
Introduction

This lab is in two parts. Please read through the entire lab description before beginning part 1; there are some parts that are easier to build and make more sense with the whole picture in mind.

Your task is to implement a reliable, stop-and-wait (part 1a) and sliding window (part 1b) transport layer on top of the IP layer. Your implementation will result in a minimal, stripped-down version of TCP called cTCP that runs as user-level code (regular TCP code usually runs in the kernel). To cTCP, you will also add a new congestion control protocol called BBR that was designed by engineers at Google and is reported to be in use at Google. cTCP can fully interoperate with regular TCP implementations. You will implement both the client and server components of cTCP.

Getting Started

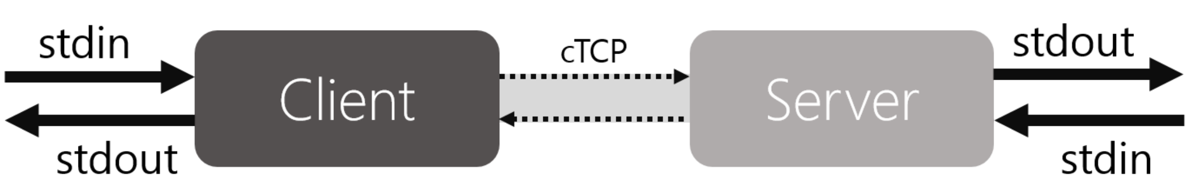
As with other labs, you will use the VM and the git repo provided to you [at the beginning] of the course.

Implementing cTCP requires being able to send IP packets which your code adds a transport (TCP) header to. Doing this in a way that doesn't confuse the operating system requires some modifications to the OS kernel. You must therefore test your code in the virtual machine that we're providing, since that VM contains the necessary code modifications (described below). You may write code elsewhere, but it must compile and be tested using the VM provided. ***Do not upgrade your Ubuntu distribution in the VM***, or you will lose this special kernel and cTCP will stop working.

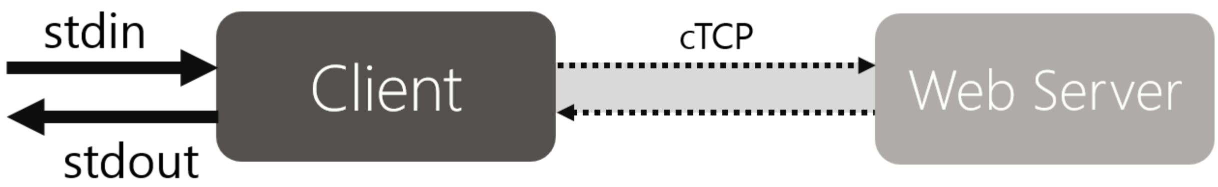
cTCP creates TCP sockets that the operating system is not aware of. When the kernel receives a TCP segment for a cTCP connection, it does not recognize the connection and immediately sends a reset (segment with the RST bit set) in response, telling the other side to terminate the connection. The modified kernel does not send resets in response to unrecognized TCP segments and so allows cTCP, running in a Linux process, to manage a TCP connection.

Part 1a: Stop and Wait cTCP

The pictures in this section show how your cTCP implementation will work after you complete Part 1. The client reads a stream of input data from standard input (STDIN), breaks it up into cTCP segments, and sends the segments to the server. The server reads these segments, ignoring corrupted and duplicate segments, outputting the corresponding data, in order, to STDOUT. Connections are bidirectional: the server also reads from STDIN and sends segments, and the client also receives segments and outputs data to STDOUT. BOTH can be sending data segments at the same time!



Each "side", or direction, of the connection can be closed independently. When both sides of the connection are closed, the client and server clean up the associated state (linked lists, structs, anything being used specifically for this connection). This cleanup is often called "tear down", as in "tear down the connection."



Your implementation is TCP-compliant, which means your client can talk to real web servers! In your client, you can type into STDIN and issue an HTTP GET request to a web server (www.google.com:80 for example):

```
GET / HTTP/1.1
Host: www.google.com
<press enter>
```

And get a response:

```
HTTP/1.1 200 OK
Date: Sun, 17 May 2015 12:00:00 GMT
Expires: -1
…
</script></div></body></html>
```

Although cTCP can interoperate with other TCPs, you do not have to implement the TCP state machine; the starter code implements most of this for you. Your implementation only has to handle connection close events (FIN).

