1. Appendix

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2. STP protocol implementation

Sender:

In the Sender part, we need to keep send a window's all segments at every time, at the same time we need to open a listener (thread) to keep listen Ack package send from the Receiver Side. Also another thread named Resender (in charge of retransmit the dropped packets). The detail are in the flow chart below

Receiver:

Receiver have a buffer to buffer all received segments' data. I use a variable named **SendAck** to record what's the next segment I want to receive from Sender. e.g. the Sender sends 2,3,4,5 and 3 is lost. So even though the 4, 5 have arrived Receiver successfully, the Receiver will send Ack segment with AckNum = 3. When 3 is received, it will update the SendAck = 6, because 4, 5 have already in the buffer.

Connection establish:

Three-way hand shaking

The Sender send a SYN then waiting for receiver's SYN&ACK.

Lastly Sender send a SYN, and the connection is established

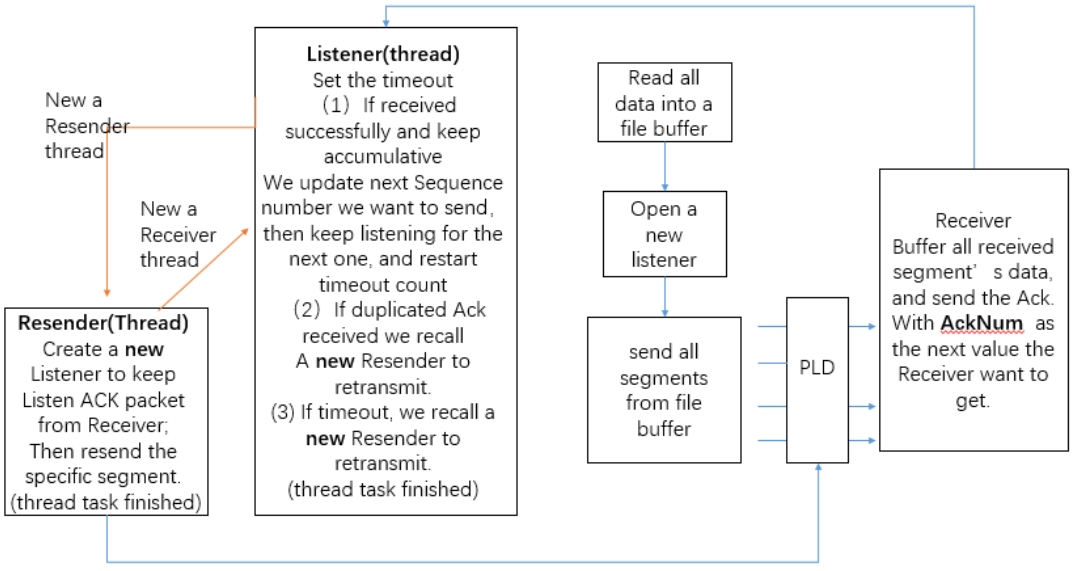
Connection closure:

