TCP Project README/Performance Analysis

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Data Structures

- TCP

```
type TCP struct {
    connections map[string]*TransControlBlock // key:localAddr.String()+remoteAddr
.String()
    socketMap   map[int]*TransControlBlock
    verbose     bool
    lock        sync.RWMutex
    ran         *rand.Rand
    network     *NetworkLayer
}
```

Above is the global TCP structure, which contains 2 hashmaps mapping socket number and socket address pair to the corresponding `TransControlBlock`, which stores the specific attributes regarding this connection. Specifically, if a socket is built with address:port pair of (192.168.0.2:4323, 192.168.0.4:5000), and this socket is assigned with socket number 3, then both 3 and string "192.168.0.2:4323192.168.0.4:5000" can get access to the corresponding Transmission Control Block within constant time. This "redundant" design is mainly for time efficiency purpose.

Other attributes include a "verbose" option, which allows user to see what happened in this connection. Since map in Go is not thread-safe, we have to lock TCP when we update `connections` and `socketMap`. Finally, `network` is a pointer pointing to the network layer. Whenever a TCP packet has been wrapped, it will be passed through `network`'s `onRecvTCPData` method.

- Transmission Control Block

```
type TransControlBlock struct {
    sockfd          int
    localAddr       *net.TCPAddr
    remoteAddr      *net.TCPAddr
    state           StateTCP
    iotype          SocketIOType
    send_unack      uint32
    send_next       uint32
    send_window     uint16
    recv_next       uint32
    recv_window     uint16
    read_buf        *CircularBuffer
    write_buf       *CircularBuffer
    retransmit      bool
    dropRate        int
    retransmitQueue *Queue
    lock            sync.RWMutex
    rto             time.Duration
    srtt            time.Duration
    lastSent        time.Time
}
```

The `TransControlBlock` structure follows closely with RFC 793's suggestion. It contains 3 parts. First is the socket pair information(e.g. local and remote address and port, file descriptor, and current state in TCP state machine). Second part is regarding read/write buffer and sliding window control variables. The third part is retransmission queue and retransmission timeout calculation variables.

- Circular Buffer

```
type CircularBuffer struct {
    data               []byte
    size               int
    appReadPtr         int
    window_left        int
    window_right       int
    numRead            int64
    numWritten         int64
    initSeqNo          uint32
    lock               sync.Mutex
    lastGetTime        time.Time
    unorderedPacketMap map[uint32][]byte
}
```

`CircularBuffer` class implements a circular buffer supporting the sliding window algorithm and flow control. It provides two standard buffer I/O methods `Put(data []byte, seqNo uint32)` and `Get(numbyte int)`, and another `PutUnordered` method handling those unordered data. Detail will be specified at later section.

- Queue(lane/queue)

Credits to GitHub user **oleiade**. This thread-safe queue helps me implement retransmission queue easily.

Performance Analysis

I used three different sized file to compute transmission speed.

- **test.mp3** 5185067 bytes (4.94MB)
- **test.pdf** 37629571 bytes (35.88MB)
- **test.zip** 92339236 bytes (88.06MB)

For performance issue, I turned off all printing functions except those related to connection creation/teardown.

Case 1: directly connected node, perfect link(dropRate=0%)

Test File	Test #1	Test #2	Test #3
test.mp3	387ms(12.76MB/s)	427ms(11.56MB/s)	529ms(9.33MB/s)
test.pdf	2.69s(13.33MB/s)	3.25s(11.04MB/s)	3.01s(11.92MB/s)
test.zip	6.46s(13.63MB/s)	7.06s(12.47MB/s)	7.25s(12.11MB/s)

All transmitted files have been `diff` ed with original file and are compeletely and correctly transmitted. The average transmission speed on two nodes connected directly to each other with a perfect link is ~12MB/s, which outperforms the minimum requirement of 8MB/s.

How to simulate:

Node A

```
$ ./bin/node A.lnx
node> recvfile output.pdf 6000
```

Node B

```
$ ./bin/node B.lnx
node> sendfile test.pdf 192.168.0.4 6000
```

where `192.168.0.4` should be node A's IP address.

Case 2: A<->B<->C, where B's dropRate=2%

Test File	Test #4	Test #5	Test #6
test.mp3	425ms(11.62MB/s)	656ms(7.53MB/s)	576ms(8.57MB/s)
test.pdf	8.93s(4.01MB/s)	5.89s(6.09MB/s)	6.58s(5.45MB/s)
test.zip	10.42s(8.45MB/s)	17.72s(4.96MB/s)	17.43s(5.05MB/s)

When intermediate link becomes lossy, the TCP gets slow as data and ACK packets are dropped randomly during the transmission. Moreover, the speed among different test cases has a **larger variance**. This instability may due to the randomness of packet dropping. Function `func (network *NetworkLayer) onRecvIpPacket()` in file `lib/network.go` implements this random packet dropping logic.