Not all websites will respond. Ones that are known to work include the following:

- www.google.com:80
- yuba.stanford.edu:80
- sing.stanford.edu:80
- www.scs.stanford.edu:80

Requirements

Delivery

- Reliable delivery. One side's output should be identical to the other side's input, regardless of a lossy, congested, or corrupting network layer. Ensure reliable transport by having the recipient acknowledge segments received from the sender; the sender detects missing acknowledgements and resends the dropped or corrupted segments. Duplicate segments should not be outputted.
- Timely delivery. Your solution should be reasonably fast. For instance, it should not take 2 seconds for every segment sent from the sender to show up at the receiver.
- Efficient delivery. Your solution should be reasonably efficient: it should not require gigabytes of memory or send many unnecessary segments. E.g., sending each segment 100 times to protect against losses is not a valid implementation strategy.

Robustness

- Large inputs. It MUST correctly handle inputs much larger than the window size.
- Binary files. It MUST be able to transfer binary (non-ASCII) files correctly. Be sure to test this!
- Retransmissions. cTCP should retransmit a segment FIVE times (i.e. send up to a total of SIX times). No more and no less. After that, it should assume that the other side of the connection is not responsive and terminate and tear down the connection.

cTCP

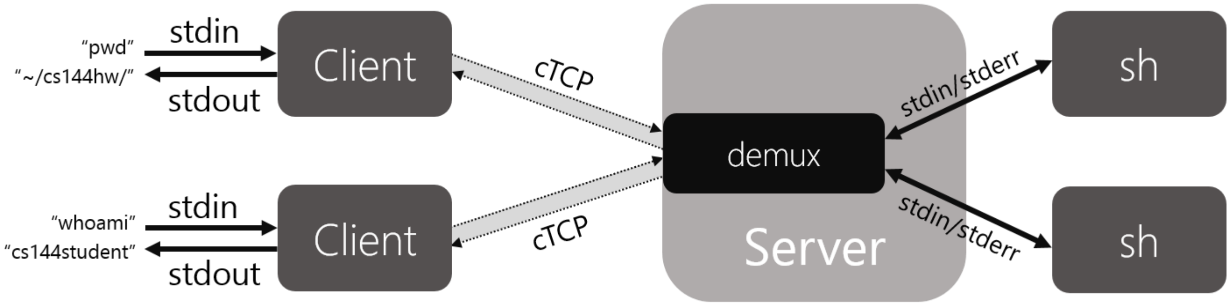
- Connection teardown. You should handle teardown properly. When you read an EOF (Ctrl + D) from STDIN, you should send a FIN segment to the other side to indicate the end of input. When you receive a FIN from the other side, you should acknowledge it, and send an EOF to your output by calling conn_output() with a len of 0. Once all received segments have been outputted and sent segments acknowledged, you may tear down the connection. Like TCP, a connection is not torn down until both sides have finished transferring data, sent FINs, and have their FINs acknowledged.
    - Note that web servers tend to piggyback ACKs on top of FINs, i.e. you will receive segments with both FIN+ ACK flags and should handle these correctly.
    - To be entirely correct, at least one side should wait around for twice the maximum segment lifetime after sending a FIN and an ACK to a received FIN in case that last ACK got lost (the way TCP uses the FIN_WAIT state), but this is not required.
    - When sending an ACK to a FIN segment, the ackno should be 1 greater than the seqno of the FIN. i.e. treat the FIN segment as a "1-byte" segment with no

data.
    - A FIN must also be acknowledged (and retransmitted if necessary).
- Cumulative acknowledgements. Recall that TCP uses cumulative acknowledgements. The acknowledgement field of a segment contains the first byte that a receiver has not yet received (i.e. the byte it is expecting next). Your implementation must follow these semantics so it can interoperate with other TCP implementations. Your cTCP implementation does not have to handle sequence number overflowing/wrapping around in the lifetime of a connection. Your connections MUST always use 1 as the initial sequence number. Remember sequence numbers are incremented in terms of number of bytes, not segments. > Real TCP connections start with a randomly generated initial sequence number for each connection (you will learn later this quarter about how not doing so can make a connection vulnerable to some attacks). However, for simplicity, sequence numbers for cTCP start at 1. The underlying library will do the translation into a randomized sequence number space.
- Stop-and-wait. Part 1a cTCP implements the stop-and-wait protocol with a window size of MAX_SEG_DATA_SIZE(constant defined in ctcp.h). For Part 1a, cTCP needs to support each direction of a connection having only a single segment with data in the network at any time. That segment will be at most (MAX_SEG_DATA_SIZE + headers) bytes in size. It may send out as many acknowledgements (non-data segments) as it wants, without having to wait.
- Piggybacked ACKs. Your cTCP needs to be able to receive and output a segment that has both data and ACK. However, your implementation does not need to send piggybacked ACKs and can output data and ACK segments separately.

