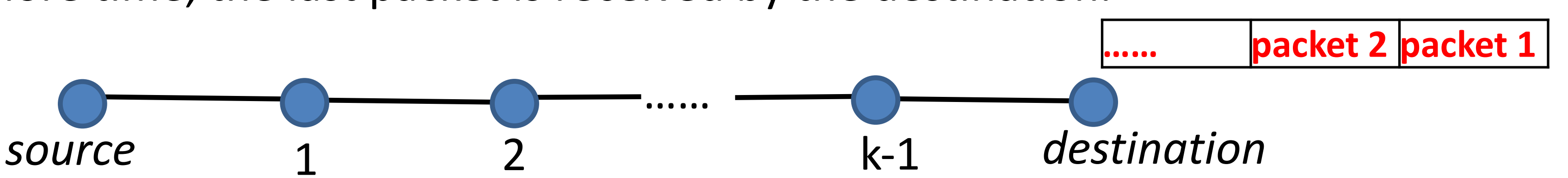


- ......

- Starting from the instant that the 1st packet is sent to the (k-1)th hop, after *p/b+d* more time, the 1st packet is sent to the kth hop (i.e., the destination).



- Starting from the instant that the 1st packet is sent to the destination, after $(x\text{-}p)/b$ more time, the last packet is received by the destination.



- Total delay: $k * (p/b+d) + (x\text{-}p)/b = x/b + (k-1)p/b + kd.$

Question 5's Answer —— Slide 3:

- With circuit switching, the total delay is
  $s + x/b + kd$.

- With packet switching, the total delay is
  $x/b + (k - 1)p/b + kd$.

- The total delay with packet switching is smaller
  if $x/b + (k - 1)p/b + kd < s + x/b + kd$;
  i.e., $(k - 1)p/b < s$.

In addition to the office hours listed in the previous slide, please feel free to contact Assistant Professor Jun ZHAO as follows to schedule appointments to ask questions. Thanks!

WhatsApp: http://personal.ntu.edu.sg/JunZhao/whatsapp.png
WeChat: https://personal.ntu.edu.sg/JunZhao/wechat.png
Singapore Phone Number (WhatsApp): 8648 3534
Email or Microsoft Teams: JunZhao@ntu.edu.sg
Skype ID: live:junzhaocmu
Office: Block N4, Room 02C-111, 50 Nanyang Ave, Singapore 639798
Homepage: http://personal.ntu.edu.sg/JunZhao/

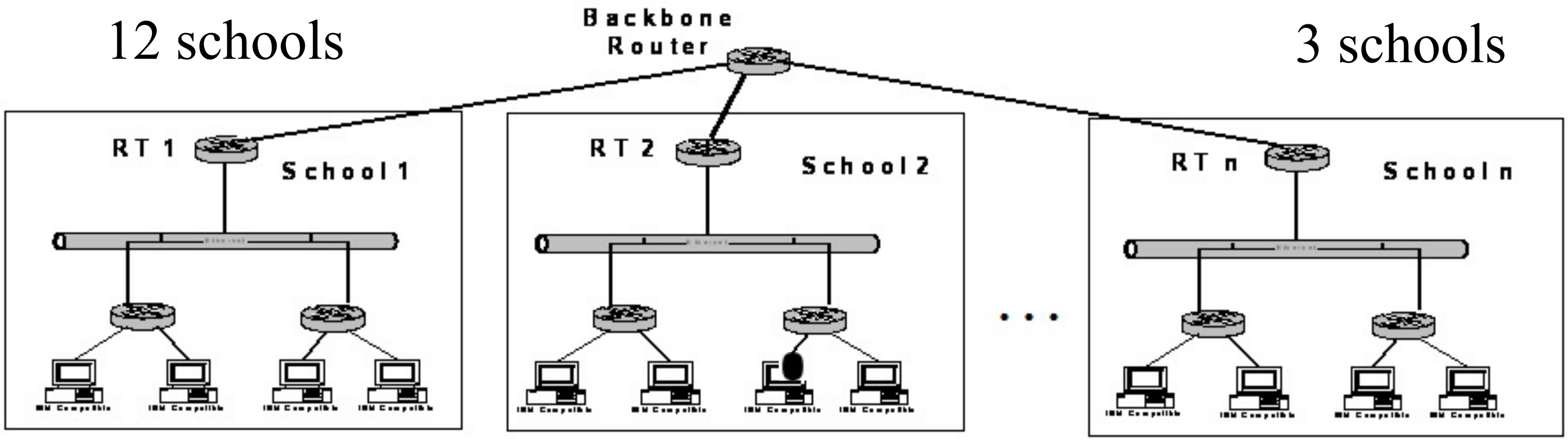
SC2008
CE3005:Computer Networks
CZ3006: Netcentric Computing

IP addressing

Q1: Assign suitable subnet address/subnet mask

12 schools

3 schools



2000 hosts

2000

4000

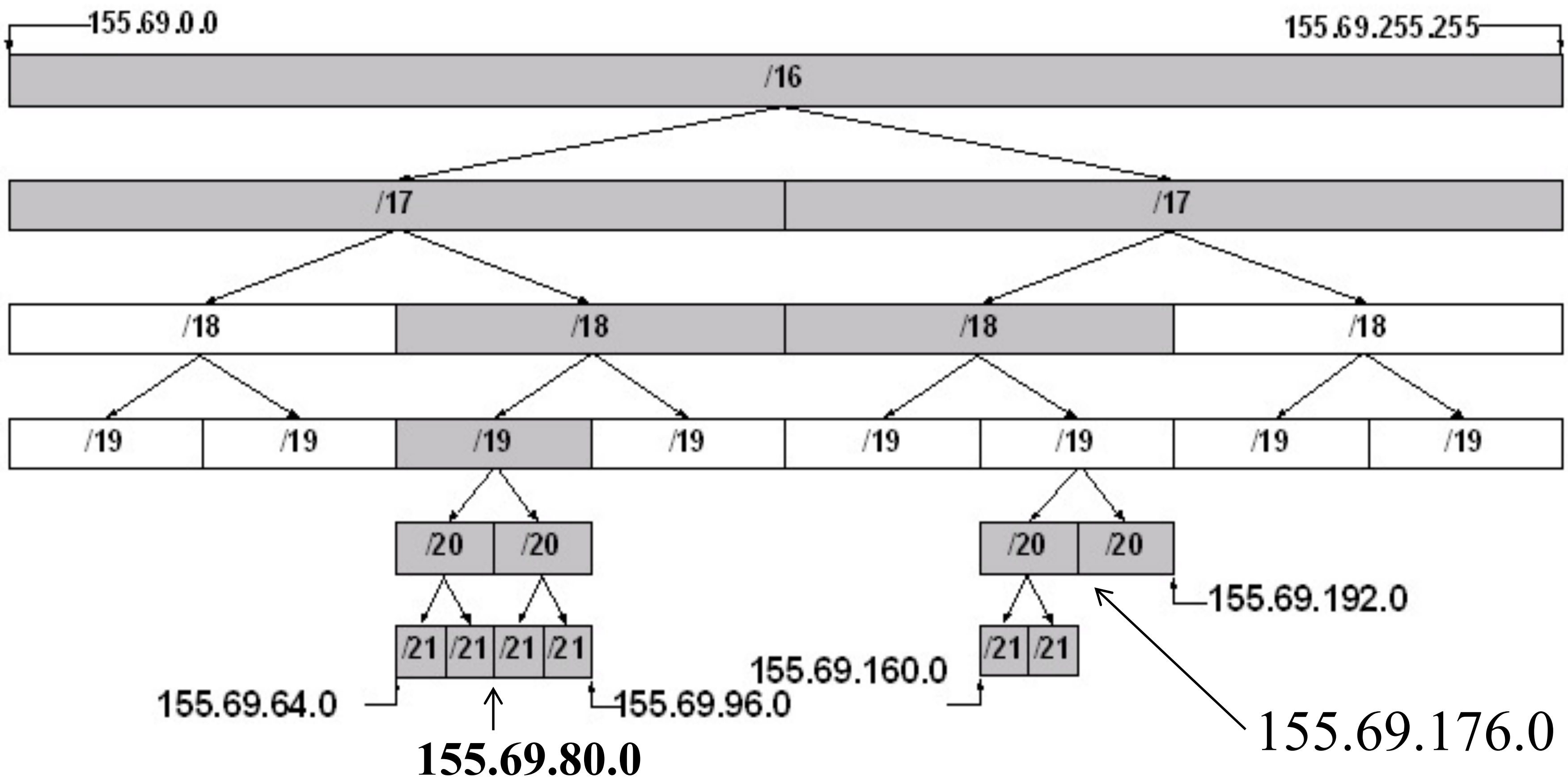
Note:

IP block: 155.69.0.0/16

If using /20 mask, # of hosts = 2^{12} - 2 = 4094

If using /21 mask, # of hosts = 2^{11} - 2= 2046

Q1: Assign suitable subnet address/subnet mask



In this /16 network, there can be 16 subnets with /20 masks, or 32 subnets with /21 masks.

Subnet mask :/20 255.255.11110000.0

Q1: Assign suitable subnet address/subnet mask

- You can choose
  - any 3 address blocks with /20, e.g.
    - 155.69.0.0/20 : 155.69.0.0 till 155.69.15.255
    - 155.69.16.0/20 :155.69.16.0 till 155.69.31.255
    - 155.69.32.0/20 :155.69.32.0. till 155.69.47.255
  - any 12 address blocks with /21, e.g.
    - 155.69.64.0/21 : 155.69.64.0 till 155.69.71.255
    - 155.69.72.0/21 : 155.69.72.0 till 155.69.79.255
    - …
  - Remember not to overlap the address block

How much address is left ? 7 blocks of /20

Q2: Broadcast Address of a Subnet

An organization is assigned a /16 IP address block. The organization has created several subnets for its network. It is known that one of the subnets is 145.32.128.0/255.255.224.0. What is the broadcast address for this subnet?

Given subnet address/subnet mask:

Dotted decimal:	145.32.128.0	/	255.255.224.0
In binary:	145.32.10000000.0	/	255.255.11100000.0

Q2: Broadcast Address of a Subnet

Given subnet address/subnet mask:

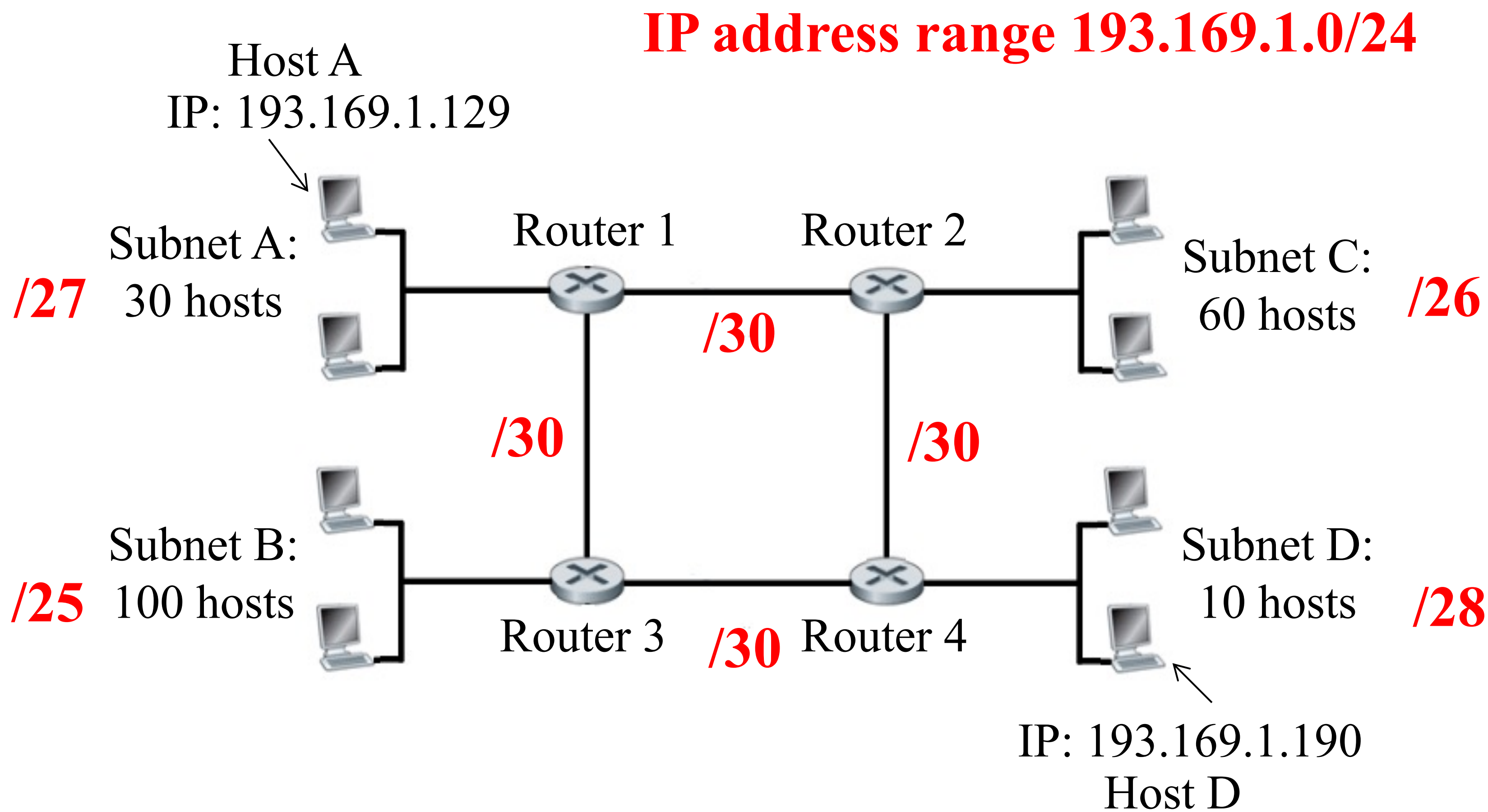
Dotted decimal:	145.32.128.0	/	255.255.224.0
In binary:	145.32.10000000.0	/	255.255.11100000.0

So, broadcast address of subnet 145.32.128.0/19:

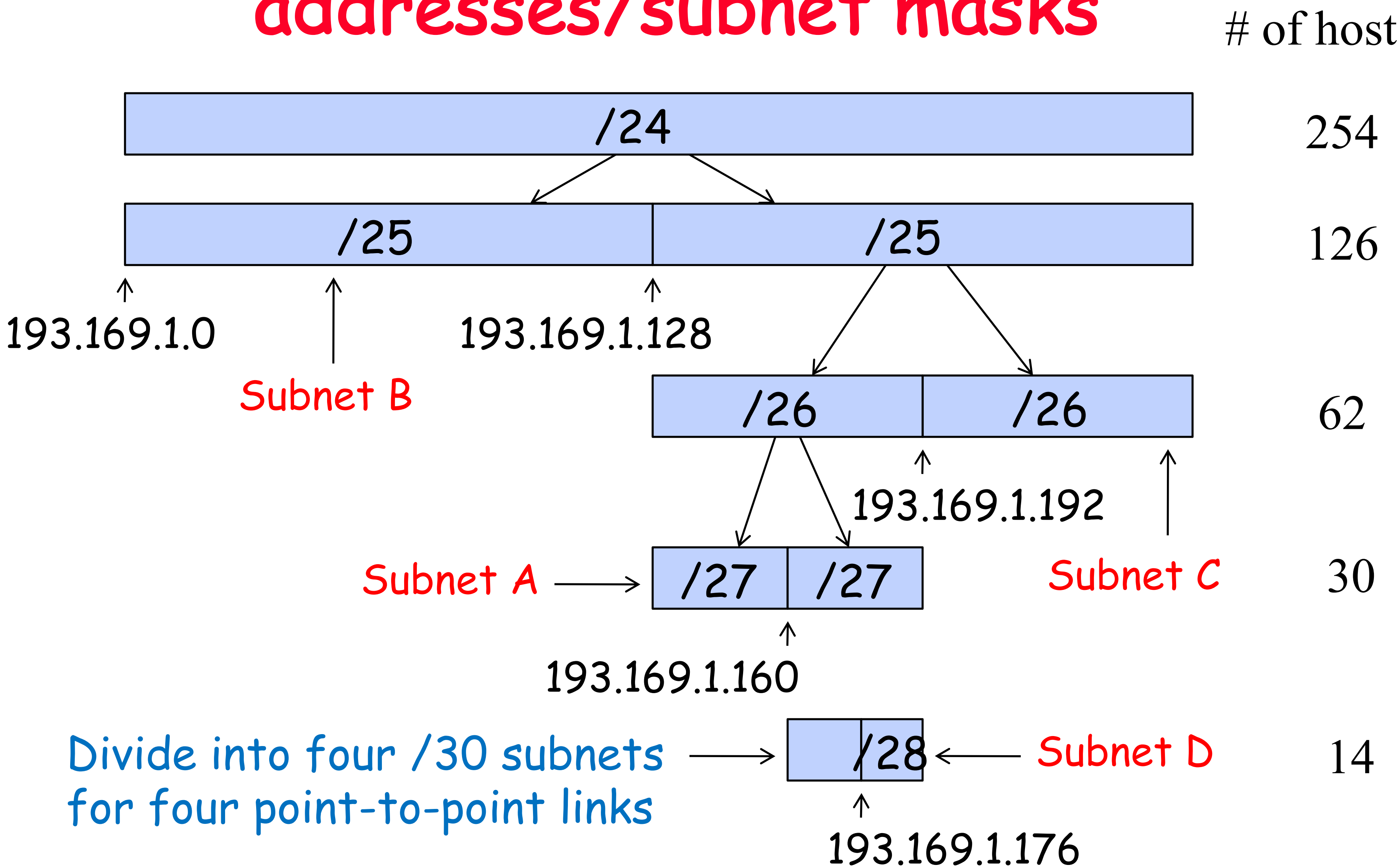
In binary:	145.32.10011111.11111111
Dotted decimal:	145.32.159.255

Q3: Assign suitable IP addresses/subnet masks

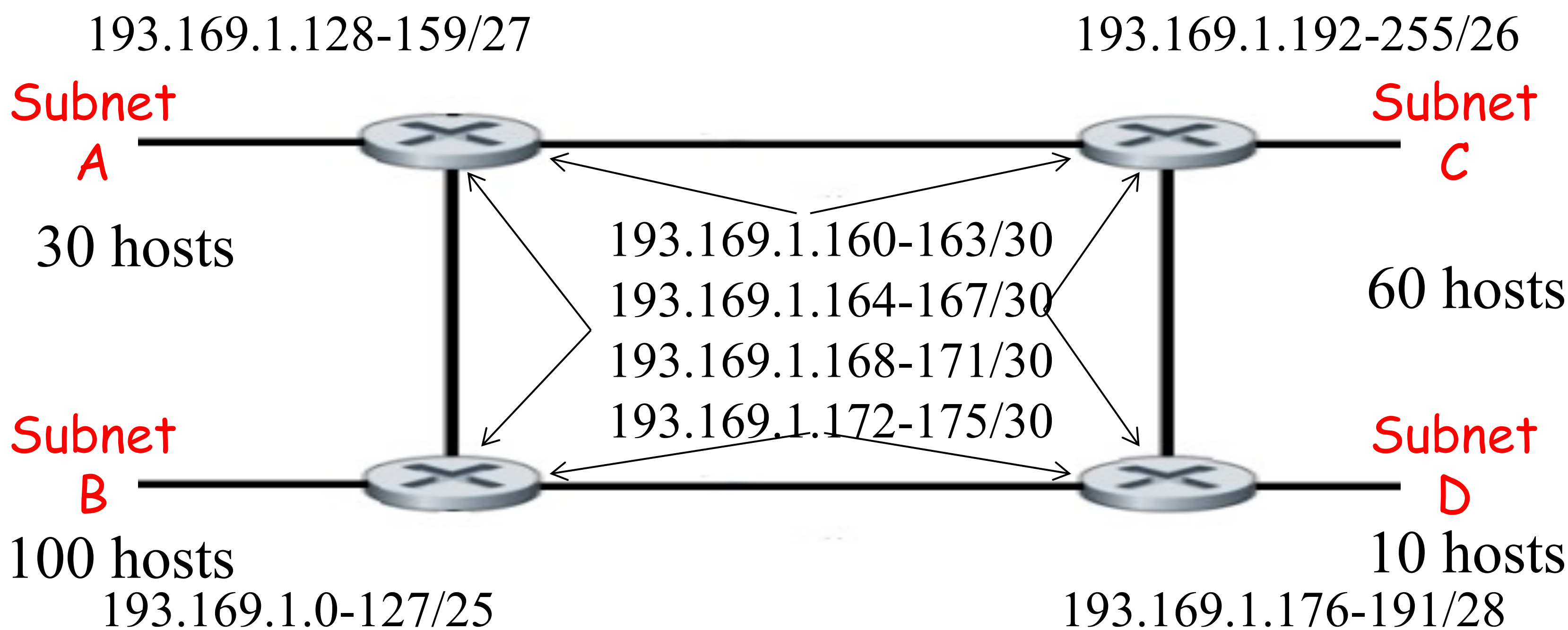
Remember that all hosts/routers in a subnet must have the same subnet id.



Q3: Assign suitable IP addresses/subnet masks



Q3: Assign suitable IP addresses/subnet masks



Q4: IP Header

Consider sending a 3000 byte datagram into a link that has an MTU of 500 bytes. Suppose the original datagram is stamped with the identification number 422. How many fragments are generated? What are their characteristics?

Q4: IP Header

Consider sending a 3000 byte datagram into a link that has an MTU of 500 bytes. Suppose the original datagram is stamped with the identification number 422. How many fragments are generated? What are their characteristics?