The sender sends a Fin to inform the receiver

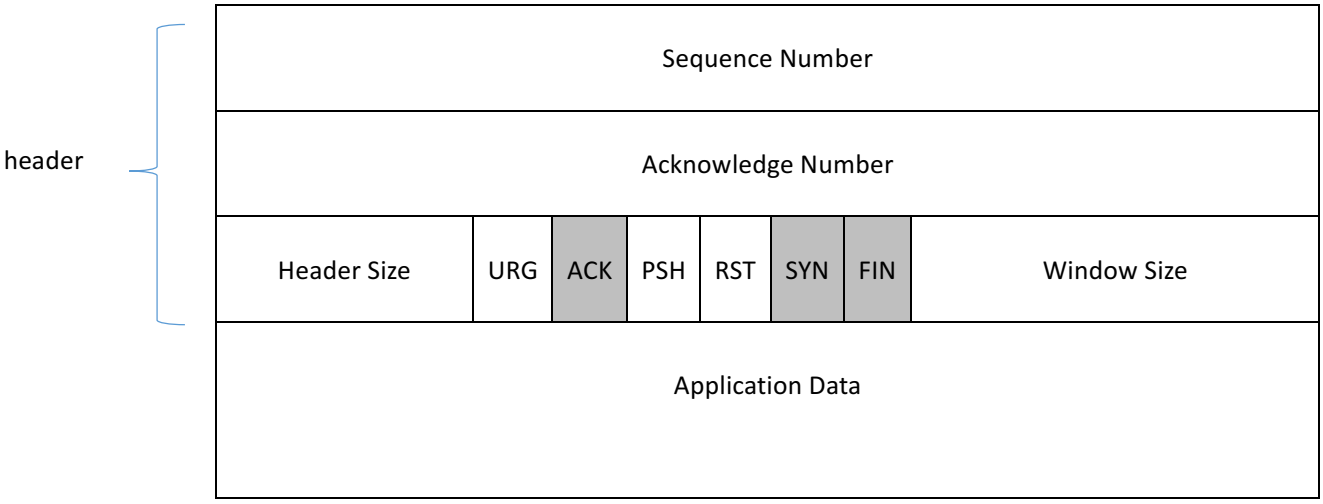
Then the Receiver send a Ack and a Fin

Lastly the Sender send a Ack and wait for a while then closed (In the waiting time I write the log file so I do not set a wait clock, just use the write log file's time as the wait time).

The flow chart below has list most feature in the programming.



3. STP Header Structure



Overall 12bytes:

(1) Sequence Number: occupy 4 bytes

(2) Acknowledge Number: occupy 4 bytes

(3) Header Size: occupy 1 byte, because Header Size is 12 bytes, 1 byte is enough to store the value

(4) All flag together: occupy 1 byte, low 6 bits [URG, ACK, PSH, RST, SYN, FIN]

We only use ACK, SYN and Fin (Ack# valid, setup and teardown) in STP. e.g. If only ACK is set, the byte value is 1<<4; if ACK and SYN are set, the byte value is 1<<4 + 1<<1

(5) Window size: occupy 2 bytes, used for flowing control

In the programming

Class Segment has some methods could be used to facilitate the operation according the header:

private member:

- Sequence Number
- Acknowledge Number
- Header Size
- ACK flag
- SYN flag
- FIN flag
- Window Size

Constructor: assign each private member with corresponding value.

Method:

createHeader() could transfer private members into bytes and store them into a byte array.

4. Question Answer

(a)

Firstly, timeout value must greater than RTT, secondly the probability of drop is small only 0.1. I check the RTT by looking at the time from Sender sends a SYN until it received a SYN & ACK.
So, I estimate the timeout is no need to be very large.

(1) timeout = 8
java Sender 127.0.0.1 2222 test1.txt 500 50 9 0.1 300
java Receiver 2222 res1.txt
(complete data please check the Sender_log.txt)

snd	61.285302	FIN	1601	0	1
rcv	63.039291	ACK	2	0	1602
rcv	63.926211	FIN	2	0	1602
snd	64.945881	ACK	1602	0	3

we can find before 61ms, the file has already transmitted completely.
(2) timeout = 10
java Sender 127.0.0.1 2222 test1.txt 500 50 10 0.1 300
java Receiver 2222 res1.txt
(complete data please check the Sender_log.txt)

snd	53.982108	FIN	1601	0	1
rcv	54.916882	ACK	2	0	1602
rcv	55.423128	FIN	2	0	1602
snd	56.463743	ACK	1602	0	3

we can find before 53ms, the file has already transmitted completely.
(3) timeout = 11
java Sender 127.0.0.1 2222 test1.txt 500 50 11 0.1 300
java Receiver 2222 res1.txt

snd	70.301453	FIN	1601	0	1
rcv	71.379928	ACK	2	0	1602
rcv	72.228269	FIN	2	0	1602
snd	73.594036	ACK	1602	0	3

