Assignment 2 – TCP Reliable Transport Lab

Instructions:

Please complete this assignment by individuals. Both graduate and undergraduate students are expected to complete this assignment.

Getting Started

The objective of this assignment is to give you hands-on experience with TCP. You will implement much of TCP, such that bytes are sent reliably in-order between TCP sockets connected over a TCP connection.

Update Cougarnet

Make sure you have the most up-to-date version of Cougarnet installed by running the following in your cougarnet directory:

```
$ git pull
$ python3 setup.py build
$ sudo python3 setup.py install
```

Remember that you can always get the most up-to-date documentation for Cougarnet here.

Resources Provided

Resources Provided

The files given to you for this lab are the following:

- **buffer.py** - a file containing a stub implementation of a TCP send and receive buffer. This is where you will do your work.
- **test_buffer.py** - a file containing unit tests, which will be used to test your TCP buffer implementations.

- **host.py** - a file containing a basic implementation of a host. Note that this is pared down version of the Host class you will implement in the next assignment in which the send_packet() method simply picks an outgoing interface, creates a frame with the broadcast address as its destination, and sends the frame out the interface. handle_udp() and handle_tcp() methods. You will also do your work here!
- **transporthost.py** - a file containing a basic implementation of a host that has transport-layer capabilities. Note that this is pared down version of the TransportHost class you implemented in the Transport-Layer Lab in which the handle_tcp() simply expects a matching TCP connection to exist and calls handle_packet() on the corresponding socket, a TCPSocket instance.
- **hello.txt, test.txt** and **jhu-y-mtn.jpg** - files that are to be transferred from one host to another to test your reliable transfer functionality.
- **mysocket.py** - a file containing a stub code for a TCP socket. You will also do your work here!
- **scenario1.cfg** - a network configuration file for testing your implementation.
- **scenario1.py** - a script that tests the functionality of your reliable transport implementation.

Helpful Reading

Read Section 3.4 ("Reliable Data Transfer") in the book.

Topology

The file scenario1.cfg describes a network topology in which hosts a and b are directly connected.

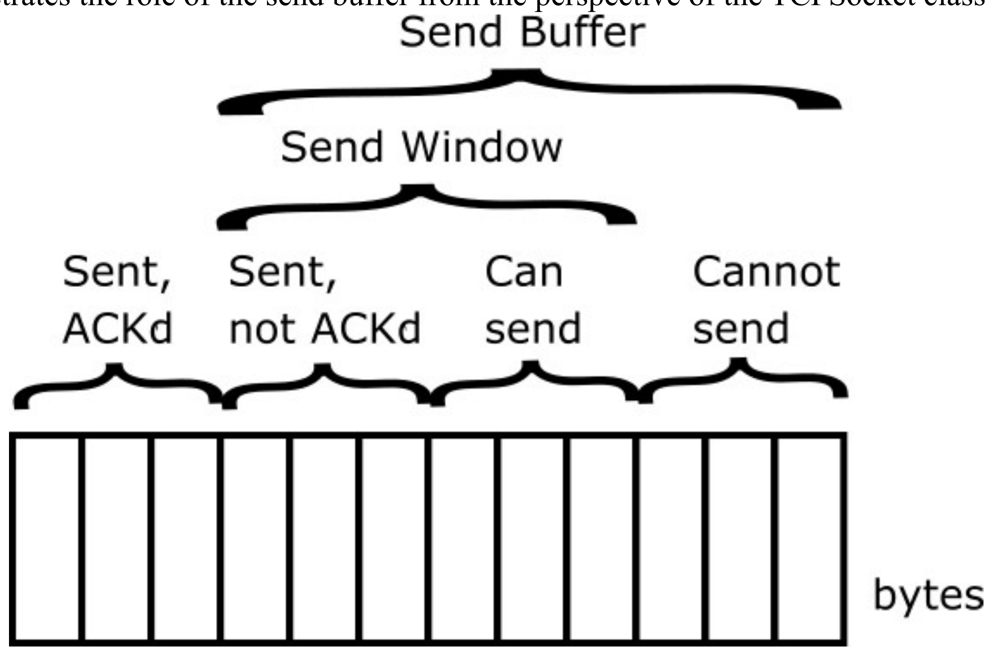
```
+---+
| a |
+---+
  |
  |
  |
+---+
| b |
+---+
```

Both hosts have the basic functionality of taking a packet, encapsulating in an Ethernet frame, and sending it to the appropriate host on its LAN/subnet. Additionally, rudimentary transport-layer multiplexing functionality has been implemented for you. What you will be adding is a reliable data channel for sending data between the two hosts, over an established connection.

Part 1 - TCP Send Buffer

In this part of the lab, you will write the code which is used by the TCP peer to buffer bytes that are intended to be reliably sent, i.e., when TCPSocket.send() is called by an application.