Part 1b: Sliding Windows and Network Services

In Part 1b, you will extend cTCP to support multiple client connections to the server, as well as supporting a sliding window of n * **`MAX_SEG_DATA_SIZE`** (defined in ctcp.h), where n >= 1.

In Part 1b, you can initiate several copies of an application on the server side that will output to STDERR. A client can communicate with a server-side application by sending messages to the server, which will pipe the messages to STDIN of the application. The STDERR output of the application is piped back to the server, which will transmit the message back to the client. This lets you hook your cTCP up to programs which act as network services.



Some applications you can test on include (but are not limited to) the following:

- `sh`
- `cat`
- `date`
- `grep --line-buffered "your_string_here"`
- `./test` (found in with the starter package)
    - This is a sample C program that takes input from STDIN and outputs to STDERR. You can modify this to make it whatever you want!
    - To compile this program, you have to run `gcc test.c -o test`

Requirements

Part 1b must support all the functionality required of Part 1a, but also the following additional functionality:

- In-order delivery. Your server and client should ensure that data is written in the correct order, even if the network layer reordered segments. Your receiver should buffer as many bytes as the client may send concurrently.
- Multiple clients. Your server should handle multiple client connections. That means it will need to keep state for each client and separate message queues, as well as other supporting structures.
    - For simplicity, the server can have at most 10 clients connecting to it simultaneously.
    - Note: Connection teardown will not work properly when multiple clients connect. You will not need to handle this case. However, connection teardown should still work if only a single client connects.
- Sliding window. Support window sizes greater than MAX_SEG_DATA_SIZE. You should be able to receive multiple segments at once, each of size at most MAX_SEG_DATA_SIZE + headers. You should buffer the appropriate number of segments for retransmissions and output.

    - Note: A sliding window of size n * MAX_SEG_DATA_SIZE may have more than n segments, if some segments are smaller than MAX_SEG_DATA_SIZE. However, the sum of the data size of these segments is less than or equal to n * MAX_SEG_DATA_SIZE.
    - You should account for the situation when two hosts have different window sizes. For example, host A may have a receive window size of 2 * MAX_SEG_DATA_SIZE and is communicating with host B who has a receive window size of 5 * MAX_SEG_DATA_SIZE.
    - Note: You will not be able to interoperate with web servers with n > 1 (i.e. a sliding window larger than MAX_SEG_DATA_SIZE in size).
- Application support - Your implementation must be able to talk to an application (like sh) started on a server.
    - Note: Connection teardown will not work on the server when an application is running there. You will not be able to type Ctrl + D. However, connection teardown must still work for a normal client-to-server connection.

Implementation Details for Part 1

The Code

One thing to think about before you start is how you will maintain your send and receive windows. There are many reasonable ways to do so, but your decisions here will significantly affect the rest of your code. In Part 1a cTCP only needs to handle one outstanding segment, but in Part 1b it will need to handle a send window as well as a receive window. Cumulative acknowledgements mean the send window can be a simple FIFO queue, but lost segments mean the receive window will need to handle when there are holes. Memory management in systems software is extremely important, so think through where and when you will allocate and free any dynamic structures you need.

Files

Here are all the files that come in the starter code. Feel free to make new helper files as needed, but if you do, make sure to add them to the Makefile. We encourage you to look at the starter code if you have questions on how it works: it is not very long.

- ctcp.h - Header file containing definitions for constants, structs, and functions you will need for the cTCP implementation. Please read this file carefully because the comments are very useful!
- ctcp.c - This is the **only** file you will need to modify. Contains the cTCP

implementation.