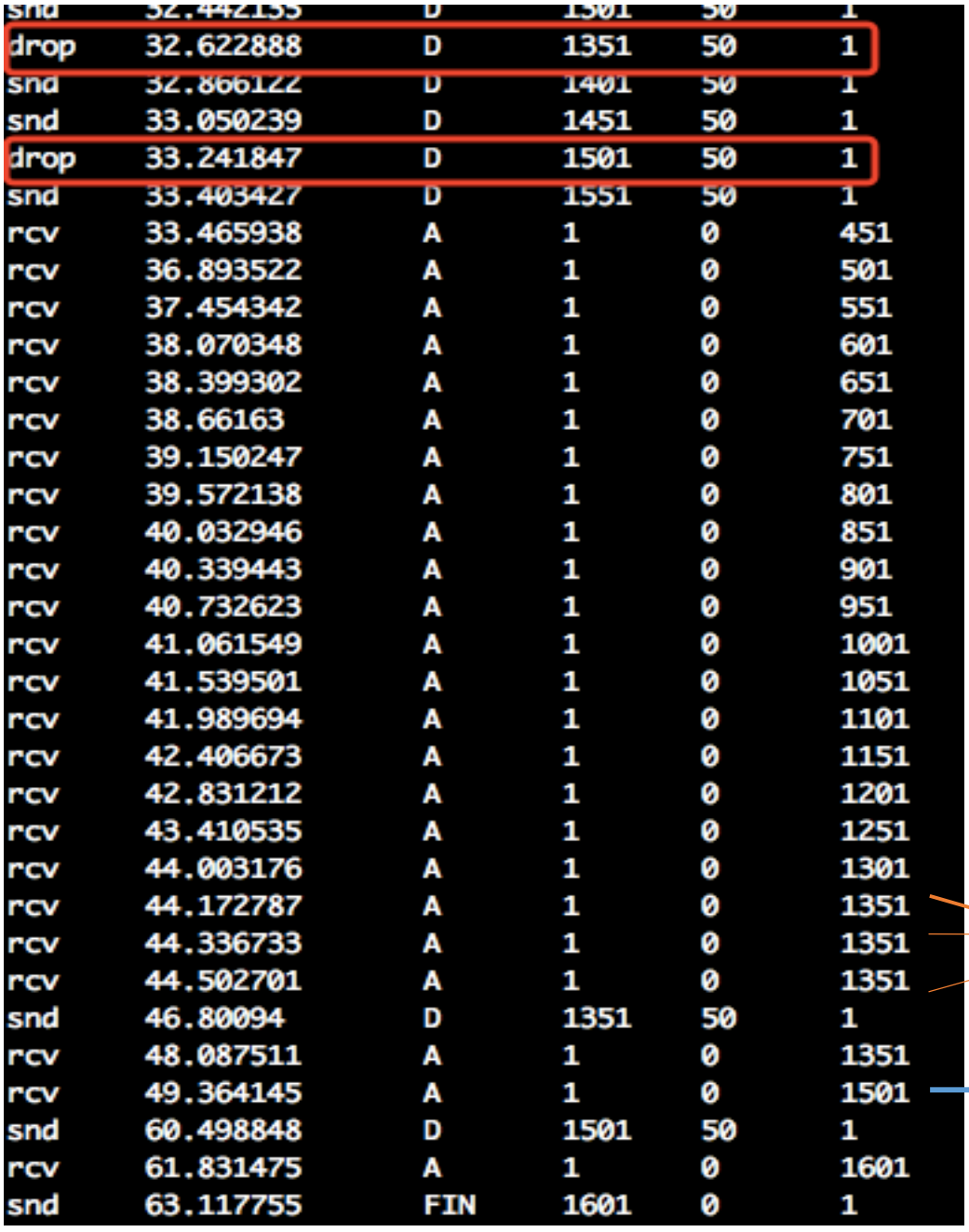
we can find before 70ms, the file has already transmitted completely.