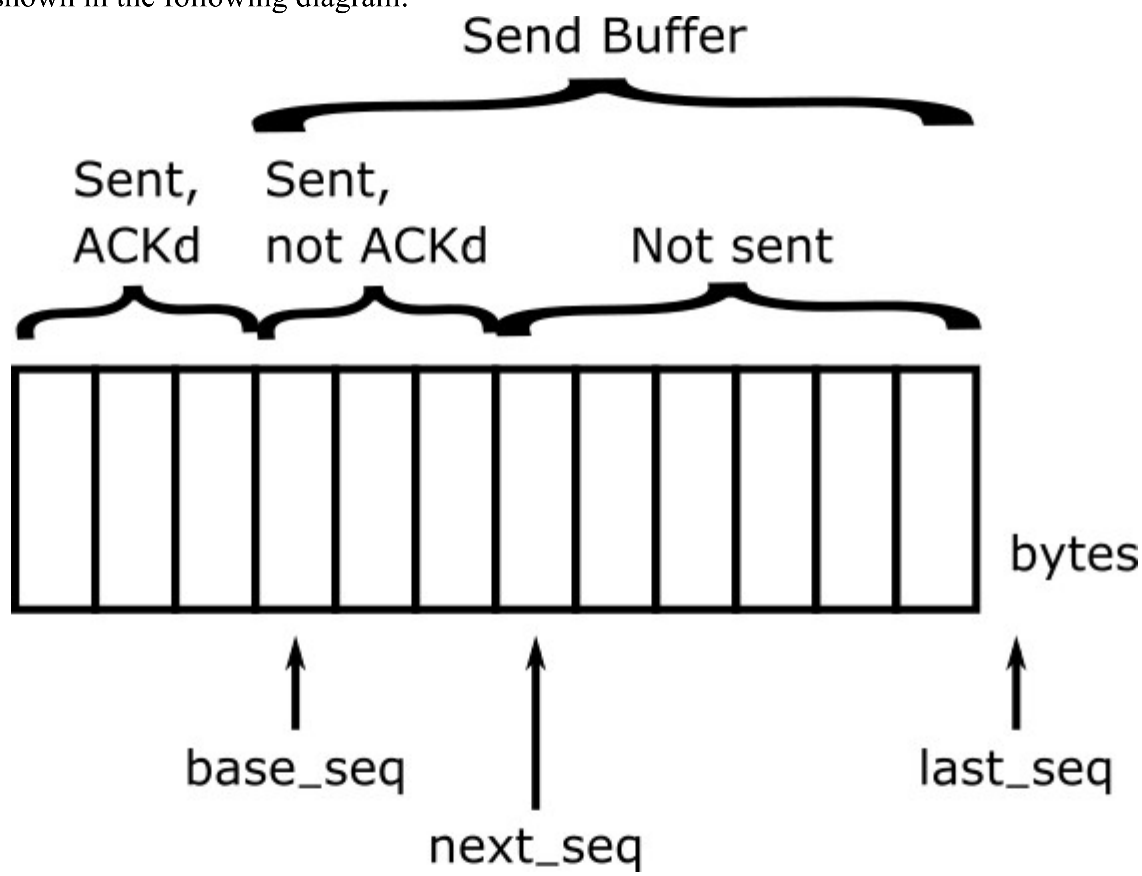
The following image illustrates the role of the send buffer from the perspective of the TCPSocket class.



The TCPSocket class implements the sliding window for reliable delivery, with the help of a TCPSendBuffer instance. However, the TCPSendBuffer class doesn't know anything about the size of the window (which will change over time); it simply keeps track of:

- all bytes that need to be sent; and
- which of those bytes have been sent, but not acknowledged.

Both of these are dictated by the TCPSocket instance using methods that will be shown hereafter. The perspective of the TCPSendBuffer is shown in the following diagram:



Note that in both images the bytes labeled "Sent, ACK'd" are technically not part of the buffer because nothing more needs to be done with them! They are simply shown for continuity.

You can think of the entire buffer as a stream of bytes. Thus, there is a sequence number associated with each byte in the buffer. The three most important numbers to keep track of are:

- base_seq - the sequence number of the first unacknowledged byte in the window.
- next_seq - the sequence number of the first yet-to-be-sent byte in the window.
- last_seq - the sequence number of the byte *after* the last byte in the buffer.

These are what help the buffer identify its current state, including the locations of the "divisions" that are labeled in the above diagram. For example:

- when base_seq = next_seq, then there are no bytes waiting to be acknowledged;
- when next_seq = last_seq, there are no bytes in the buffer that haven't been sent; and
- when base_seq < next_seq < last_seq, there is at least one byte that has been sent but not acknowledged, and at least one byte that has not been sent at all.

Note again that there is nothing for the buffer to know where the boundary of the window is. Thus, in the last case, the buffer does not actually know whether or not the bytes(s) could be sent. *That* is the job if the TCPSocket class. Interfacing with the buffer is therefore a largely a matter of calling methods like put(), get(), slide(), and get_for_resend(), which are explained subsequently.

Instructions

Complete In the file buffer.py, flesh out the following methods for the TCPSendBuffer class.

- put() - This method takes the following as an argument:
    - data: raw bytes (bytes) to be sent across a TCP connection.

With this method data is added to the buffer. This is called by TCPSocket.send(), such that all bytes are initially buffered and only sent if/when there is room in the window.

For example, suppose the following are the values associated with a TCPSendBuffer instance:

```
buffer = b'abcdefg'
last_seq = 1064
```

When put(b'hijk') is called, then those would be changed to the following:

```
buffer = b'abcdefghijk'
last_seq = 1068
```

- get() - This method takes the following as an argument:
  - size: the number of bytes (int), at most, to be retrieved from the buffer.

  This method retrieves (at most) the next size bytes of data that have not yet been sent, so they can be sent using send_packet(). It returns a tuple of (bytes, int), where the first element is the bytes themselves and the second is the starting sequence number. Typically, size is max segment size (MSS). If size exceeds the amount of data in the buffer, then only the remaining bytes are returned.

  This method is typically called in two cases: (1) when data is sent by the application to the socket (i.e., with TCPSocket.send()), and the window size allows for at least some of those bytes to be sent immediately; or (2) when an acknowledgment for new data is received, such that the window can be slid and more bytes sent.