How to simulate:

Node A

```
$ ./bin/node A.lnx
node> recvfile output.pdf 6000
```

Node B

```
$ ./bin/node B.lnx
node> lossyforward 2
set faulty node forwarding dropRate to 2 percent
node>
```

Node C

```
$ ./bin/node B.lnx
node> sendfile test.pdf 192.168.0.4 6000
```

where `192.168.0.4` should be node A's IP address.

Unordered packet handling

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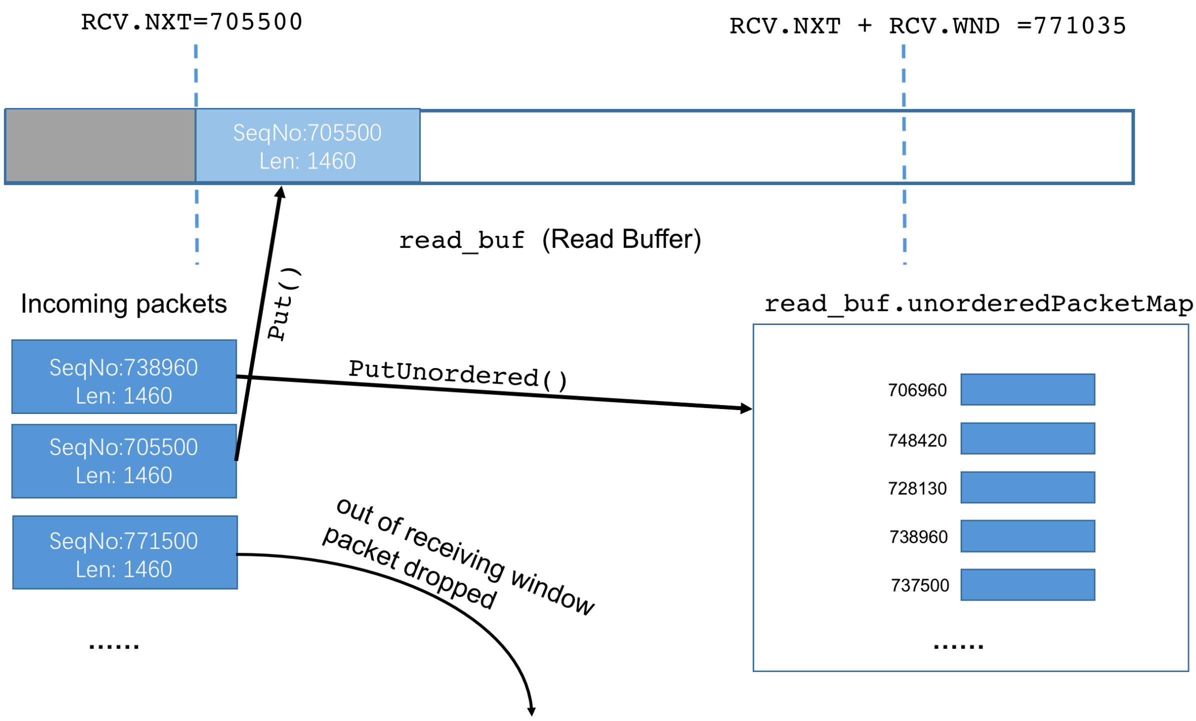
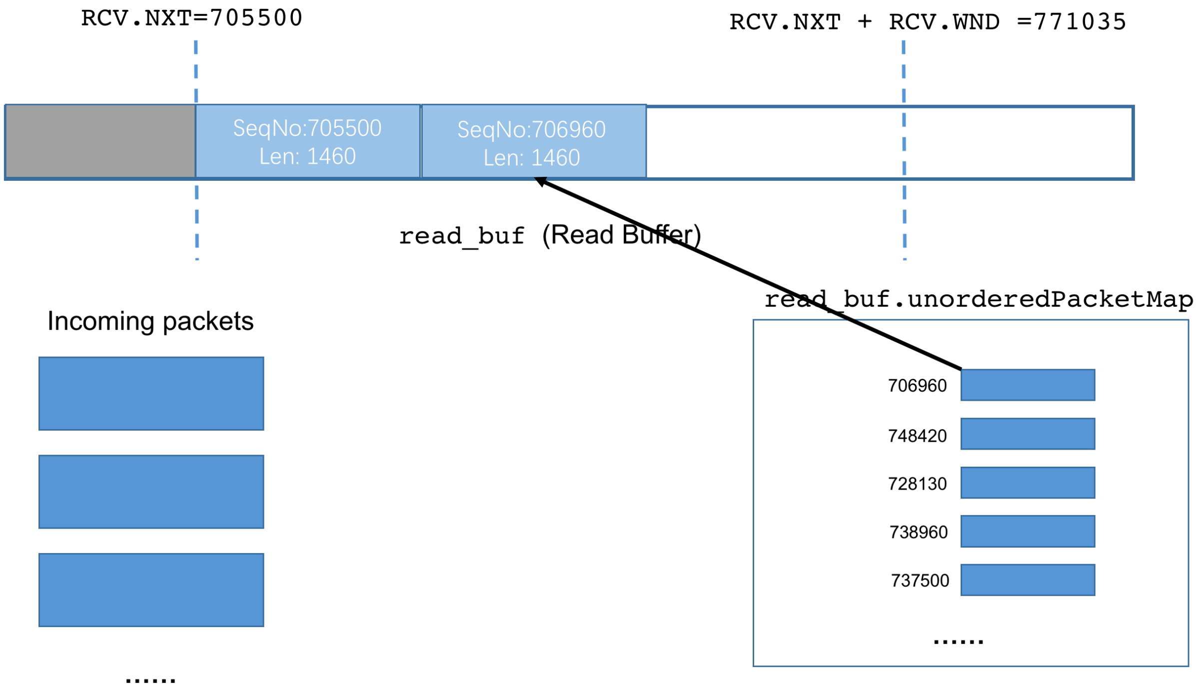