So, **the most suitable timeout = 10**;

Experiment:

the Send_log.txt of timeout = 10, pdrop = 0.1, MWS = 500 bytes, MSS = 50 bytes, seed = 300

(complete log please check the Sender_log.txt, below is partial)



drop	32.622888	D	1351	50	1
snd	33.050239	D	1451	50	1
drop	33.241847	D	1501	50	1
rcv	33.465938	A	1	0	451
rcv	36.893522	A	1	0	501
rcv	37.454342	A	1	0	551
rcv	38.070348	A	1	0	601
rcv	38.399302	A	1	0	651
rcv	38.66163	A	1	0	701
rcv	39.150247	A	1	0	751
rcv	39.572138	A	1	0	801
rcv	40.032946	A	1	0	851
rcv	40.339443	A	1	0	901
rcv	40.732623	A	1	0	951
rcv	41.061549	A	1	0	1001
rcv	41.539501	A	1	0	1051
rcv	41.989694	A	1	0	1101
rcv	42.406673	A	1	0	1151
rcv	42.831212	A	1	0	1201
rcv	43.410535	A	1	0	1251
rcv	44.003176	A	1	0	1301
rcv	44.172787	A	1	0	1351
rcv	44.336733	A	1	0	1351
rcv	44.502701	A	1	0	1351
snd	46.80094	D	1351	50	1
rcv	48.087511	A	1	0	1351
rcv	49.364145	A	1	0	1501
snd	60.498848	D	1501	50	1
rcv	61.831475	A	1	0	1601
snd	63.117755	FIN	1601	0	1

duplicate Acks

timeout so resend

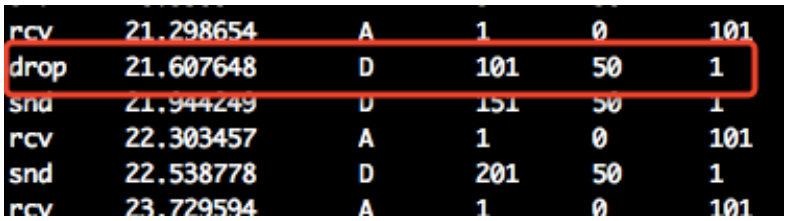
Additional experiment:

(complete log please check the Sender_log.txt, below is partial)

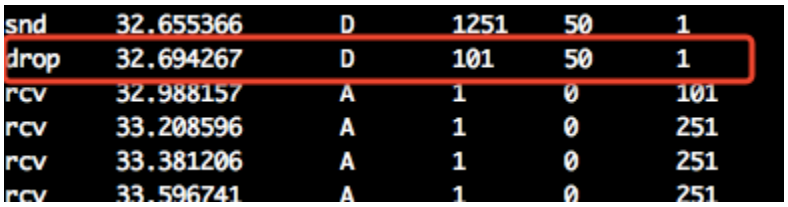
```
rcv     134.953665      A       1       0       1601
snd     136.841058      FIN     1601    0       1
rcv     137.764291      ACK     2       0       1602
rcv     138.616284      FIN     2       0       1602
snd     140.037227      ACK     1602    0       3
Amount of Data Transferred (in bytes): 1593
Number of Data Segments Sent (excluding retransmissions): 32
Number of Packets Dropped (by the PLD module): 13
Number of Retransmitted Segments: 12
Number of Duplicate Acknowledgements received: 30
```

we could find there are 13 Packets dropped, and retransmitted segment 12, which means there one segment dropped twice.

first time:



second time



And the pdrop increase , the chance of retransmit also increase. So, the overall time increases.

(b)

java Sender 127.0.0.1 2222 test2.txt 500 50 10 0.1 300

	Tcurrent = 10	4 * Tcurrent = 40	Tcurrent / 4 = 2
package number	40 + 5 = 45	40 + 5 = 45	40 + 5 = 45
Overall time	38.205509	46.550078	33.780733

Because the MSS, seed and file not changed, so STP package number is unchanged.

In terms of overall time, if a segment is dropped, and received 3 same ACKs, the Sender would reset the timer and resend the segment again:

if time out interval is very small (Tcuurent/4) Sender could hard to get 3 duplicates, but resend it after time out;

if time out interval is large (4 * Tcurrent), Sender could easier to get 3 duplicates and resend. But if Sender didn't get 3 duplicates, the segment will wait the time out finished, then it could be resend and timer will be reset.

So, if time out is too small, it would frequently resend the lost segment, which wastes the bandwidth. Reversely, if time out is large, it would waste time. So using time out and fast retransmit together could find a balanced way in this situation. Even though it may not be the best one, but it could fit most situations.