  For example, suppose the following are the values associated with a TCPSendBuffer instance:

```
buffer = b'abcdefghijk' base_seq
= 1057
next_seq = 1061
```

  When get(4) is called, then those would be changed to the following:

```
buffer = b'abcdefghijk' # unchanged base_seq
= 1057
```

next_seq = 1065

And the data returned would be: (b'efgh', 1061)

- get_for_resend() - This method takes the following as an argument:
    - size: the number of bytes (int), at most, to be retrieved from the buffer.

  This method retrieves the next size bytes of data that have previously been sent but not yet acknowledged. Typically size is MSS. If size exceeds the amount of data in the buffer, then only the remaining bytes are returned.

  This is typically called after a loss event, such as timeout or triple-duplcate ACK.

  Note that this method is very much like get(), with the major differences being 1) the *starting* place of the bytes to be returned and 2) the fact that next_seq is not changed after get_for_resend() is called.

  For example, suppose the following are the values associated with a TCPSendBuffer instance:

  buffer = b'abcdefghijk' base_seq
  = 1057
  next_seq = 1061

  When get_for_resend(4) is called, the return value would be: (b'abcd', 1057), and the values of buffer, base_seq, and next_seq would remain unchanged.

- slide() - This method takes the following as an argument:
    - seq: the sequence number returned in the acknowledgment field of a TCP packet, i.e., acknowledging all bytes previous to that sequence number.

This method acknowledges bytes from the buffer that have previously been sent but not acknowledged, opening the door for more bytes to be sent. For example, suppose the following are the values associated with TCPSendBuffer instance:

```
buffer = b'abcdefghijk'
base_seq = 1057
```

When slide(1061) is called, then those would be changed to the following:

```
buffer = b'efghijk'
base_seq = 1061
```

Also flesh out the following utility methods:

- bytes_not_yet_sent() - return an int representing the number of bytes not-yet-sent in the buffer.
- bytes_outstanding() - return an int representing the number of bytes sent but not yet acknowledged.

Testing

The file test_buffer.py contains a suite of tests that use python's [unittest module] to demonstrate the usage and test the functionality of TCPSendBuffer. Run the following to test your buffer implementation (Note that this will also test the functionality of the TCPReceiveBuffer class, which you will implement in Part 2. Thus, you will likely get failures for test related to TCPReceiveBuffer at this point).

```
$ python3 test_buffer.py
```

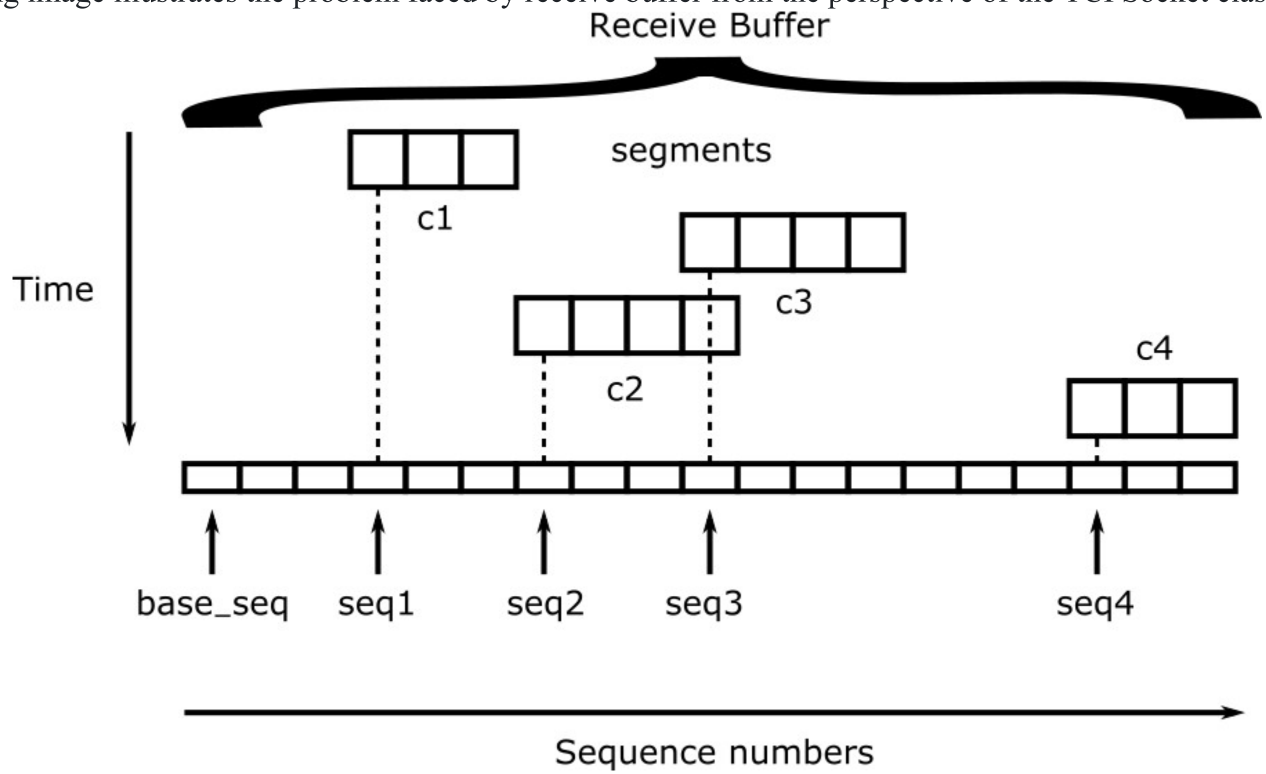
or, alternatively:

```
$ python3 -m unittest test_buffer.py
```

