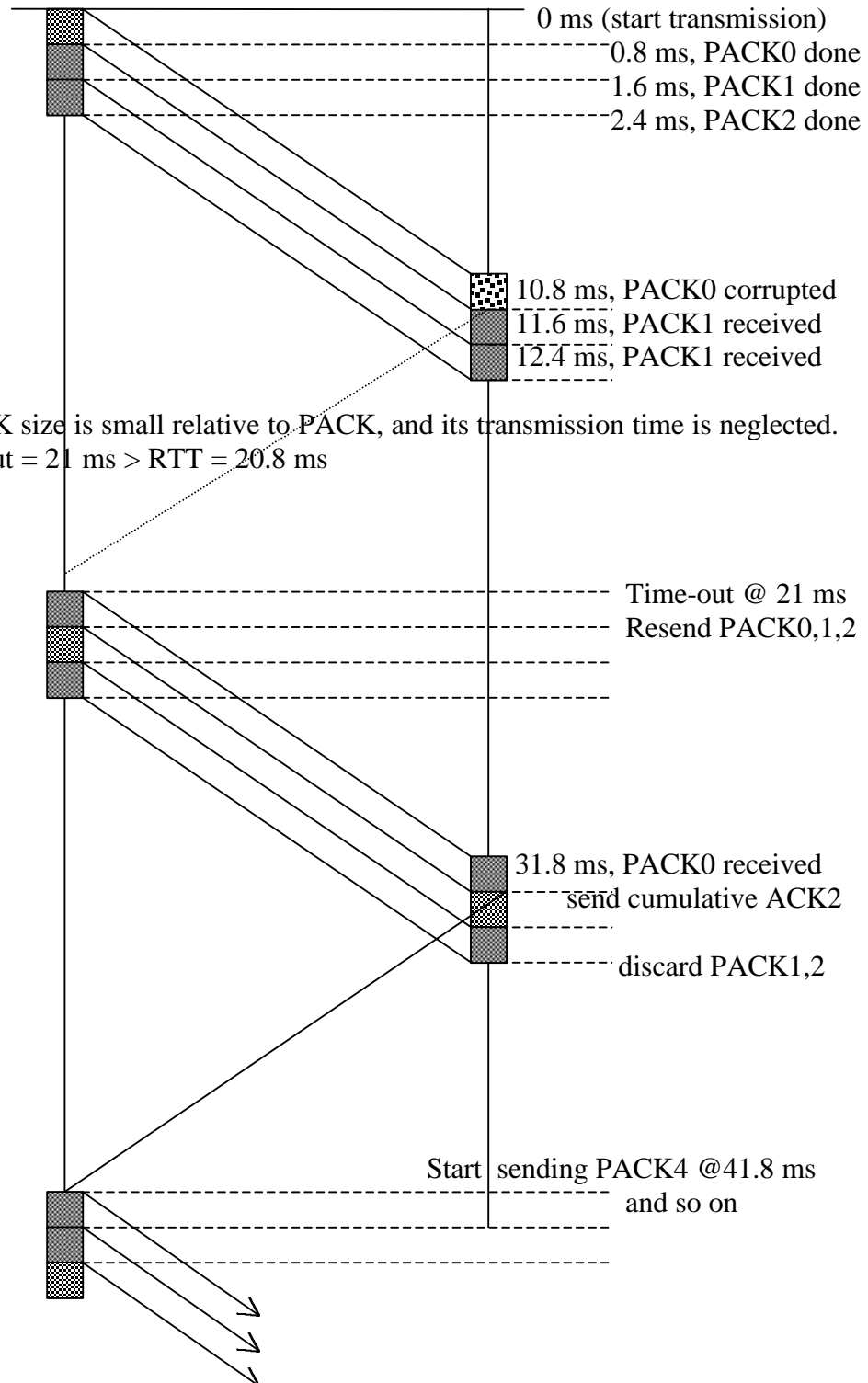


## 2.993: Principles of Internet Computing

### Homework #4 Solutions

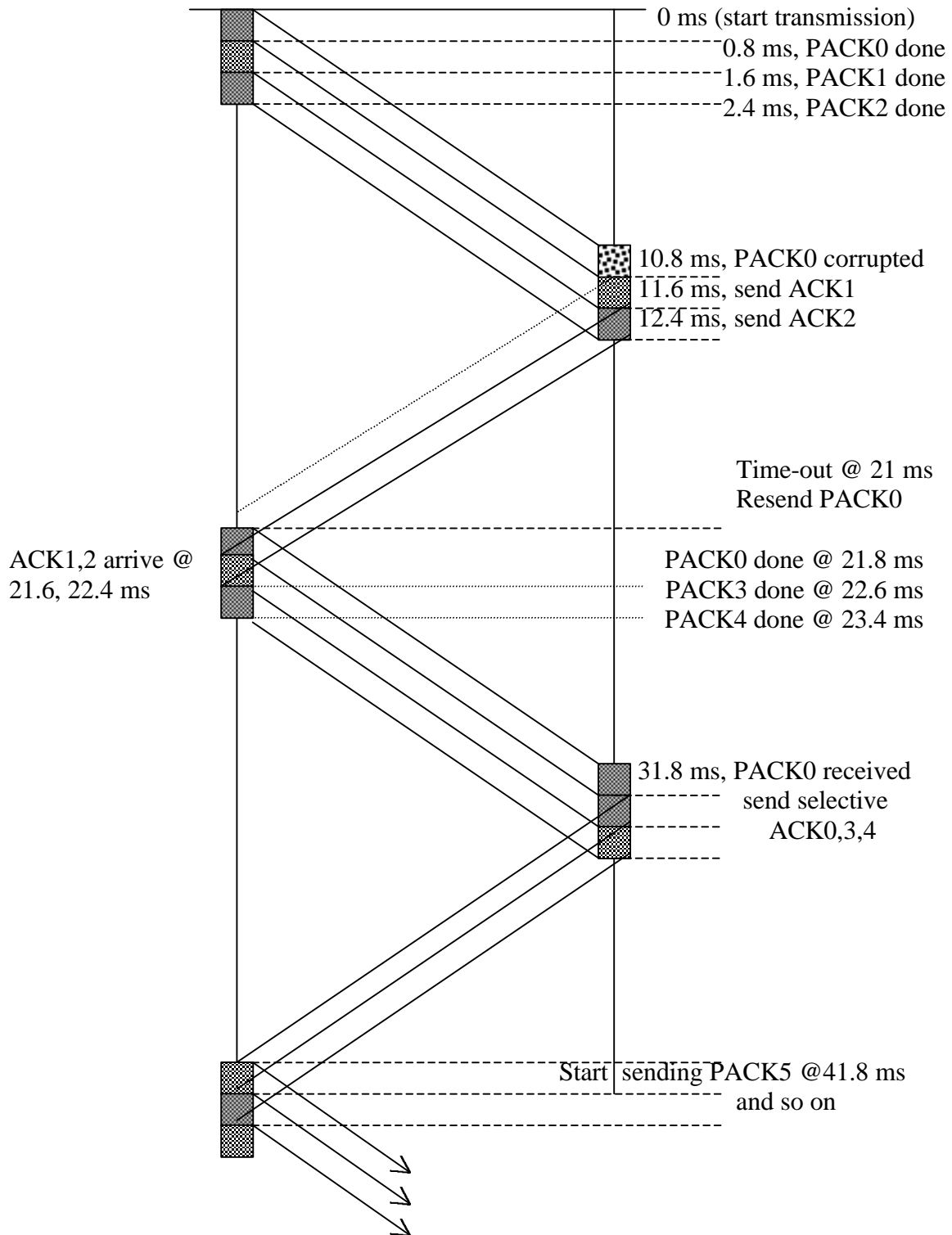
#### 1.(a) Cumulative ACKs

propagation delay = 10 ms, packet transmission time ~ 0.8 ms



We assume ACK size is small relative to PACK, and its transmission time is neglected.  
We pick time-out = 21 ms > RTT = 20.8 ms

(b) selective ACKs



2. Cumulative ACKs has less complexity because ACKs are not sent for every packet. Selective ACKs can improve the throughput.