The first figure explains three kinds of incoming tcp packets. Packet receiver expected next, packet within receiving window, and packet which is out of receiving window.

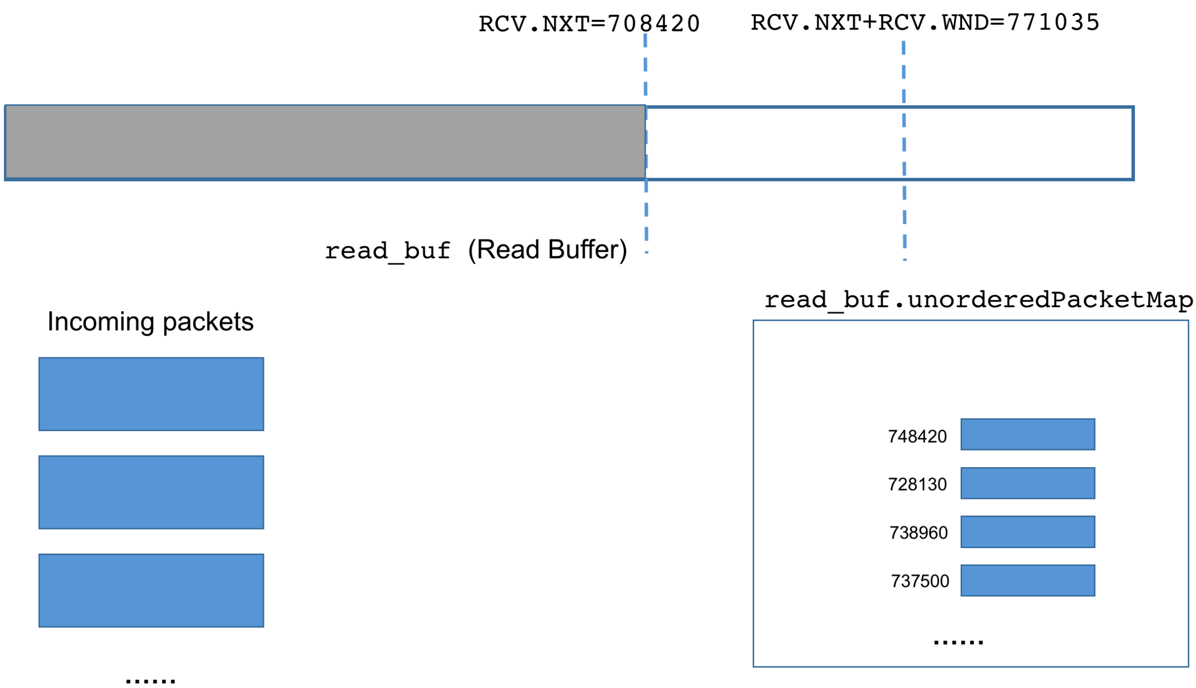
The first case(SeqNo:705500) is ideal situation, where `recv_next == seqNo`. We can simply write it into the receive buffer using `Put()` method.

The second case(SeqNo:738960) is also common. In this scenario, we cannot put it right into the buffer but rather into a temporary map. When all packets between `RCV.NXT` and this segment's `SeqNo` have been received, then read buffer could concatenate all of them together without asking for retransmission. This helps maintain a relatively high speed when intermediate links are lossy.

The third case(SeqNo:771500) happens when receiver is unable to handle incoming data as fast as transmission. Based on TCP's flow control mechanism, those packets will be dropped.



After packet SeqNo 705500 is put onto receiving buffer, it also checks whether its next packet has been stored in the `unorderedPacketMap`. If so, it will recursively concatenate them one by one until there is no more such packet.



This figure shows the result after packet concatenation. `RCV.NXT` has been incremented by the total length of concatenated packets. And `unorderedPacketMap` also removes those unordered packets that have been put into receiving buffer.