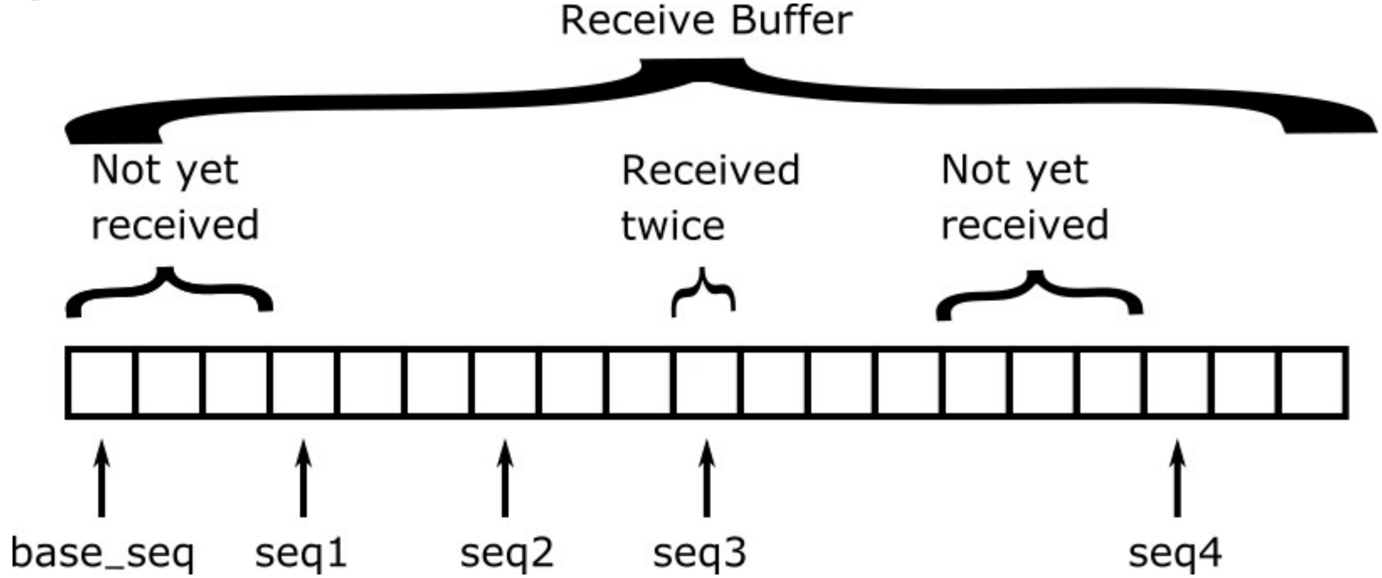
Part 2 - TCP Receive Buffer

In this part of the lab, you will write the code which is used by the TCP peer to buffer bytes that have been received until they represent a continuous set of in-order bytes, suitable for an application to call TCPSocket.recv().

The following image illustrates the problem faced by receive buffer from the perspective of the TCPSocket class.



It receives different segments of data, possible overlapping, possibly out of order, and possibly with holes in between. Each segment has a starting sequence number (i.e., from the seq field of the TCP header accompanying the segment) and a length. In the above example, segments c1, c3, c2, and c4 are received in that order, and they begin with sequence numbers seq1, seq3, seq2, and seq4, respectively. Segments c1 and c4 have length 3, and segments c2 and c3 have length 4. Using their sequence number and length, these segments can be stitched together, with duplicate bytes removed, once all the bytes have been received. For example, a byte-level representation of the receive buffer illustrated above is shown below:



A few things to note in this particular example:

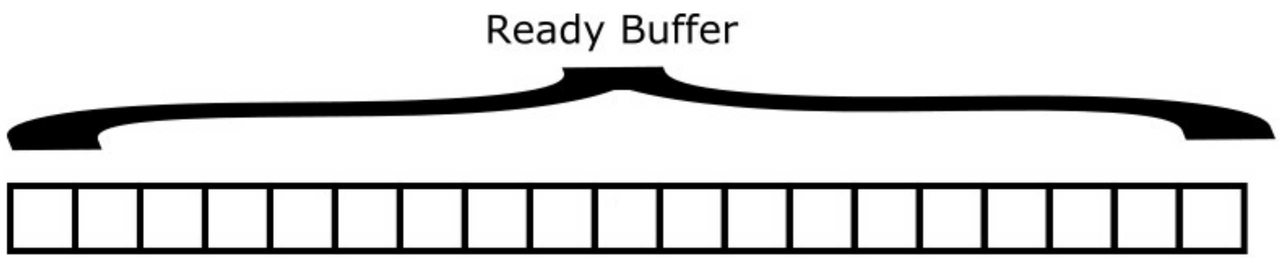
- The first three bytes (i.e., starting with base_seq) have not yet been received.
- Segment c3 was received before segment c2, even though seq2 comes before seq3.
- Segments c2 and c3 overlap by one byte, i.e., the byte at seq3 has been received twice.
- The three bytes immediately before seq4 have not yet been received.

As a result:

- Segment c2 must eventually be put before segment c3, even though it arrived first.

- As soon as the segment(s) containing the first three bytes are received, filling the hole at the beginning, the series of bytes from seq1 through up until the next set of missing bytes, i.e., those immediately preceding seq4.
- The "duplicate" byte by which c2 and c3 overlap must be thrown out.

Once an in-order sequence of bytes is ready, it can be sent to the ready buffer, which is simply a queue of bytes from which a TCPSocket() reads when its recv() method is called.



Instructions

In the file buffer.py, flesh out the following methods for the TCPReceiveBuffer class.