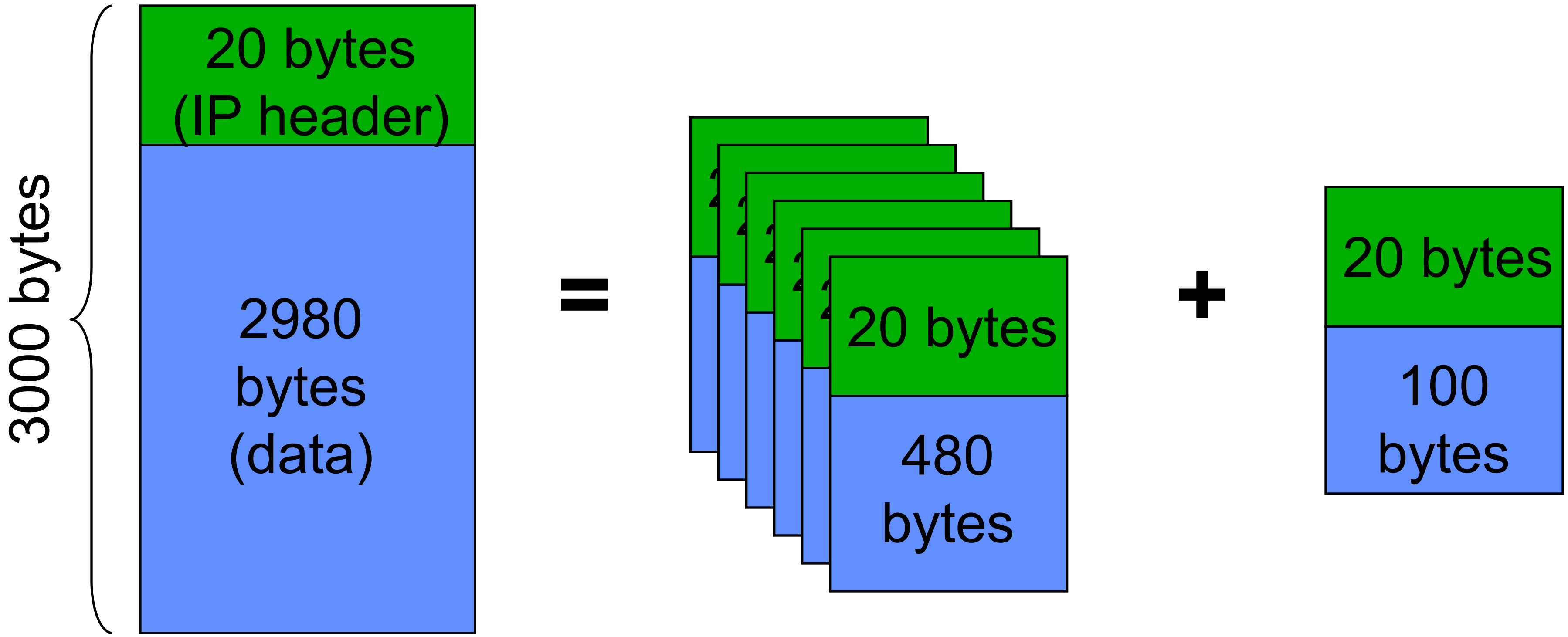
Size of the datagram = 3000 bytes

Total data in the datagram = 2980 bytes

Max. data in each fragment = 500 - 20 = 480 bytes

Number of fragments = 2980 / 480 = 6.21, i.e., 7 fragments

Q4: IP Fragmentation



Q4: IP Header

Bits 0 8 16 24 31

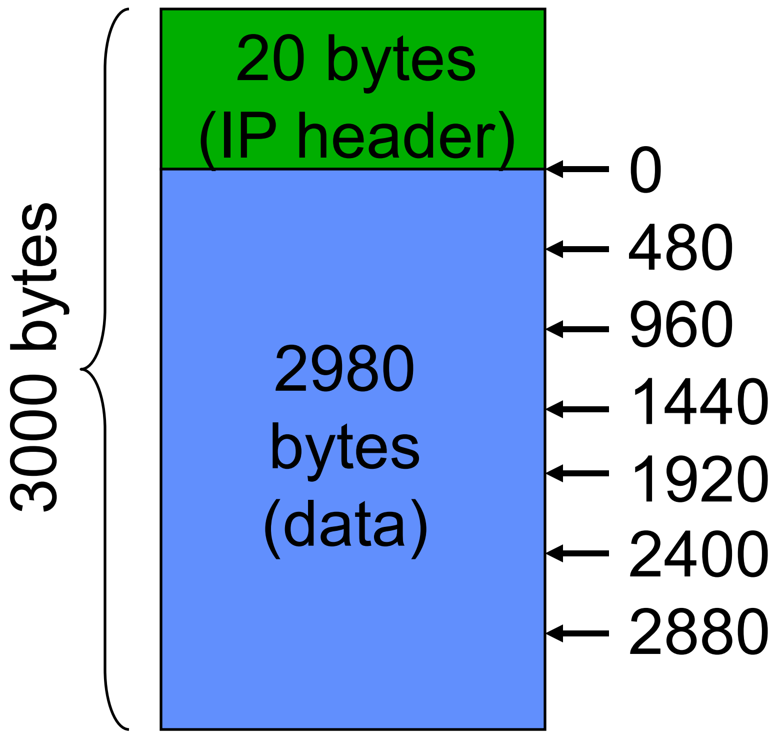
Bits 0–3	Bits 4–7	Bits 8–15	Bits 16–18	Bits 19–31
Version	IHL	Type of Service	Total Length	
Identification 422			DF MF	Fragmentation Offset
Time to Live		Protocol	Header Checksum	
Source Address				
Destination Address				
Optional Header Fields + Padding				

DF: Don't Fragment

MF: More Fragments

Bit not used

Q4: IP Fragmentation



Fragment	length	ID	MF	offset
1	500	422	1	0
2	500	422	1	60
3	500	422	1	120
4	500	422	1	180
5	500	422	1	240
6	500	422	1	300
7	120	422	0	360

‘offset’ is in 8-byte unit

(ie 60 means 60*8 = 480 bytes)

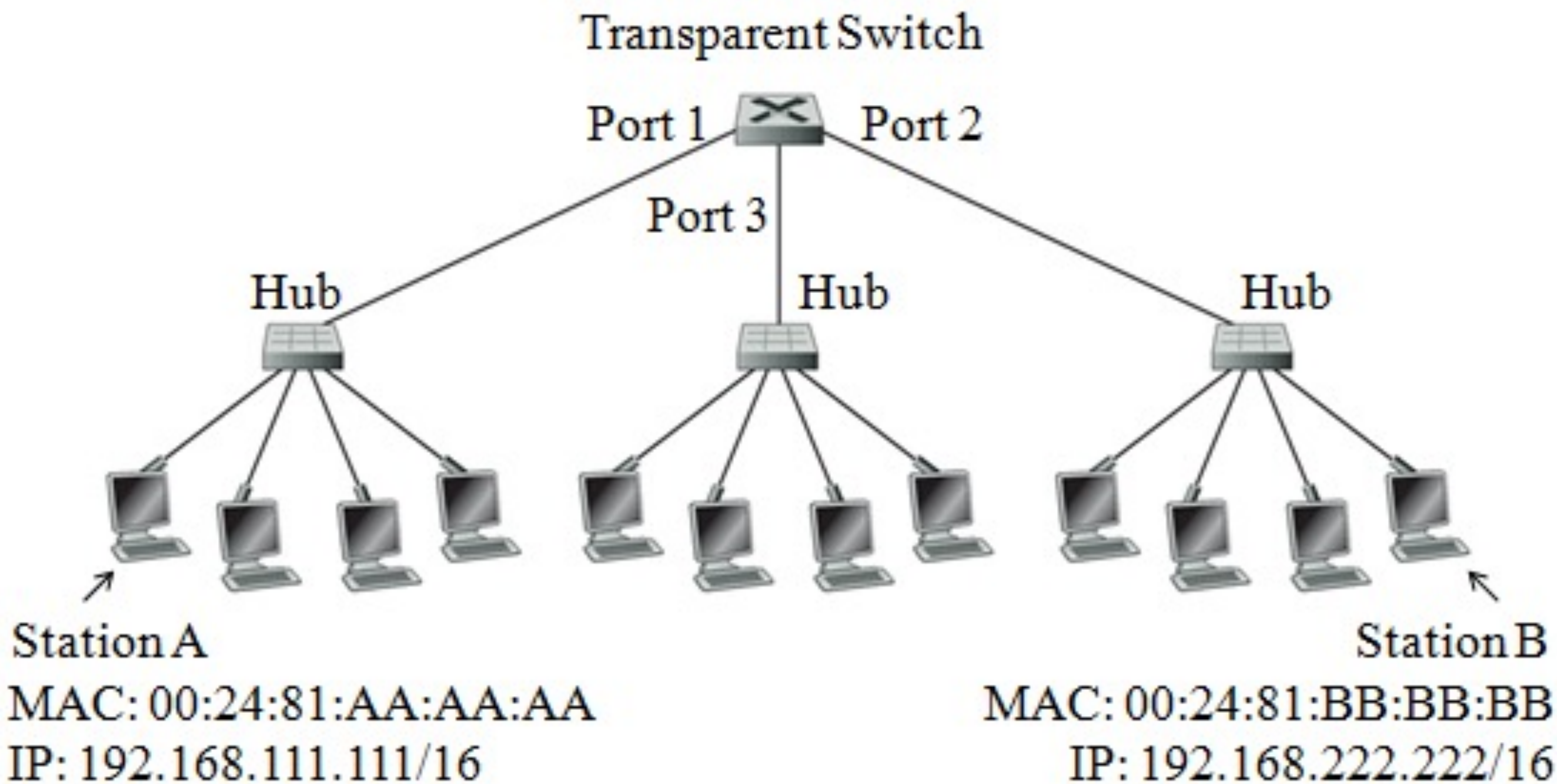
In addition to the office hours listed in the previous slide, please feel free to contact Assistant Professor Jun ZHAO as follows to schedule appointments to ask questions. Thanks!

WhatsApp: http://personal.ntu.edu.sg/JunZhao/whatsapp.png
WeChat: https://personal.ntu.edu.sg/JunZhao/wechat.png
Singapore Phone Number (WhatsApp): 8648 3534
Email or Microsoft Teams: JunZhao@ntu.edu.sg
Skype ID: live:junzhaocmu
Office: Block N4, Room 02C-111, 50 Nanyang Ave, Singapore 639798
Homepage: http://personal.ntu.edu.sg/JunZhao/

SC2008
CE3005:Computer Networks
CZ3006: Netcentric Computing

IP routing

Q1


Transparent Switch
Port 1
Port 2
Port 3
Hub
Hub
Hub
Station A
MAC: 00:24:81:AA:AA:AA
IP: 192.168.111.111/16
Station B
MAC: 00:24:81:BB:BB:BB
IP: 192.168.222.222/16

Q1: ping, ARP and switch operation

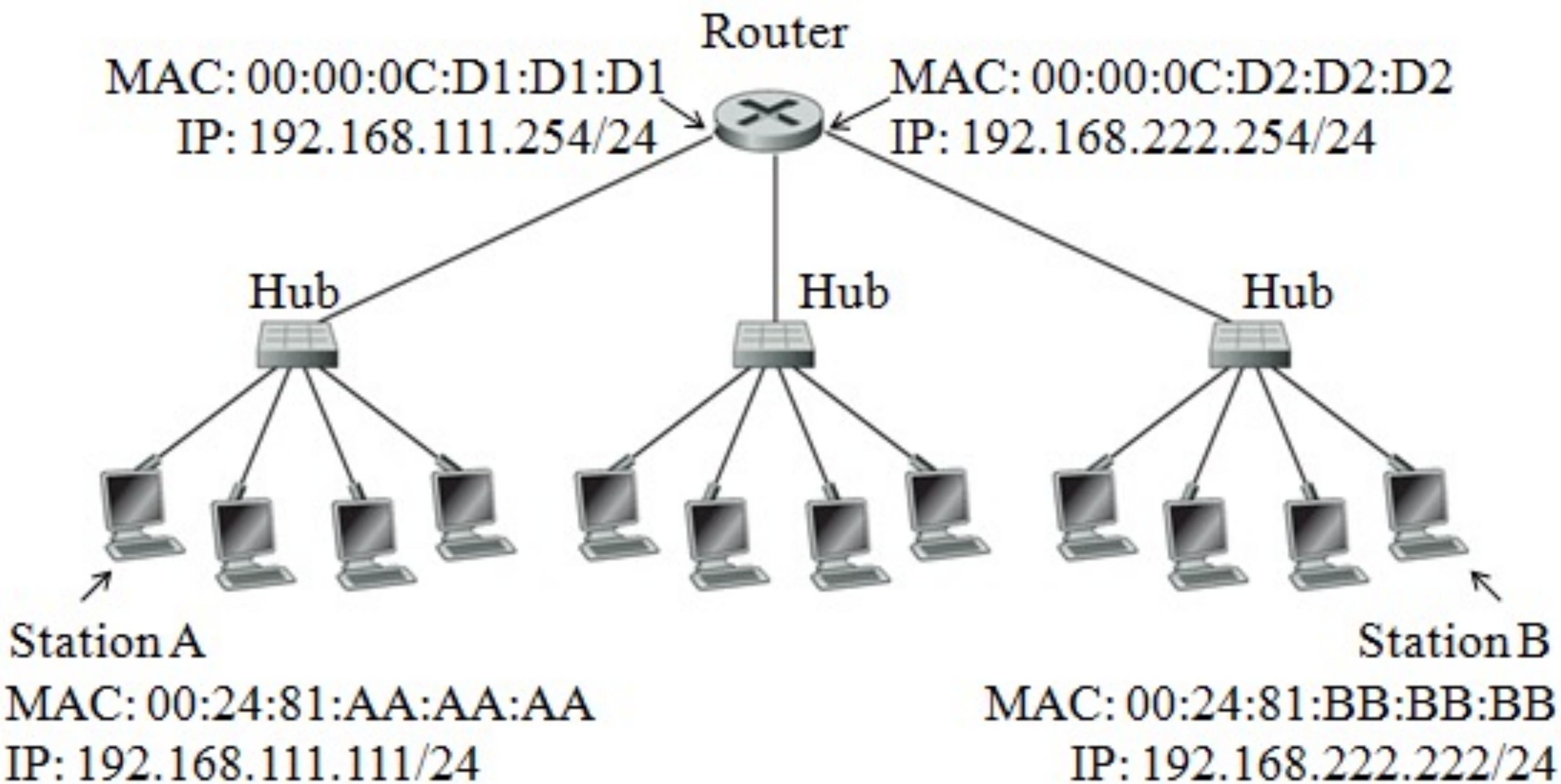
Frame	MAC Address		Purpose of Frame	Actions taken by Switch
	Source	Destination		
1.	00-24-81-AA-AA-AA	FF-FF-FF-FF-FF-FF	ARP request for 192.168.222.222	- new entry forwarding table 00-24-81-AA-AA-AA/port 1/ time - broadcast ARP request frame to both port 2, 3
2.	00-24-81-BB-BB-BB	00-24-81-AA-AA-AA	ARP reply	- new entry forwarding table 00-24-81-BB-BB-BB/port 2/ time - forward ARP reply to port 1