- ctcp_linked_list.h/ctcp_linked_list.c - Basic linked list functions. Optionally use this to keep track of unacknowledged segments, etc.
- ctcp_sys_internal.h/ctcp_sys_internal.c - Internal library that handles cTCP to TCP translations, etc. You will not need to look at this code at all and will not be using anything from it.
- ctcp_sys.h - Definitions for system and connection-related functions like reading input, writing output, sending a segment to a connection, etc. You will be using most of these functions!
- ctcp_utils.h/ctcp_utils.c - Helper functions like checksum, current time, etc.
- test.c - Sample application that can be used. Feel free to modify this however you want. It will not be used in grading.
- Makefile - No need to edit unless you're adding new files.
- README - Information on how to build and run cTCP
- ctcp_README - You will fill this out. See Section 6's cTCP README.

Data Structures

Segments

You will be sending and receiving cTCP segments (which are like TCP segments with a few fields taken out for simplicity -- the library will transform cTCP segments into actual TCP segments. You can assume no TCP options will be used.). This struct is defined in ctcp_sys.h.

```
typedef struct ctcp_segment {
    uint32_t seqno;    /* Sequence number (in bytes) */
    uint32_t ackno;    /* Acknowledgement number (in bytes) */
    uint16_t len;      /* Total segment length (including ctcp header) */
    uint32_t flags;    /* Flags */
    uint16_t window;   /* Window size, in bytes */
    uint16_t cksum;    /* Checksum */
    char data[];       /* Start of data */
} ctcp_segment_t;
```

Every segment will have a 32-bit sequence number (seqno), a 32-bit acknowledgement number (ackno), as well as 0 or more bytes of payload (data). All fields (other than data) must be in network byte order (meaning you will have to use [htonl()/htons()]() to write those fields and ntohl/ntohs to read them).

- seqno: Each segment transmitted must be numbered with a sequence number. The first segment in a stream has a sequence number of 1. The sequence number indicate bytes. That means if you send a segment with 10 bytes of data that has a seqno of 10000, then the next segment you send should have a seqno of 10010.
- ackno: Cumulative acknowledgement number. This says that the sender of the segment has received all bytes earlier than ackno, and is waiting for a segment with a sequence number of ackno. Note that ackno is the sequence number you are waiting for that you have not received yet.
- len: Length of the segment. Should include the header and payload data size.
- flags: TCP flags. This should be 0 for normal segments, and ACK for ack segments (you can set its value by doing segment->flags |= ack_flag, where ack_flag is the ACK flag converted to the right byte-order). To signal the close of a connection, you must use the FIN flag. Flags are defined in ctcp.h.
- window: The TCP window size. Should be an integer multiple of MAX_SEG_DATA_SIZE (defined in ctcp.h).
- cksum: 16-bit cTCP checksum. To set this, you must set the cksum field to 0, then call on the cksum()function (defined in ctcp_utils.h) over the entire segment. Note that you shouldn't call htons() on the checksum value produced by the cksum() function -- it is already in network byte order.
- data: Payload to the cTCP segment. This should be empty for FIN/ACK-only segments.

Connection State

The ctcp_state_t structure (in ctcp.c) encapsulates the state of each connection. You should add more fields to this struct as you see fit to keep your per-connection state. Some fields that might be useful to keep include the current sequence number, acknowledgement number, etc. A new ctcp_state_t is created by ctcp_init(), which is called by the library when there is a new connection.

```
    struct ctcp_state  {
    struct ctcp_state *next;   /* Next in linked list */
    struct ctcp_state **prev;  /* Prev in linked list */

    conn_t *conn;      /* Connection object -- needed in order
to
                            figure out destination when sending
*/
```

```
    linked_list_t *segments;  /* Segments sent to this
connection*/

     /* FIXME: Add other needed fields. */
};
```

This ctcp_state_t struct is passed into cTCP functions so it can be used to store received segments, etc. There is also a global list of ctcp_state_t structs called state_list (in ctcp.c) that can be used to figure out which segments need resending and which connections to time out.

Configuration

The ctcp_config_t struct (in ctcp.h) contains values to adjust your cTCP implementation.

```
typedef struct {
        uint16_t  recv_window;    /*  Receive  window  size  (a  multiple  of
MAX_SEG_DATA_SIZE) */
    uint16_t send_window;  /* Send window size (receive window size of other
host) */
    int timer;             /* How often ctcp_timer is called, in ms */
    int rt_timeout;        /* Retransmission timeout, in ms */
} ctcp_config_t;
```

It is passed in when there is a new connection (in ctcp_init() in ctcp.c). You may want to store this struct or its values in a global variable or within the ctcp_state_t struct for later access.