- put() - This method takes the following arguments:
    - data: raw bytes (bytes) that have been received in a TCP packet
    - sequence: the sequence number associated with the first byte of the data

With this method data is added to the buffer. This is called by TCPSocket.handle_data(), such that all segments are initially buffered and only made available to the ready buffer when there is data at base_seq (i.e., no "hole" at the beginning). The suggested implementation is to map incoming segments of data by sequence number in a dictionary.

For example, suppose a TCPReceiveBuffer instance, buf is initialized thus:

```
buf = TCPReceiveBuffer(2021)
buf.put(b'def', 2024) buf.put(b'mn',
2033)
```

The segments might be stored in a dictionary in the instance variable buffer with the following value:

```
{2024: b'def', 2033: b'mn'}
```

The following rules should be applied when adding segments to the receive buffer:

- If a segment is received, and the sum of its starting sequence number plus its length (i.e., the sequence number following this segment) is less than or equal tobase_seq, then ignore it. It is old data.

  For example, if base_seq = 2021, and a segment with sequence 2001 and length 4 is received, it is discarded (i.e.., because 2001 + 4 <= 2021).

- If a segment is received, and its starting sequence number is less than base_seq, but its length makes it extend to base_seq or beyond, then trim the first bytes off, so that it starts with base_seq and is stored in the dictionary accordingly.

  For example, if base_seq = 2021, and data with sequence 2019 and length 4 is received, the first two bytes of the segment are discarded, the remaining two bytes are given starting sequence 2021.

- If a segment arrives with the same sequence number as another segment that has previously been received, keep only the segment that is the longest.

- After every segment is added, iterate through each segment in the buffer, in order. For each segment, consider the segment immediately preceding it. If the length of the preceding segment makes it extend to the sequence of the current segment or beyond, then trim the first bytes off of the current segment, so that its new sequence number corresponds to the first non-duplicate byte. Use the updated sequence number to map the segment.

  Don't forget to delete the old reference to the segment!

For example, if a segment exists in the buffer with sequence 2026 and length 4, and a new segment is received with sequence 2024 and length 3, then the reference to segment with sequence 2026 is removed, and it is replaced with the segment starting with sequence 2027 consisting of the last three bytes of what was previously segment with sequence 2026.

Consider the following example:

```
buf = TCPReceiveBuffer(2021)
buf.put(b'foo', 2001)
buf.put(b'fghi', 2026)
buf.put(b'def', 2024)
buf.put(b'mn', 2033)
```

This results in the following value of buf.buffer:

```
{2024: b'def', 2027: b'ghi', 2033: b'mn'})
```

- get() - This method takes no arguments. It retrieves the largest set of contiguous (i.e., no "holes") bytes that have been received, starting with base_seq, eliminating any duplicates along the way. It updates base_seq to the sequence number of the next segment expected. It returns a a tuple of (bytes, int), where the first element is the bytes themselves and the second is the sequence number of the starting sequence of bytes. This method is typically called by TCPSocket.handle_data(), immediately after put() is called. The idea is to check the buffer immediately after data has been received to see if any is ready to be put into the ready buffer.

To build the string of bytes to be returned, start with the segment that starts with sequence base_seq, and append the data from each segment, in sequence order, until there is a gap in the bytes. Then update base_seq, and return the sequence of bytes along with the previous value of base_seq.

For example, consider the following example:

```
buf = TCPReceiveBuffer(2021)
```

```
buf.put(b'fghi', 2026)
buf.put(b'def', 2024)
buf.put(b'mn', 2033)
```

There are holes in the data starting at sequence numbers 2021 and 2030. Thus, calling get() with the buffer in this state would result in a return value of:

```
(b'', 2021)
```

Note that the sequence number returned should always be the value of base_seq, before it is incremented in the get() method.

But when the following is called, the first hole is filled:

```
buf.put(b'abc', 2021)
```

So now when get() is called again, the value returned is:

```
(b'abcdefghi', 2021)
```

At this point, the member values associated with the TCPSendBuffer instance are as follows:

```
base_seq = 2030
buffer = {2033: b'mn'}
```

Testing

The file test_buffer.py contains a suite of tests that use python's unittest module to demonstrate the usage and test the functionality of TCPReceiveBuffer. Run the following to test your buffer implementation:

```
$ python3 test_buffer.py
```

or, alternatively:

```
$ python3 -m unittest test_buffer.py
```

Part 3 – Reliable Delivery

With a working sender buffer and receiver buffer, you can now create a working implementation of a reliable transport!

First, note that you do not need to worry about connection setup--the TCPSocket objects are instantiated with a state of ESTABLISHED, as if the three-way handshake has already occurred. A special method, bypass_handshake(), is called to initialize the initial sequence numbers of each side of the connection, in lieu of the three-way handshake. This method also initializes the TCPSendBuffer and TCPReceiveBuffer instances associated with the TCPSocket instance, instance variables send_buffer and receive_buffer, respectively.

Second, note that the send() and recv() methods are already implemented for you as well. The send() method is the one that sets everything in motion! It is called by the application when it wants to reliably send data to its peer. It simply puts everything in send_buffer and then calls send_if_possible() to send what it can send immediately given the congestion window. The recv() method simply pulls (up to) the requested number of bytes from ready_buffer, which is just a buffer of bytes (type bytes) that have been retrieved from receive_buffer; receive_buffer does all the heavy lifting of sorting and filling in gaps.

Finally, the transport-layer multiplexing functionality has been implemented for you, in a version of TransportHost with a very basic implementation of TransportHost.handle_tcp() (see the Transport-Layer Lab).