Q1: ping, ARP and switch operation

Frame	MAC Address		Purpose of Frame	Actions taken by Switch
	Source	Destination		
3.	00-24-81-AA-AA-AA	00-24-81-BB-BB-BB	ping request for 192.168.222.222	- update forwarding table 00-24-81-AA-AA-AA/port 1/ new time - forward ping request to port 2
4.	00-24-81-BB-BB-BB	00-24-81-AA-AA-AA	ping reply	- update forwarding table 00-24-81-BB-BB-BB/port 2/ new time - forward ping reply to port 1

For simplicity, assume ping sends 1 packet instead of 4.

Q2



Router
MAC: 00:00:0C:D1:D1:D1
IP: 192.168.111.254/24
MAC: 00:00:0C:D2:D2:D2
IP: 192.168.222.254/24
Hub
Hub
Hub
Station A
MAC: 00:24:81:AA:AA:AA
IP: 192.168.111.111/24
Station B
MAC: 00:24:81:BB:BB:BB
IP: 192.168.222.222/24

: ping, ARP and router operation

Frame	MAC Address		Purpose of Frame	Actions taken by Router
	Source	Destination		
1.	00-24-81-AA-AA-AA	FF-FF-FF-FF-FF-FF	ARP request for default gateway 192.168.111.254	- new entry ARP cache 192.168.111.111/00-24-81-AA-AA-AA
2.	00-00-0C-D1-D1-D1	00-24-81-AA-AA-AA	ARP reply	- send ARP reply

Q2: ping, ARP and router operation

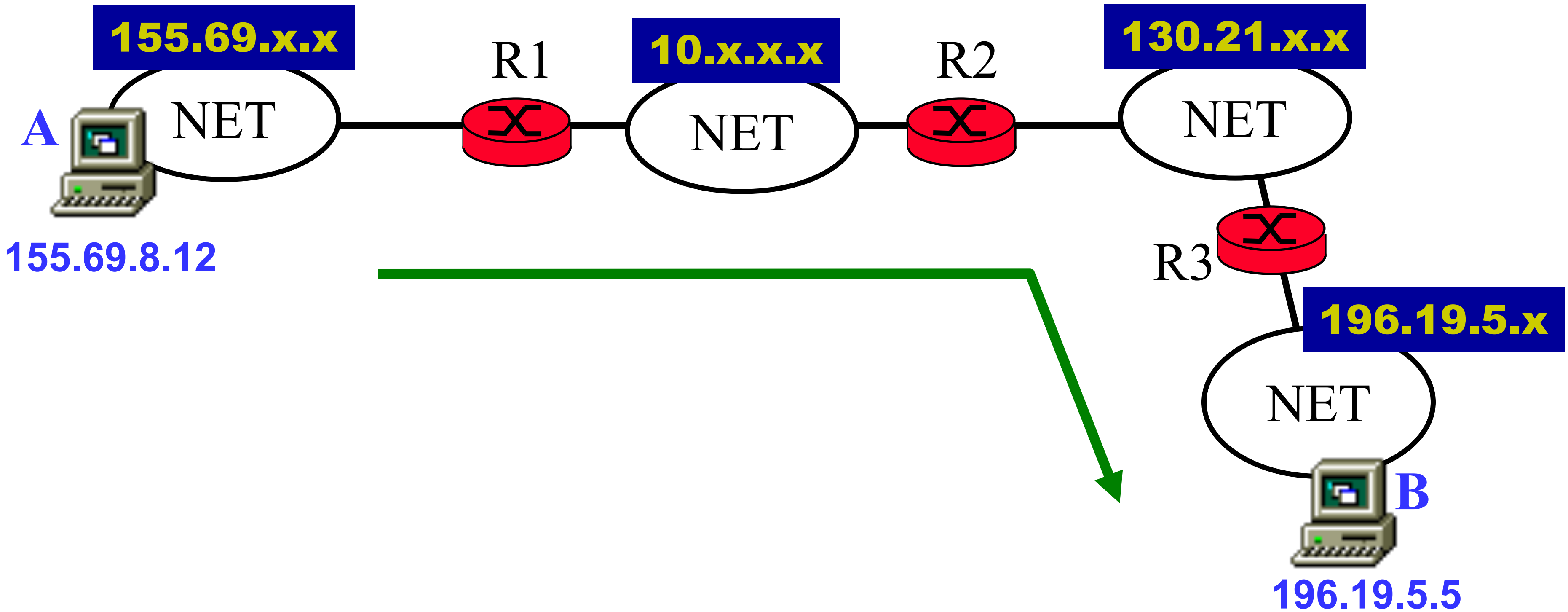
Frame	MAC Address		Purpose of Frame	Actions taken by Router
	Source	Destination		
3.	00-24-81-AA-AA-AA	00-00-0C-D1-D1-D1	ping request for 192.168.222.222	- broadcast ARP request for 192.168.222.222 at subnet 192.168.222.0/24 - receive ARP reply - new entry ARP cache 192.168.222.222/00-24-81-BB-BB-BB - forward ping request to 192.168.222.222
4.	00-00-0C-D1-D1-D1	00-24-81-AA-AA-AA	ping reply	- receive ping reply - forward ping reply to 192.168.111.111

For simplicity, assume ping sends 1 packet instead of 4.

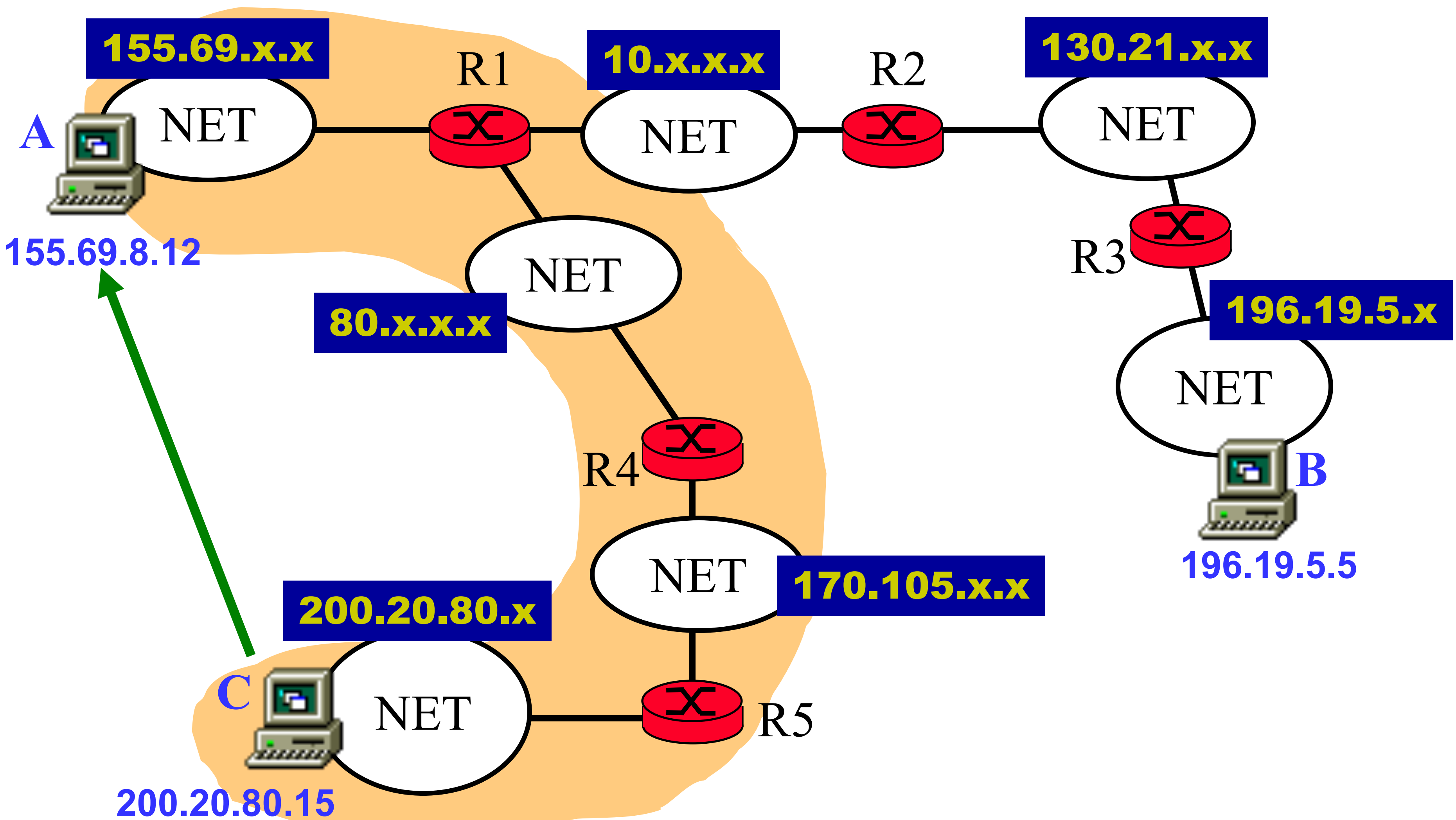
Q3

From Station A to B	From Station C to A	From Station B to C
155.69.8.10	200.20.80.12	196.19.5.104
10.203.20.10	170.105.10.21	130.21.80.90
130.21.10.30	80.90.10.3	90.80.120.10
196.19.5.5	155.69.8.12	170.105.10.20
		200.20.80.15