3.

(a) For Go-Back-N,  $RWS = 1$ .

(b) the minimum required sequence size for a sliding window is  $(SWS + RWS)$ .

If  $SWS=N$  and  $RWS=M$ , consider the following worst-case scenario: The sender sends  $N$  packets. All are received in order at the receiver, but all of its ACKs are corrupted. The receiver is expecting packets from  $(N+1)$  to  $(N+M)$ . After time-out, the sender repeats the same packets, but the receiver is expecting new ones. For  $RWS=M$ , the confusion is avoided if the (minimum) total number of SeqNum is  $(N+M)$ .

i) 6            ii) 5            iii)  $X+Y$

4. We know  $RTT \times \text{bandwidth} = \text{window size}$

Thus,  $\text{throughput} = \text{window size}/RTT$

i)        (a) 1 KB/40 ms        (b) 1KB/20ms  
(Keep in mind though that the actual throughput is the same.)

ii) This question is related to a comparison between end-to-end and link-to-link flow controls. As you know, the Internet (with IP) uses datagram (connectionless) communication and end-to-end sliding window (TCP). But a virtual-circuit (connection-oriented) communication is used for such networks as X.25 or ATM. The latter has the advantage of providing some sort of quality of service guarantee (throughput, delay). This, however, requires coordination among routers within the network, which increases complexity. For the Internet, where many different networks may be interconnected, the function of flow control is left to the end user.

We will revisit this topic later in the semester. You can also read 4.1.2 and 4.1.3 from the text.

5.

During 1<sup>st</sup> RTT,  $PACK=1$  is sent. The  $snd\_wnd=1$ . After receiving  $ACK=2$  at the end of 1<sup>st</sup> RTT,  $snd\_wnd$  is incremented to 2, and  $PACK=2,3$  are sent at the start of 2<sup>nd</sup> RTT. After  $ACK=3$  is received at the end of 2<sup>nd</sup> RTT,  $snd\_wnd=3$ . During 3<sup>rd</sup> RTT, after receiving  $ACK=4$ ,  $snd\_wnd=4$ . Thus, during 3<sup>rd</sup> RTT, there are four outstanding packets,  $PACK=4,5,6,7$ . When  $ACK=5$  is received at the end of 3<sup>rd</sup> RTT, the slow-start threshold begins.  $ACKs=5,6,7,8$  contribute  $1/4$  each.  $ACKs=6,7,8$  arrive during 4<sup>th</sup> RTT. When  $ACK=8$  (for  $PACK=7$ ) is received at the end of 4<sup>th</sup> RTT,  $snd\_wnd=5$ . Thus, during 4<sup>th</sup> RTT, there are five outstanding packets,  $PACK=8,9,10,11,12$ . The acknowledgments,  $ACK=9,10,11,12,13$  which arrive during 5<sup>th</sup> RTT, each contribute  $1/5$  to the window size.

Also, during 5<sup>th</sup> RTT, PACK=13,14,15,16,17 are sent. After snd\_wnd=6, at the start of 6<sup>th</sup> RTT, 6 packets can be outstanding. Thus, PACK=18 is also sent. But delay-bandwidth product is only 3, and the bottleneck buffer size is 2. Thus, PACK=18 is dropped, and it causes fast retransmission.

| RTT# | window range | packets sent      | ssthresh | buffer size |
|------|--------------|-------------------|----------|-------------|
| 1    | 1 - 2        | 1                 | 4        | 0           |
| 2    | 2 - 4        | 2,3               | 4        | 0           |
| 3    | 4 - 5        | 4,5,6,7           | 4        | 1           |
| 4    | 5 - 6        | 8,9,10,11,12      | 4        | 2           |
| 5    | 6 -          | 13,14,15,16,17,18 | 4        | 2/drop      |