However, his lab requires that you have the TCPHeader and IPv4Header classes fleshed out, as well as the TCPSocket.create_packet() and TCPSocket.send_packet() methods, as directed in the Transport-Layer Lab

A few words about TCP configuration parameters. The Maximum Segment Size (MSS) is 1,000 bytes. However, you can find that in the mss instance variable of the TCPSocket instance (i.e., you don't need to hardcode it). The timeout interval for your

TCPSocket class is always 1 second and is stored in the value of the timeout instance variable. However, the TCPSocket.start_timer() and TCPSocket.cancel_timer() have been implemented for you.

The book (and the TCP spec) mention a delayed acknowledgment. You will not implement a delayed acknowledgment in your implementation; yours will send an acknowledgement for every segment you receive. Note that your acknowledgment may or may not acknowledge new data.

Instructions

Integrate the following code from your implementation of the Lab-Transport Layer:
The TCPHeader and IPv4Header classes from headers.py.

- The TCPSocket.create_packet() and TCPSocket.send_packet() methods from mysocket.py.

In the file mysocket.py, flesh out the following sender-side methods for the TCPSocket class.

- send_if_possible() – This method takes no arguments. Basically, it just grabs segments of data from its TCPSendBuffer instance, i.e., using send_buffer.get(), no larger than MSS (stored in the mss instance var) and sends them to the TCP peer (i.e., using the send_packet() method). It continues doing this until the number of outstanding bytes (i.e., those that are "in-flight" or unacknowledged) exceeds the congestion window size. Any time a packet is sent, if the timer is not already set (i.e., the timer instance variable is None), then it is set using the start_timer() method, which has been implemented for you.

    This method is called in two places: the send() method and the handle_ack() method. The send() method (called by an application) calls send_if_possible() after it has buffered all the data in the send_buffer, so it can immediately send what is allowed by the congestion window. The handle_ack() method (called when a TCP packet with the ACK flag set is received) calls send_if_possible() after new data has been acknowledged. This acknowledgment of new data results in a "sliding" of the send window (i.e., by calling slide()), so more segments can be sent.

- handle_ack() - This method takes the following as an argument:
    - pkt: an IP packet, complete with IP header. Generally, this could be either an IPv4 or an IPv6 packet, but for the purposes of this lab, it will just be IPv4.

With this method, bytes previously sent are acknowledged. The method checks the acknowledgment number in the TCP header and acknowledges any new data by sliding the window (i.e., send_buffer.slide()). It also cancels the timer (i.e., using cancel_timer(), which has been implemented for you) and restarts it (i.e., using start_timer()) if there are still bytes outstanding. This ensures that the timer is always associated with the oldest unacknowledged segment. Finally, it calls send_if_possible() to send any segments that are allowed within the newly-slid window.

This method is called by the TCPSocket.handle_packet() method whenever a packet is received in which the ACK flag is set.

- retransmit() - This method takes no arguments. Its purpose is simply to grab the oldest unacknowledged segment from the TCPSendBuffer instance, send_buffer, and retransmit it. After re-sending the segment, it re-starts the timer, so the timer is always associated with the oldest unacknowledged segment.

  This method is called after a loss event--either after a timeout (i.e., as scheduled by the timer) or a triple-duplicate ACK (i.e., discovered in handle_ack()--see Part 4).

In the file mysocket.py, flesh out the following receiver-side methods for the TCPSocket class.

- handle_data() - This method takes the following as an argument:
  - pkt: an IP packet, complete with IP header. Generally, this could be either an IPv4 or an IPv6 packet, but for the purposes of this lab, it will just be IPv4.

  This method extracts the segment data from the packet as well as the sequence number associated with that segment, from the TCP packet. The segment data is put into the TCPReceiveBuffer instance, receive_buffer, by calling receive_buffer.put(). Then it calls receive_buffer.get() to retrieve the longest contiguous set of bytes that results from receiving the latest segment. Any data retrieved is appended to the ready_buffer instance variable of the TCPSocket instance. A call to TCPSocket.recv() (i.e., by the application) reads and returns data from ready_buffer.

Whenever TCP segment data is received, an acknowledgment should be sent. That acknowledgment will always correspond to the next in-order byte expected from the other side. If there are holes in the data received, then the acknowledgment will be a duplicate; otherwise it will be new. See the return value of receive_buffer.get().

Finally, if new data has been added to the ready_buffer, then notify the application by calling self._notify_on_data().

This method is called by the TCPSocket.handle_packet() method whenever a packet is received in which there is TCP payload data, i.e., a TCP segment.

As you implement the TCP functionality in the methods above, you will find it helpful for each participant in the TCP connection to keep track of both the sequence number that corresponds to bytes that have been received by its peer and the sequence number that corresponds to bytes that have sent by its peer. The former is stored as the instance variable seq, and the latter is stored as the instance variable ack. They are initialized as follows:

```
self.seq = self.base_seq_self + 1
self.ack = self.base_seq_other + 1
```

They should be maintained in the methods that you have fleshed out above. In particular, ack should be updated every time a segment received yields new contiguous data (i.e., see TCPReceiveBuffer.get()). Likewise, seq should be updated with the value of the acknowledgment field, every time new data is received.

Testing

Note: you might see the following error while testing your TCP implementation, either here or later on:

```
BlockingIOError: [errno 11] Resource temporarily unavailable
```

The reason is that the send buffers of the sockets that are used to pass packets between virtual hosts are getting filled up, and the sockets are set up for non-blocking I/O. However, you should be able to ignore the errors, as your system (fortunately) *implements* reliable transport, which means that if some segments are not received, they will be re-sent :)

In short, there is a more elegant way that this might be handled in Cougarnet, but your code should get past this okay.