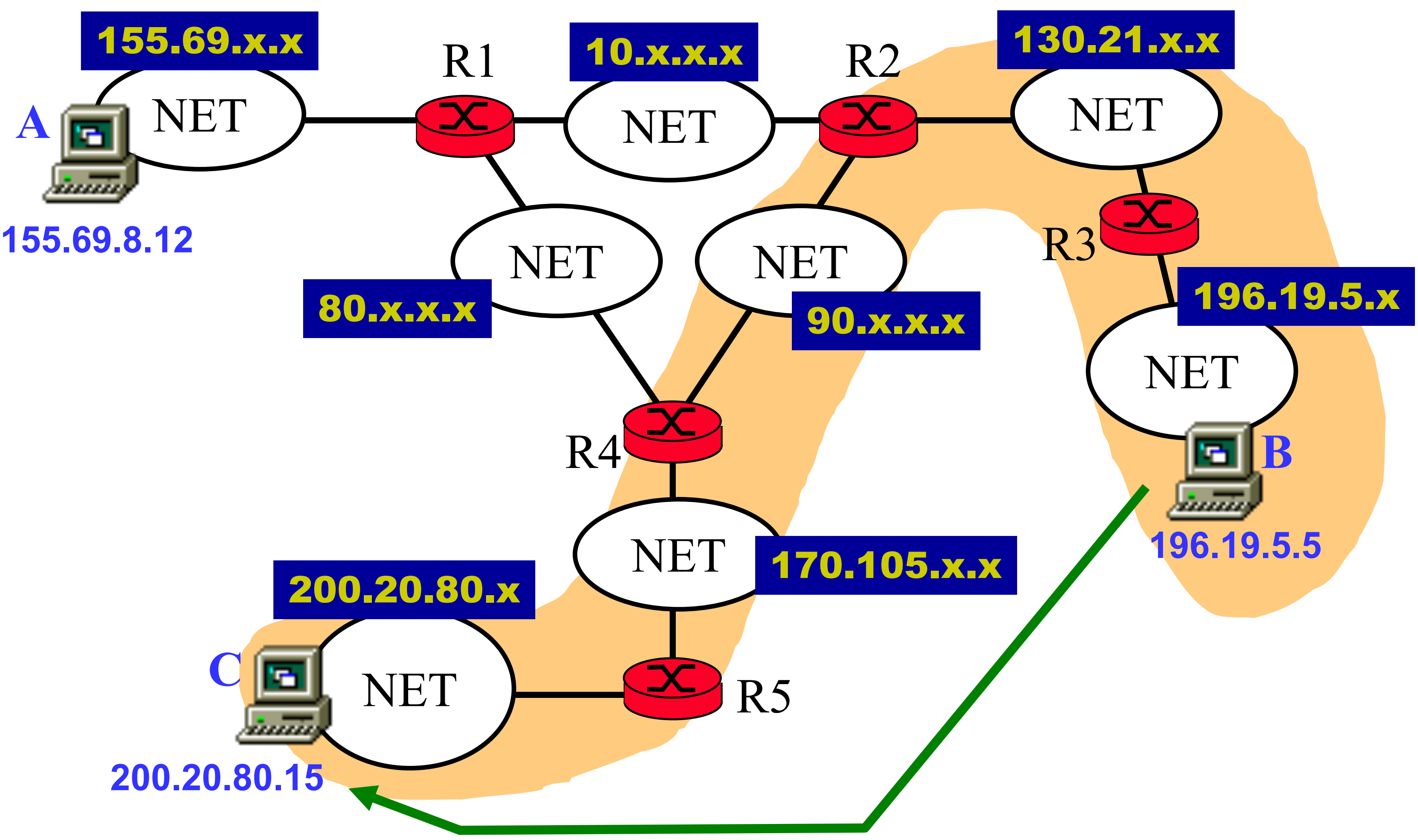
Q3: tracert A→B



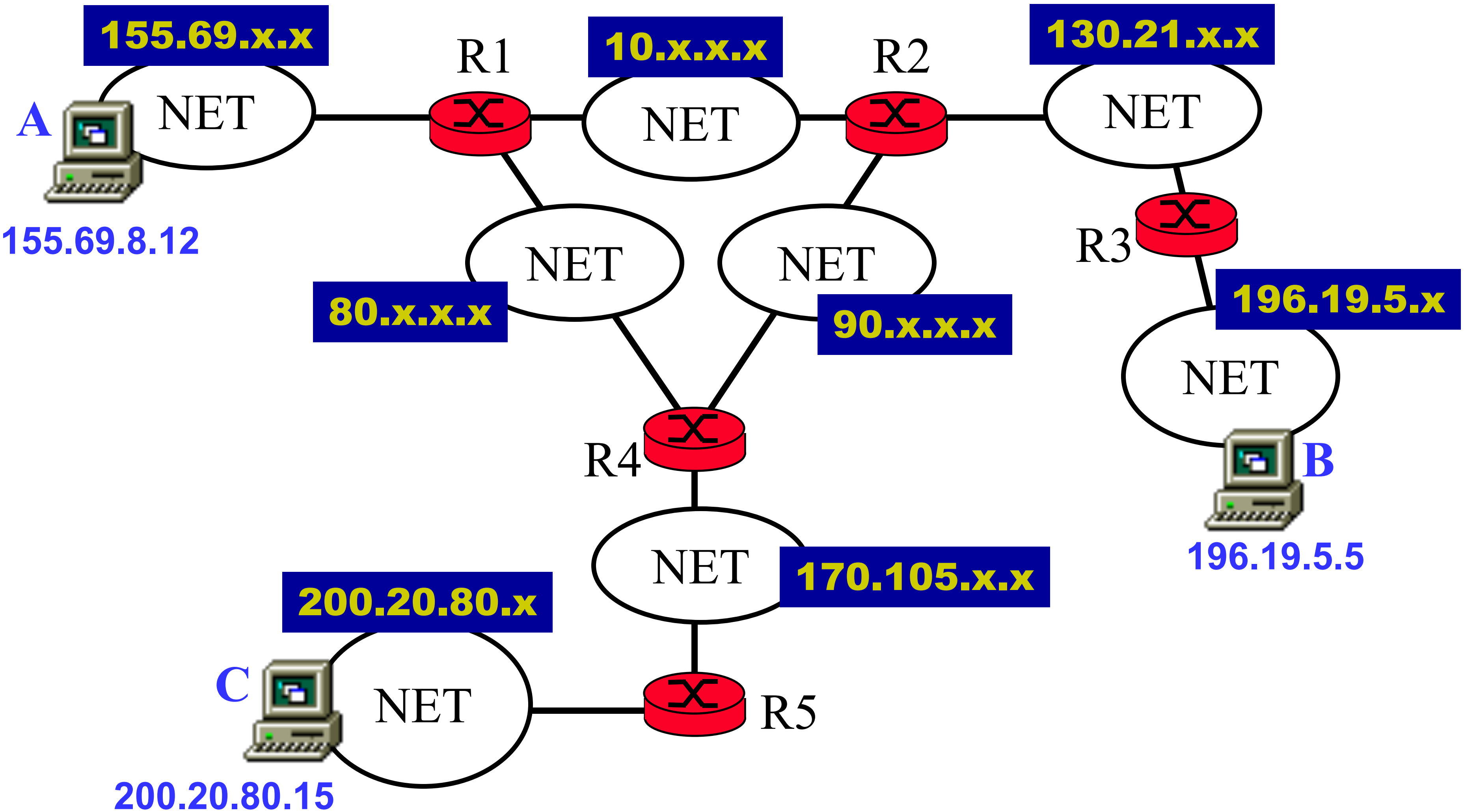
Q3: tracert C→A



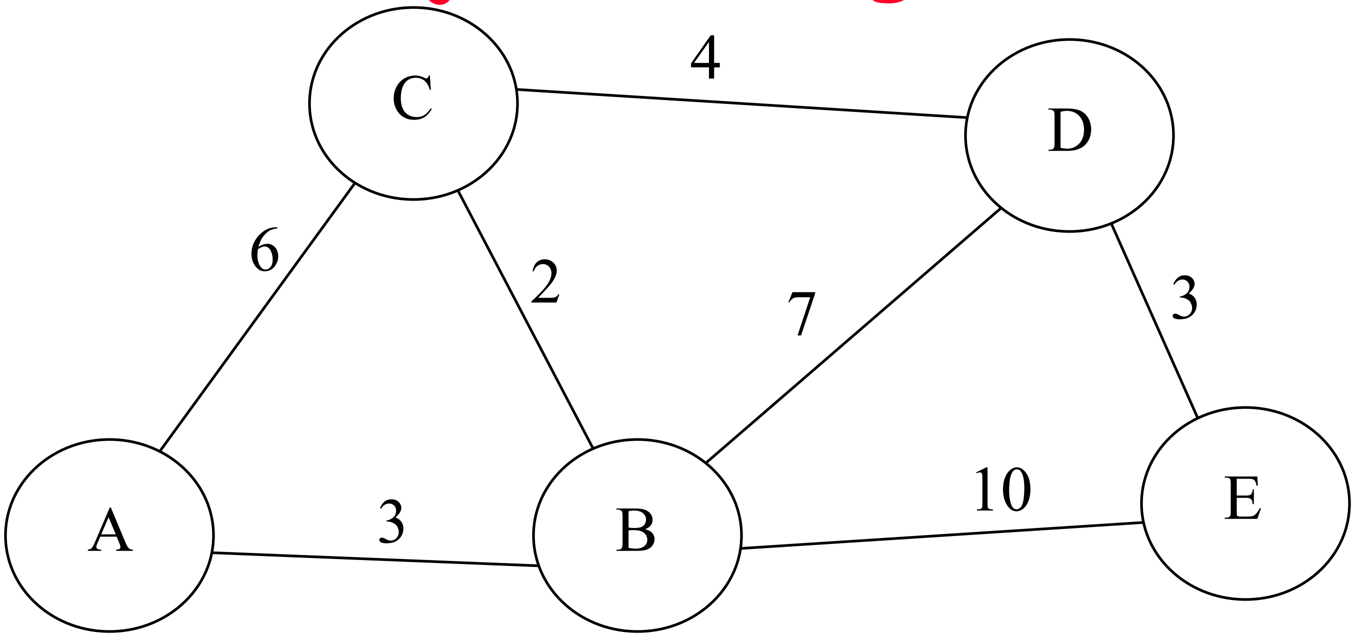
Q3: tracert B→C



Q3: tracert


155.69.x.x
R1
10.x.x.x
R2
130.21.x.x
A
NET
NET
NET
155.69.8.12
NET
NET
R3
80.x.x.x
90.x.x.x
196.19.5.x
NET
R4
B
NET
170.105.x.x
196.19.5.5
200.20.80.x
C
NET
R5
200.20.80.15

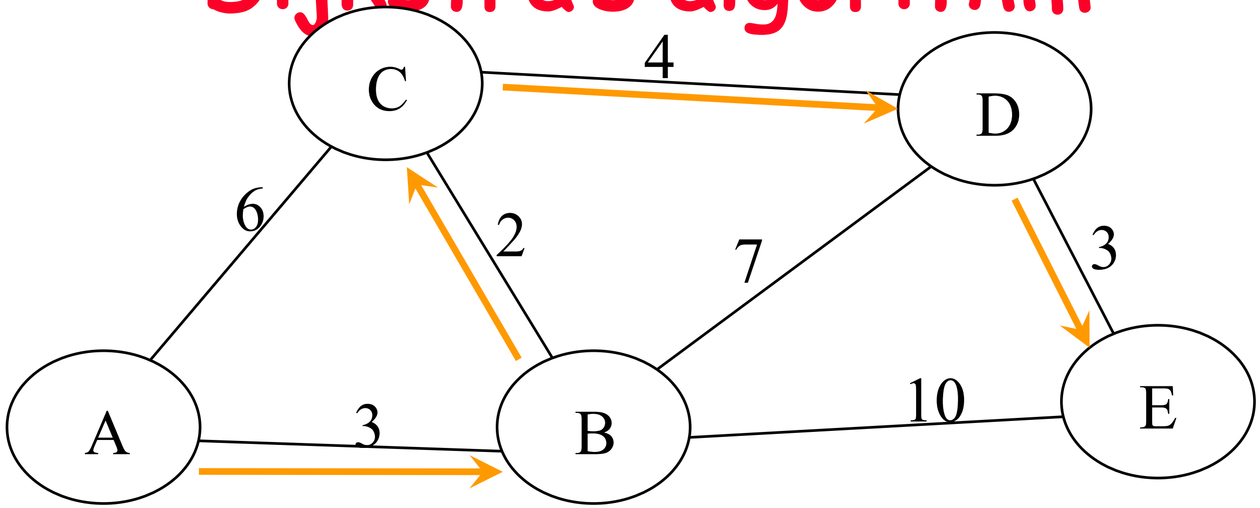
Q4: Link state routing protocol – Dijkstra's algorithm



LSA:

A	B	C	D	E
B=3	A=3	A=6	B=7	B=10
C=6	C=2	B=2	C=4	D=3
	D=7	D=4	E=3	
	E=10			

Q4: Link state routing protocol - Dijkstra's algorithm



Iterations of the algorithm

	B	C	D	E
{A}	3✓	6	∞	∞
{A,B}	3	5✓	10	13
{A,B,C}	3	5	9✓	13
{A,B,C,D}	3	5	9	12✓
{A,B,C,D,E}	3	5	9	12

In addition to the office hours listed in the previous slide, please feel free to contact Assistant Professor Jun ZHAO as follows to schedule appointments to ask questions. Thanks!