No Loss

First, test your TCP implementation to transfer the very small file hello.txt over the TCP connection:

```
$ cougarnet --wireshark a-b --vars loss=0,window=10000,file=hello.txt,fast_retransmit=off scenario1.cfg
```

Because it is so small, it will all fit within a single segment (and, of course, in a single window). Check the wireshark output to make sure the sequence, acknowledgment numbers, and segment lengths look correct. Then check that it was actually received properly:

```
$ cat downloads/hello.txt
hello world
$ sha1sum hello.txt downloads/hello.txt
22596363b3de40b06f981fb85d82312e8c0ed511  hello.txt
22596363b3de40b06f981fb85d82312e8c0ed511  downloads/hello.txt
```

(The file in the downloads directory is the transferred version of the file.)

Now, test your TCP implementation to transfer the much larger test.txt over the TCP connection without any loss using a fixed congestion window size of 10,000 bytes:

```
$ cougarnet --wireshark a-b --vars loss=0,window=10000,file=test.txt,fast_retransmit=off scenario1.cfg
```

(Running with --wireshark is optional, but you might find it helpful.)

The file should transfer in no more than a second or two, and it should be in tact:

```
$ sha1sum test.txt downloads/test.txt
e742dc9de5bac34d82117e015f597378a205e5c1  test.txt e742dc9de5bac34d82117e015f597378a205e5c1
downloads/test.txt
```

When this is working, test on an even larger file, jhu-y-mtn.jpg:

```
$ cougarnet --vars loss=0,window=10000,file=jhu-y-mtn.jpg,fast_retransmit=off scenario1.cfg
```

This one should transfer in roughly 12 seconds and should also be in tact:

```
$ sha1sum jhu-y-mtn.jpg downloads/jhu-y-mtn.jpg
6d82cbd6949c0bb89a9071b821bb62ed73a462ff  jhu-y-mtn.jpg
6d82cbd6949c0bb89a9071b821bb62ed73a462ff  downloads/jhu-y-mtn.jpg
```

Finally, transfer the image file again with a larger window size of 50,000 bytes:

```
$ cougarnet --vars loss=0,window=50000,file=jhu-y-mtn.jpg,fast_retransmit=off scenario1.cfg
```

The larger window should cut the transfer time down to about 5 seconds.

Some Loss

Test your TCP implementation to transfer the file test.txt over the TCP connection with a 5% loss rate each direction:

```
$ cougarnet --vars loss=5,window=10000,file=test.txt,fast_retransmit=off scenario1.cfg
```

The file should still transfer properly (i.e., as shown by sha1sum), though it might take 10 - 15 seconds with the timeouts and retransmissions.

When this is working, test on a larger file, jhu-y-mtn.jpg, with lower loss rate and larger congestion window:

```
$ cougarnet --vars loss=1,window=50000,file=jhu-y-mtn.jpg,fast_retransmit=off scenario1.cfg
```

The file should still transfer properly (i.e., as shown by sha1sum), though it might take up to 60 seconds or more with the timeouts and retransmissions.

All Together

When transmissions are working, with and without loss, make sure they are all working with the --terminal=none option:

```
$ cougarnet --vars loss=0,window=10000,file=test.txt,fast_retransmit=off --terminal=none scenario1.cfg
$ cougarnet --vars loss=0,window=10000,file=jhu-y-mtn.jpg,fast_retransmit=off --terminal=none scenario1.cfg
$ cougarnet --vars loss=0,window=50000,file=jhu-y-mtn.jpg,fast_retransmit=off --terminal=none scenario1.cfg
$ cougarnet --vars loss=5,window=10000,file=test.txt,fast_retransmit=off --terminal=none scenario1.cfg
$ cougarnet --vars loss=1,window=50000,file=jhu-y-mtn.jpg,fast_retransmit=off --terminal=none scenario1.cfg
```

Instructions

Implement fast retransmit in your reliable delivery. This means that TCP detects a loss event when there are three duplicate ACKs (meaning the fourth acknowledgment in a row with the same value), and then immediately retransmits instead of waiting for the retransmission timer. Do this only when the value of the fast_retransmit instance variable is True. You should do this work in the handle_ack() method. When fast retransmit is invoked, you do not need to do anything with the running timer.

It is possible that more than four duplicate ACKs are received. These additional duplicates should be ignored. Fast retransmit should only be done once and not repeated until a new, larger acknowledgment is received.

Testing

Test your fast retransmit functionality, but first running the tests without packet loss, to make sure they still work as expected:

```
$ cougarnet --vars loss=0,window=10000,file=test.txt,fast_retransmit=on --terminal=none scenario1.cfg
$ cougarnet --vars loss=0,window=10000,file=jhu-y-mtn.jpg,fast_retransmit=on --terminal=none scenario1.cfg $
cougarnet --vars loss=0,window=50000,file=jhu-y-mtn.jpg,fast_retransmit=on --terminal=none scenario1.cfg
```

Running the tests with packet loss should result in faster transmission times:

```
$ cougarnet --vars loss=5,window=10000,file=test.txt,fast_retransmit=on scenario1.cfg
$ cougarnet --vars loss=1,window=50000,file=jhu-y-mtn.jpg,fast_retransmit=on scenario1.cfg
```

Specifically, the file test.txt should transfer in no more than a second or two, and it should be in tact:

```
$ sha1sum test.txt downloads/test.txt
e742dc9de5bac34d82117e015f597378a205e5c1  test.txt
e742dc9de5bac34d82117e015f597378a205e5c1  downloads/test.txt
```

```
6d82cbd6949c0bb89a9071b821bb62ed73a462ff  jhu -y-mtn.jpg
6d82cbd6949c0bb89a9071b821bb62ed73a462ff  downloads/jhu -y-mtn.jpg
```

Likewise, the file jhu-y-mtn.jpg should transfer in about 5 or 6 seconds, and it should be in tact: $ sha1sum jhu-y-mtn.jpg downloads/jhu-y-mtn.jpg

Finally test with the --terminal=none option:

```
$ cougarnet --vars loss=5,window=10000,file=test.txt,fast_retransmit=on --terminal=none scenario1.cfg
$ cougarnet --vars loss=1,window=50000,file=jhu-y-mtn.jpg,fast_retransmit=on --terminal=none scenario1.cfg
```

Instructions

Add congestion control to your TCPSocket class by implementing TCP Tahoe. If (and only if) the congestion_control instance variable has a value of tahoe, have your implementation do the following:

- Slow start threshold. At the beginning of the connection, set ssthresh to 64000 bytes. Every time there is a loss event, decrease slow start to half of cwnd.
- Slow start. Every time the sender receives an acknowledgment for new data, increment cwnd by the number of new bytes received (see note below). Note that this will typically be equivalent to one MSS, but it is possible for more than one MSS worth of data to be acknowledged (e.g., in the case that a retransmitted segment is acknowledged along with later segments that were received before the lost one was).
- Congestion Avoidance (Additive Increase). When cwnd exceeds or equals ssthresh, begin congestion avoidance with additive increase. Now every time the sender receives an acknowledgment for new data, cwnd is incremented by the bytes_ackd*MSS/cwnd, where bytes_ackd is the number of new bytes acknowledged--again, typically one MSS (see the note below).
- Loss Event (Multiplicative Decrease). When a loss event is detected (a timeout or triple-duplicate ACK), then set ssthresh to half the value of cwnd and set cwnd to one MSS. In the case that half of cwnd is less than the value of MSS, then set the threshold to one MSS instead.

Note: Whether increasing or decreasing the value of cwnd, it should always be a multiple of MSS. Here is a hint for doing that. Rather than directly incrementing cwnd directly, keep a separate variable, (e.g., cwnd_inc), and increment cwnd_inc by the appropriate amount. Whenever cwnd_inc is greater than or equal to one MSS, then add one MSS to cwnd and decrease cwnd_inc by one MSS (you might do this multiple times in a row). If there is a loss event, reset cwnd_inc to zero.

Finally, when the TCP connection starts, the value of cwnd should always be one MSS. However, you do not need to do that explicitly; it is already set for you with the value you pass to window on the command line, as seen in Part 3.

Testing

The file scenario2.cfg is just like scenario1.cfg but additionally allows you to specify a congestion control algorithm on the command line, e.g., tahoe. Also, it gives you more time to begin packet capture on Wireshark.
Run the following command to run the file transfer with no loss:

```
$ cougarnet –wireshark a-b –vars loss=0,window=1000,file=jhu -y-mtn.jpg,fast_retransmit=on,congestion_control=tahoe
scenario2.cfg
```

(Note: This is one line.)

When the file is done transmitting, select one of the packets in the capture window. Then select "Statistics" from the Wireshark menu. Then hover over "TCP Stream Graphs" in the menu that appears and click on "Time Sequence (Stevens"). When the plot appears, you will probably need to click the "Switch Direction" button to get the correct view. Click "zooms" next to "Mouse", and then highlight the rectangle between coordinates (0, 0) (the origin) and (1, 250,000) to zoom in on this region. If you have implemented it properly, you should see the exponential growth associated with the slow start state (beginning at 0) followed by congestion avoidance (starting at around 0.8 seconds). If you need a reminder of the meaning of the Time Sequence plot or whether or not it is showing correct behavior, refer to the Transport-Layer Homework

If everything looks good, click "Save As…", and save the file as tahoe-noloss.png. Now run the following command to test with 0.1% loss:

```
$ cougarnet –wireshark a-b –vars loss=0.1,window=1000,file=jhu -y-mtn.jpg,fast_retransmit=on,congestion_control=tahoe
scenario2.cfg
```

(Note: This is one line.)

When the file is done transmitting, select one of the packets in the capture window. Then select "Statistics" from the Wireshark menu. Hover over "TCP Stream Graphs" in the menu that appears. This time click on "Window Scaling". When the plot appears, you will probably need to click the "Switch Direction" button to get the correct view. The Window Scaling plot shows the observed congestion window over time (Remember the only party that knows the actual value of cwnd is the sender). You should see a sawtooth pattern in the observed window size, with periodic increases followed by drastic reductions to zero. If you look closely, you should also be able to see the slow start and congestion avoidance phases, but they are harder to pick out in this plot (Note that you can also get a view of the Time Sequence graph of this data, by selecting it from the "Type" dropdown).

If everything looks good, click "Save As...", and save the file as tahoe-someloss.png. Make sure the image is of the Window Scaling graph, not the Time Sequence graph.

Finally, make sure that your files transferred correctly and that your implementation still works as expected with Parts 3 and 4.

What to turn in:

Please note that you should submit your assignment on Gradescope. (We will not accept submissions on Canvas.)

Use the following commands to create a directory, place your working files in it, and tar it up:

```
$ mkdir tcp-lab
$ cp buffer.py mysocket.py tcp-lab
$ tar -zcvf tcp-lab.tar.gz tcp-lab
```

If you have done Part 5, please use these commands to also include your image files.

```
$ mkdir tcp-lab
$ cp buffer.py mysocket.py tahoe-noloss.png tahoe-someloss.png tcp-lab $
tar -zcvf tcp-lab.tar.gz tcp-lab
```