WhatsApp: http://personal.ntu.edu.sg/JunZhao/whatsapp.png
WeChat: https://personal.ntu.edu.sg/JunZhao/wechat.png
Singapore Phone Number (WhatsApp): 8648 3534
Email or Microsoft Teams: JunZhao@ntu.edu.sg
Skype ID: live:junzhaocmu
Office: Block N4, Room 02C-111, 50 Nanyang Ave, Singapore 639798
Homepage: http://personal.ntu.edu.sg/JunZhao/

SC2008
CE3005:Computer Networks
CZ3006:Netcentric Computing

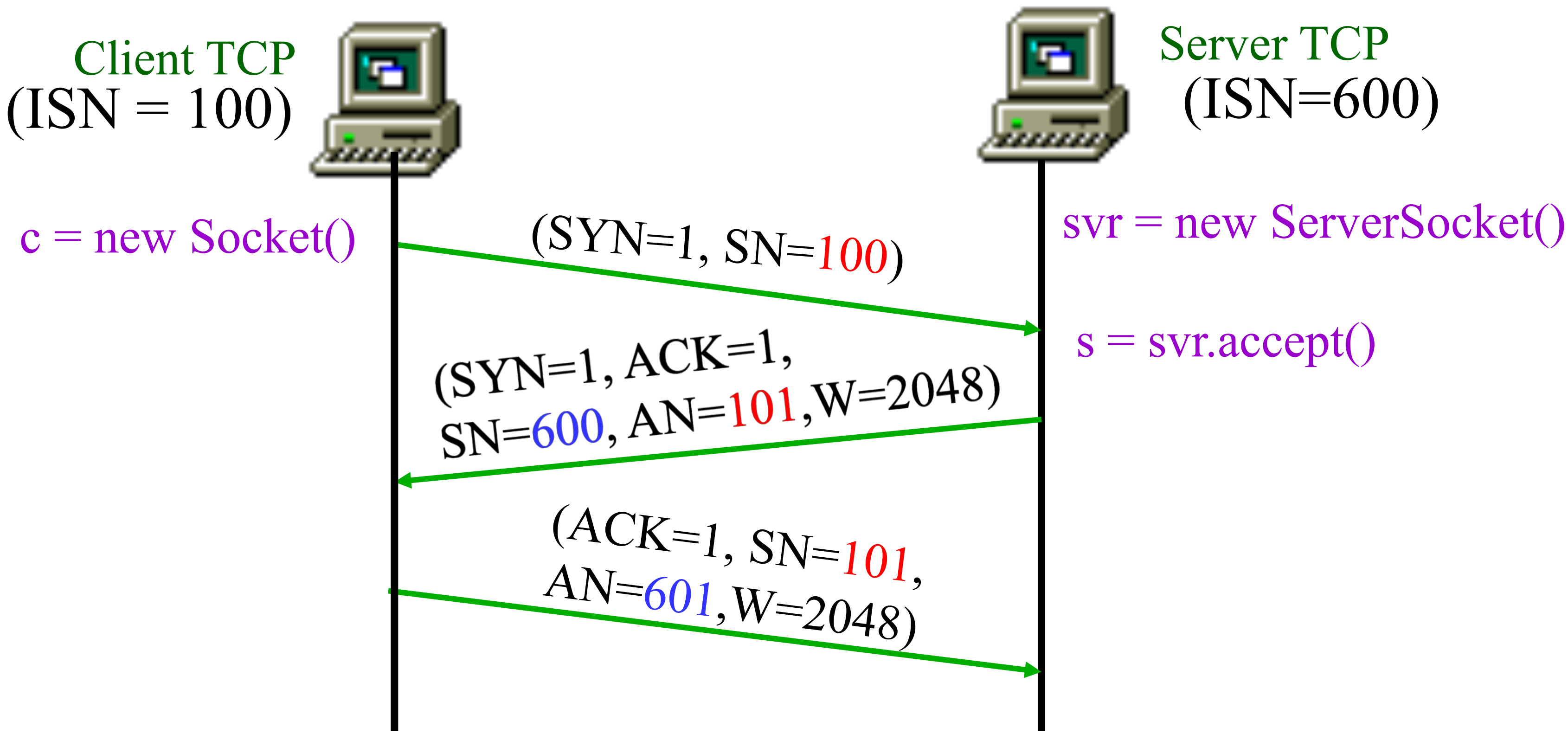
Transport layer

Q1. Information provided

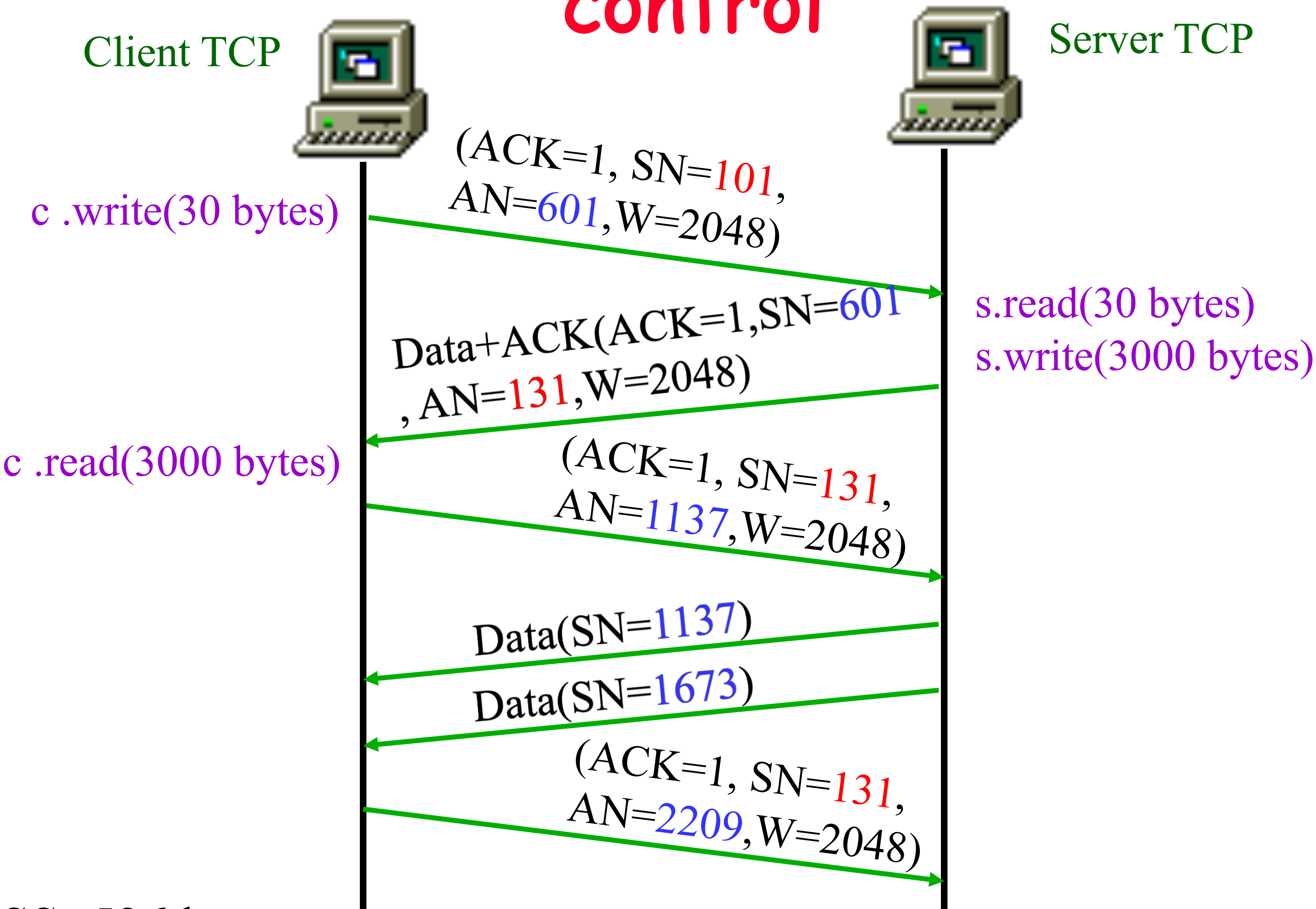
Assume that the Initial Sequence Number (ISN) for the Client TCP is 100 and the ISN for the Server TCP is 600. Both window sizes are fixed at 2048 bytes. The Maximum Segment Size (MSS) is 536 bytes, and the initial congestion window size is 1 MSS.

Client TCP	Server TCP
	svr = new ServerSocket()
c = new Socket()	
	s = svr.accept()
c.write(30 bytes)	
	s.read(30 bytes)
	s.write(3000 bytes)
c.read(3000 bytes)	
c.close()	
	s.close()

Q1: TCP - connection establishment

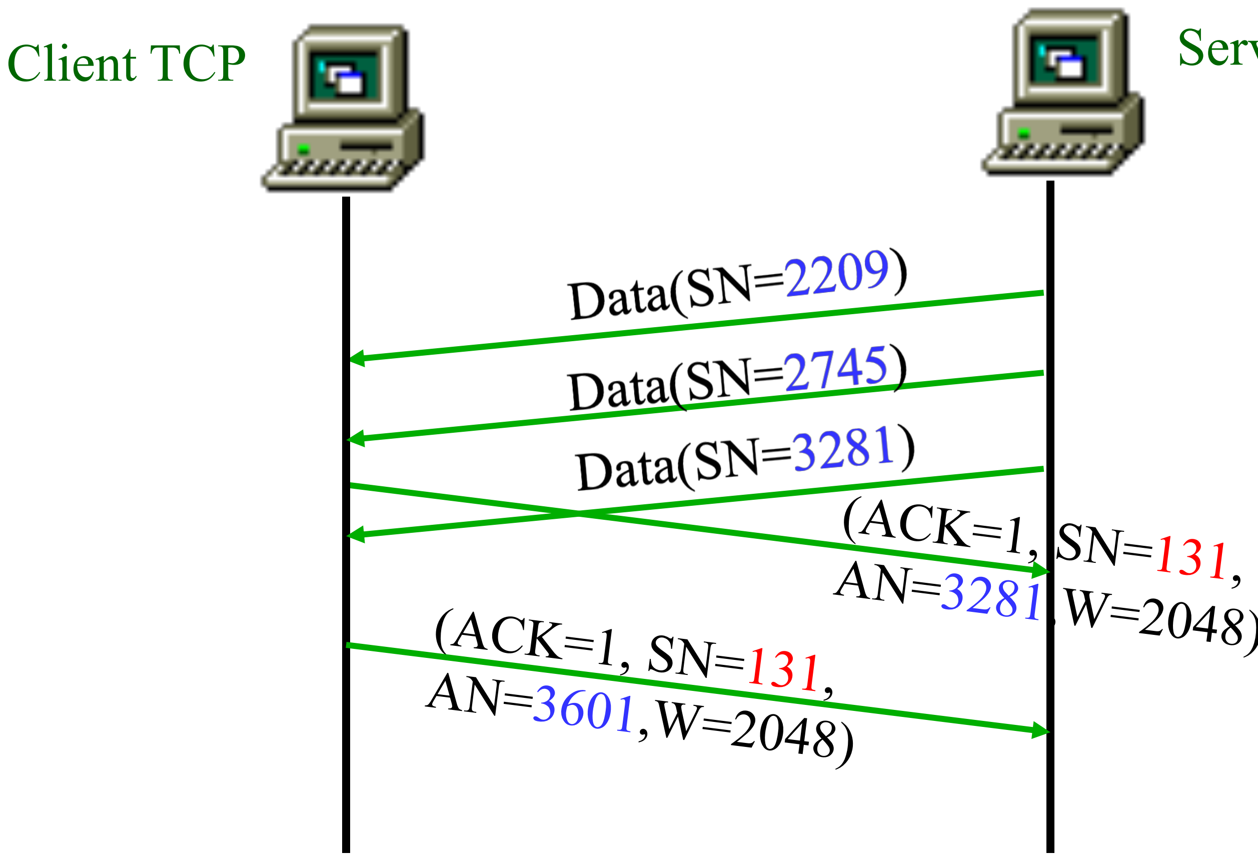


Q1: TCP – flow and congestion control



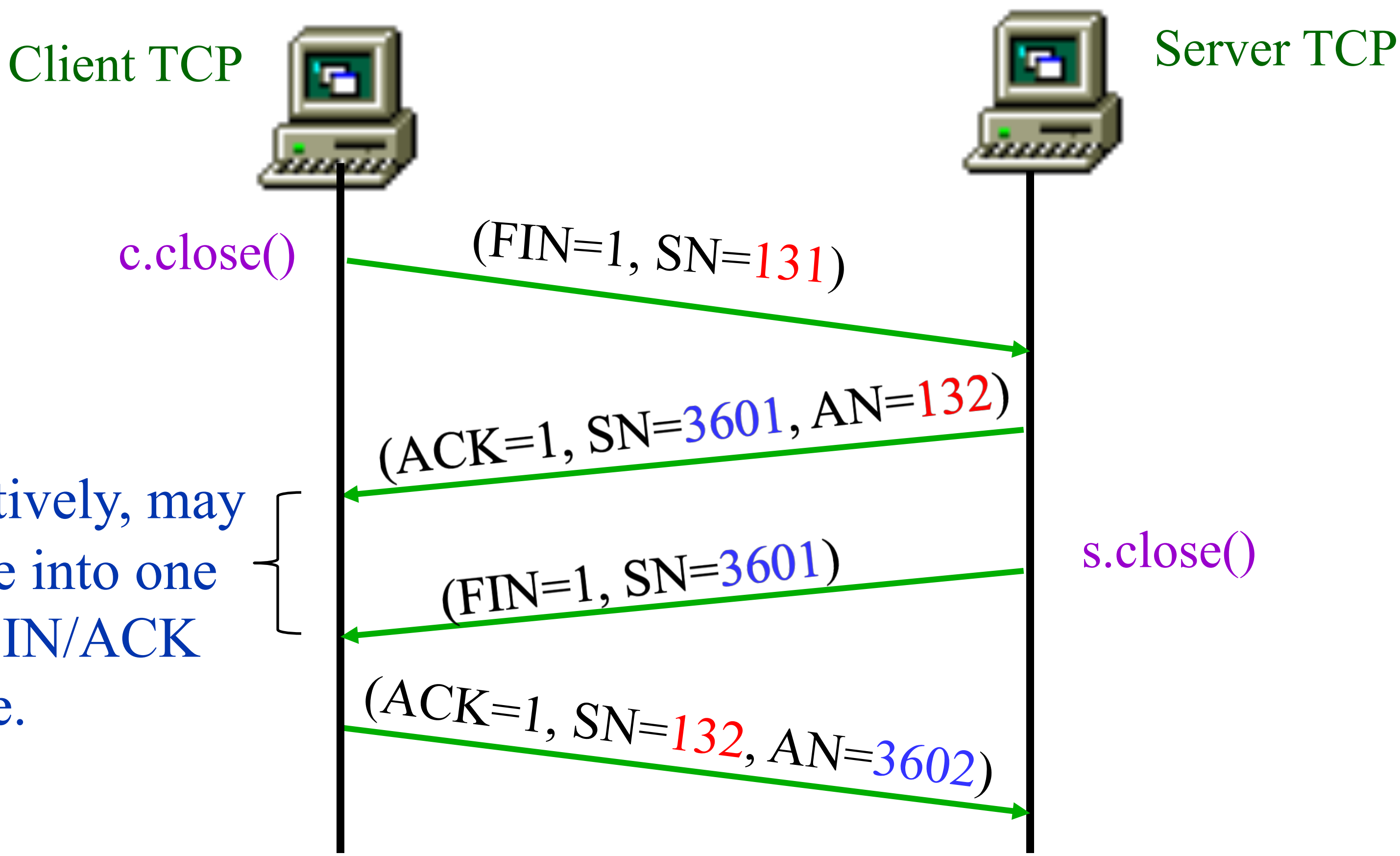
MSS=536 bytes

Q1: TCP – flow and congestion control



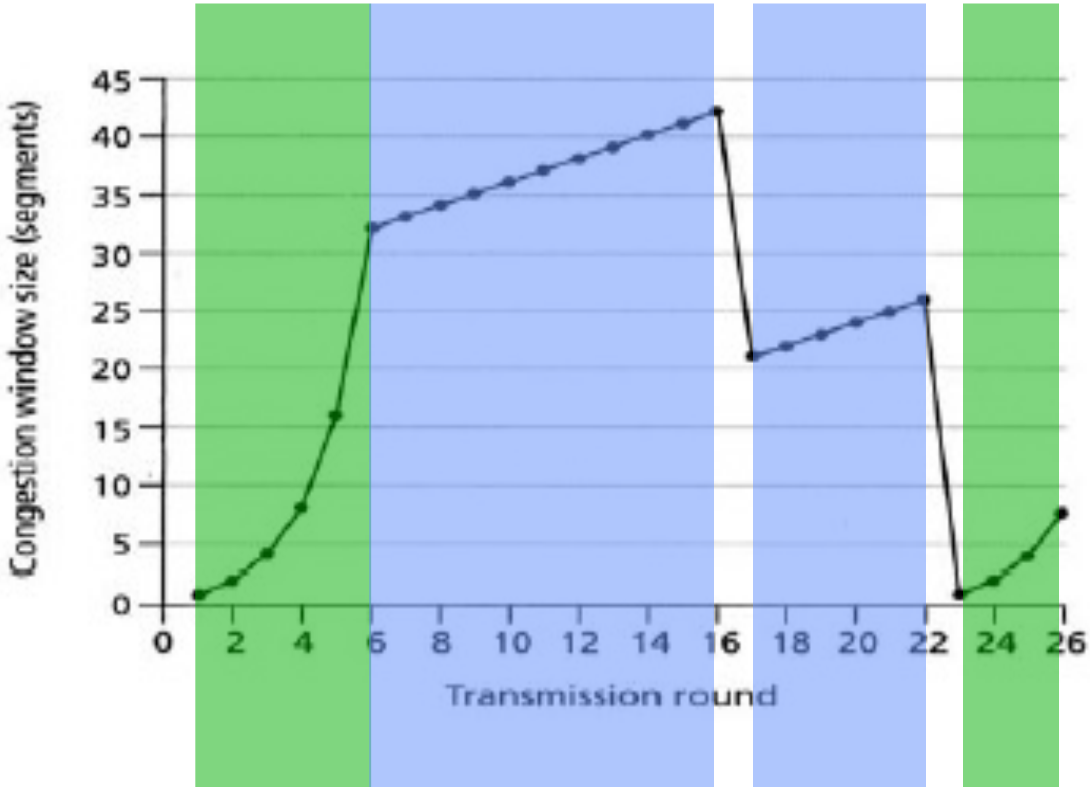
MSS=536 bytes

Q1: TCP - connection termination



Alternatively, may combine into one single FIN/ACK message.

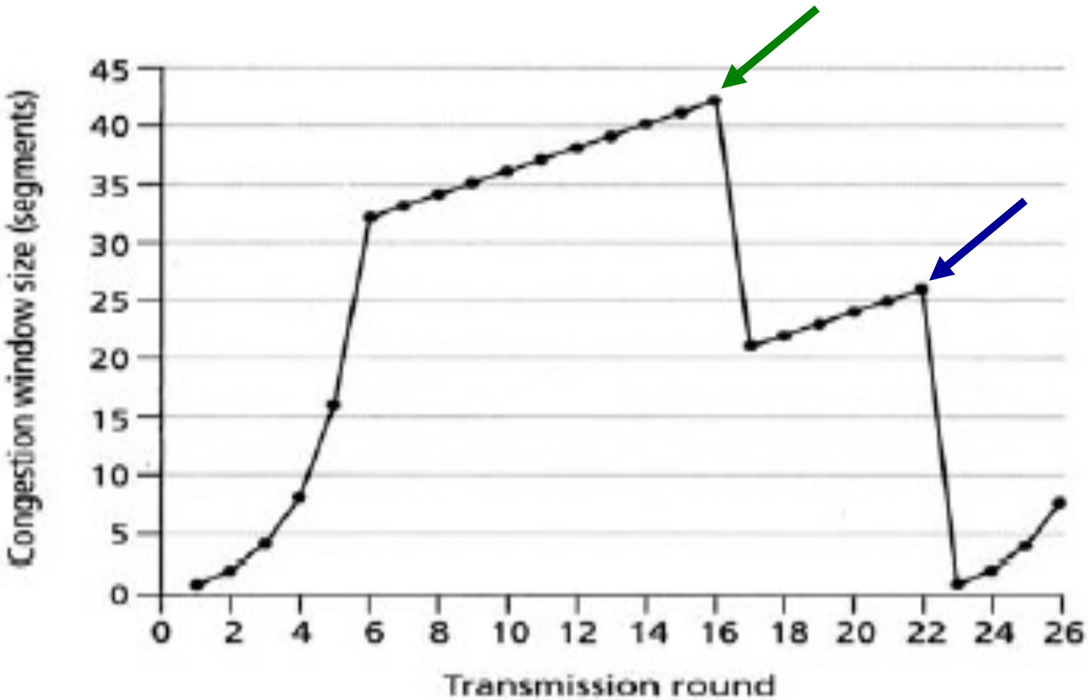
Q2: TCP Congestion Control



(a) Slow start [1,6] & [23,26]

(b) Congestion avoidance [6,16] & [17,22]

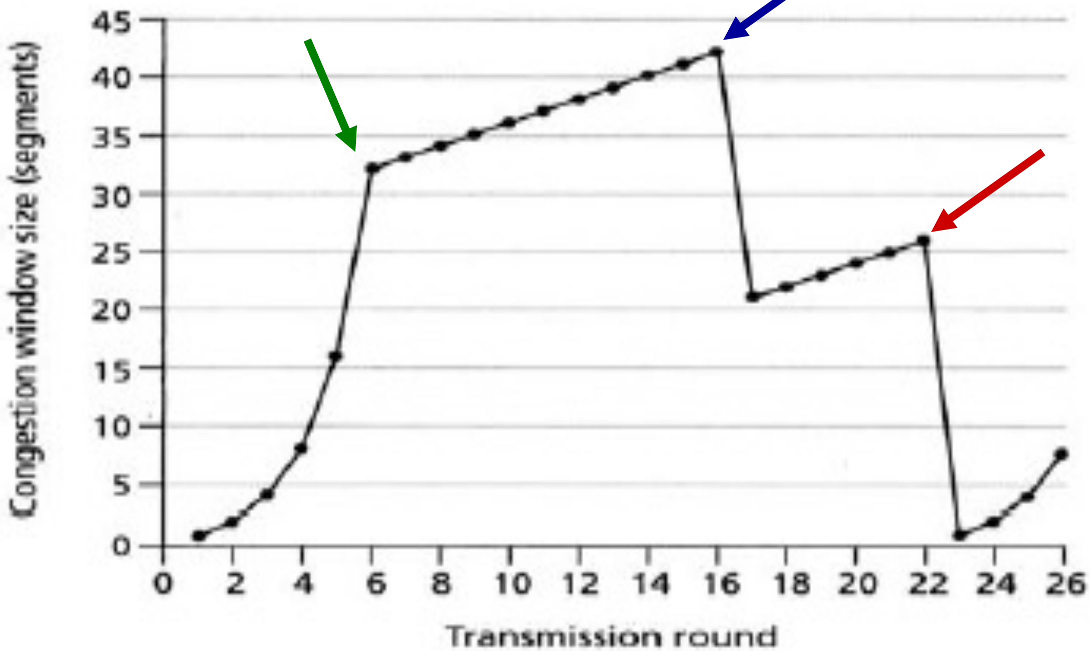
Q2: TCP



(c) At 16th transmission round, the host experienced a triple duplicate ACKs, because otherwise it will drop its cwnd to 1

(d) At 22nd transmission round, the host experienced a timeout of ACK, hence it drops its cwnd to 1

Q2: TCP

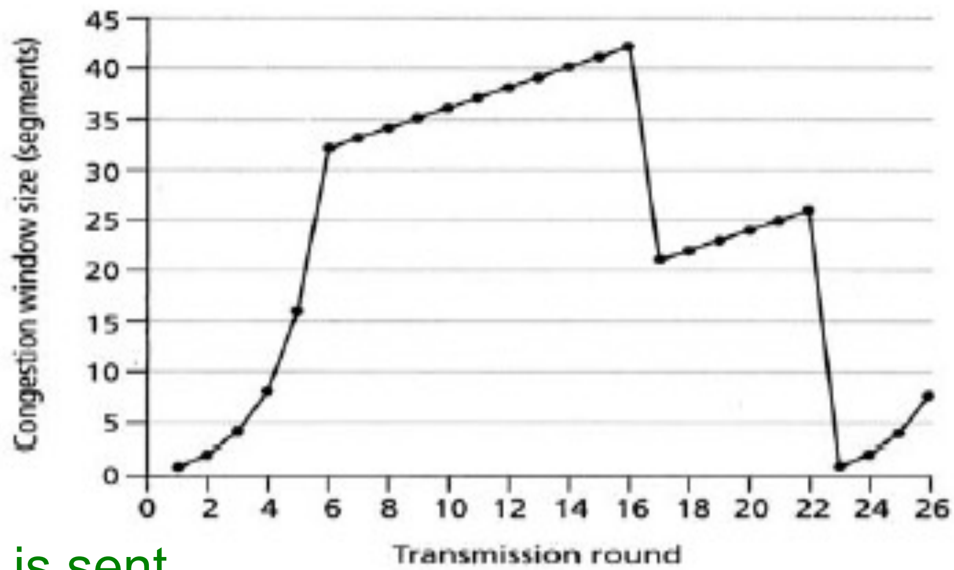


(e) Threshold at 1st = 32 (see 6th transmission round)

(f) Threshold at 18th = 42/2 = 21 (see 16th transmission round)

(g) Threshold at 24th = 26/2 = 13 (see 22nd transmission round)

Q2: TCP



(h)

During 1st, round, segment 1 is sent

During 2nd, round, segment 2-3 are sent

During 3rd, round, segment 4-7 are sent

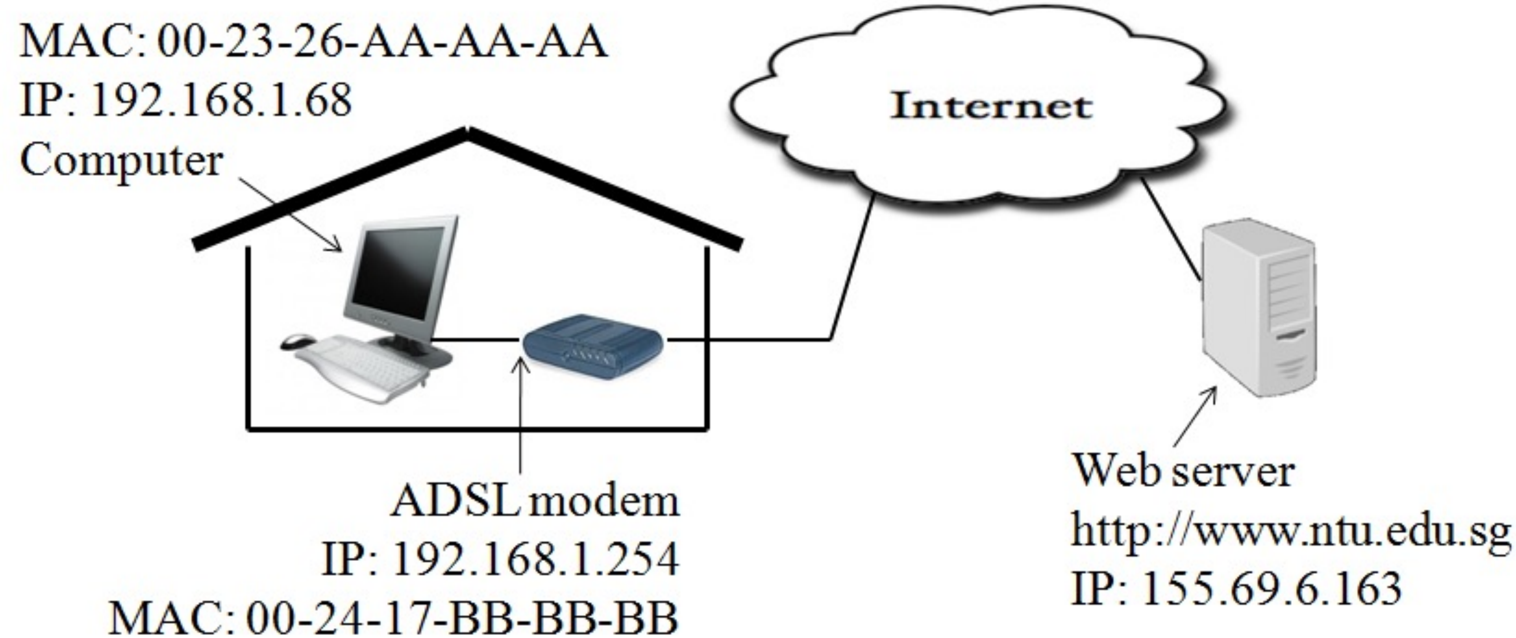
During 4th, round, segment 8-15 are sent

During 5th, round, segment 16-31 are sent

During 6th, round, segment 32-63 are sent

During 7th, round, segment 64-96 are sent <<< segment70 sent

Q3(a): Understanding Internet




```
c:\>ipconfig /all
Ethernet adapter Local Area Connection:
    Physical Address . . . . . . . . . . . : 00-23-26-AA-AA-AA
    DHCP Enabled . . . . . . . . . . . . : Yes
    IPv4 Address . . . . . . . . . . . . . . : 192.168.1.68
    Subnet Mask . . . . . . . . . . . . . . : 255.255.255.0
    Default Gateway . . . . . . . . . . . : 192.168.1.254
    DHCP Server . . . . . . . . . . . . .  : 192.168.1.254
    DNS Server . . . . . . . . . . . . . . . : 192.168.1.254
```

Q3(a): Understanding Internet

Roles performed by ADSL modem:

- DHCP server: configure host with IP address, subnet mask, etc.
- DNS server: resolve domain name to corresponding IP address
- Default gateway: forward packets to outside networks not directly reachable by the host
- NAT: enable host to use private IP address by translating it to public IP address and vice versa

Q3(b): Understanding Internet

Visit http://www.ntu.edu.sg

Frame	MAC Address		IP Address (if applicable)		Purpose of Frame
	Source	Destination	Source	Destination	
1.	00-23-26-AA-AA-AA	FF-FF-FF-FF-FF-FF	-	-	ARP request for 192.168.1.254
2.	00-24-17-BB-BB-BB	00-23-26-AA-AA-AA	-	-	ARP reply
3.	00-23-26-AA-AA-AA	00-24-17-BB-BB-BB	192.168.1. 68	192.168.1.254	DNS request for www.ntu.edu.sg
4.	00-24-17-BB-BB-BB	00-23-26-AA-AA-AA	192.168.1.254	192.168.1.68	DNS reply 155.69.6.163

Q3(b): Understanding Internet

Visit http://www.ntu.edu.sg

Frame	MAC Address		IP Address (if applicable)		Purpose of Frame
	Source	Destination	Source	Destination	
5.	00-23-26-AA-AA-AA	00-24-17-BB-BB-BB	192.168.1.68	155.69.6.163	TCP 3-way handshake
6.	00-24-17-BB-BB-BB	00-23-26-AA-AA-AA	155.69.6.163	192.168.1.68	TCP 3-way handshake
7.	00-23-26-AA-AA-AA	00-24-17-BB-BB-BB	192.168.1.68	155.69.6.163	TCP 3-way handshake

Q3(b): Understanding Internet

Visit http://www.ntu.edu.sg

Frame	MAC Address		IP Address (if applicable)		Purpose of Frame
	Source	Destination	Source	Destination	
8.	00-23-26-AA-AA-AA	00-24-17-BB-BB-BB	192.168.1.68	155.69.6.163	HTTP request
9.	00-24-17-BB-BB-BB	00-23-26-AA-AA-AA	155.69.6.163	192.168.1.68	HTTP reply

Q4: TCP throughput

Information Provided

- Link information:
    - Link speed = 1Gbps,
    - RTT = 100 milliseconds
- File size: 1 GByte
- TCP congestion control configuration:
    - Maximum segment size 1 Kbyte
    - Maximum number of segment 16

Q4 Solution

- In one RTT, the maximum amount of data that is transmitted is
  - 1KB x 16 = 16KB
- Since there are 10 RTT in one second, as RTT is 100 millisecond.
  - Throughput = 16KB x 10 = 160KB per second
- Duration of transfer
  - 1,000,000 KB/160KB = 6,250 seconds

In addition to the office hours listed in the previous slide, please feel free to contact Assistant Professor Jun ZHAO as follows to schedule appointments to ask questions. Thanks!

WhatsApp: http://personal.ntu.edu.sg/JunZhao/whatsapp.png
WeChat: https://personal.ntu.edu.sg/JunZhao/wechat.png
Singapore Phone Number (WhatsApp): 8648 3534
Email or Microsoft Teams: JunZhao@ntu.edu.sg
Skype ID: live:junzhaocmu
Office: Block N4, Room 02C-111, 50 Nanyang Ave, Singapore 639798
Homepage: http://personal.ntu.edu.sg/JunZhao/