Functions

Functions To Implement

Your task is to implement the following functions in ctcp.c (more details about what they should do can be found in ctcp.h):

- ctcp_init(): Initialize state associated with a connection. This is called by the library when a new connection is made.
- ctcp_destroy(): Destroys connection state for a connection. You should call either

when 5 retransmission attempts have been made on the same segment OR when ***all of the following*** hold:
    - You have received a FIN from the other side.
    - You have read an EOF or error from your input (conn_input returned -1) and have sent a FIN.
    - All segments you have sent have been acknowledged.
- ctcp_read(): This is called if there is input to be read. Create a segment from the input and send it to the connection associated with this input.
- ctcp_receive(): This is called when a segment is received. You should send ACKs accordingly and output the segment's data to STDOUT.
- ctcp_output(): This should output cTCP segments and is called by ctcp_receive() if a segment is ready to be outputted. You should flow control the sender by not acknowledging segments if there is no buffer space for outputting.
- ctcp_timer(): Called periodically at specified rate. You can use this timer to inspect segments and retransmit ones that have not been acknowledged. You can also use this to determine if the other end of the connection has died (if they are unresponsive to your retransmissions).

Helper Functions

The following functions are needed when implementing the above functions (the following functions can be found in ctcp_sys.h):

- conn_bufspace(): Checks how much space is available for output. conn_output() can only write as many bytes as reported by conn_bufspace().
- conn_input(): Call on this to read input that then needs to be put into segments to send off.
- conn_send(): Call on this to send a cTCP segment to a destination associated with a provided connection object.
- conn_output(): Call on this to produce output to STDOUT from the segments you have received. You will first have to call on conn_bufspace() to see if there is enough space for output.

Linked list helper functions can be found in ctcp_linked_list.h. Other helper functions can be found in ctcp_utils.h.

Testing

Building and Running the Code

More specific details on how to build and run can be found in the `README` included with the starter code. To build the code, simply run `make` in the command-line.

Server Mode

```
sudo ./ctcp -m -s -p [server port]
sudo ./ctcp -m -s -p [server port] -w [serer send window multiple] -- [program args]
(Part 1b)
```

Client Mode

```
sudo ./ctcp -m -c [server]:[server port] -p [client port]
sudo ./ctcp -m -c [server]:[server port] -p -w [client send window multiple]    (Part 1b)
```

Debugging

Note: Any debug statements should be printed to STDERR, not STDOUT. This is because STDOUT is being used for the actual program output and it will be confusing for the grader. Your final solution must not output any debugging information (as it will cause tests to fail).

To output to STDERR, use the following function:

```
fprintf(stderr, "Debug message here", ...);
fprintf(stderr, "Here's an int: %d", int_number);
```

We encourage you to use gdb and valgrind. These tools should make debugging your code easier and using them is a valuable skill to have going forward.

Logging

Logging can be enabled with -l on one host to view a log of all segments sent and received:

```
sudo ./ctcp [arguments] -l
```

This will create a .csv file of the form [timestamp]-[port], where timestamp is the timestamp that the host started, and port is the port associated with this host. This should have all the header information you need to track segments and to see if the right fields are set.

The Data field contains a hex dump of the data sent. You can convert to this to ASCII using any hex-to-ASCII converter.

Interoperation

Your program must interoperate with the reference implementation, reference, included in the assignment package. It must also be able to communicate with web servers (such as www.google.com). Some web servers may not respond; however, it must work with google.com.

Unreliability

Since connections between machines are pretty reliable, we've included tools that will help you simulate an environment with bad network connectivity. They allow you to drop, corrupt, delay, and duplicate segments that you send over the network.

To invoke the tools, run your program with one or more of the following command-line options:

```
--drop <drop percentage>
--corrupt <corrupt percentage>
--delay <delay percentage>
--duplicate <duplicate percentage>
```

The percentage values are integers in the range 0-100 (inclusive), and they determine the probability that an outgoing segment will be affected by the associated action. You can use the following flag to set the random seed value:

```
--seed <seed value>
```

For example, running the following command will drop 5% and corrupt 3% of all outgoing segments:

```
./ctcp --drop 5 --corrupt 3 -c localhost:5555 -p 6666
```

If you also include the -d command-line argument, the program will print out more details about the alterations that are being made to your segment.

The tools will alter your segments deterministically. This means that if you run your program twice using the same seed value, the segments will be affected in an identical fashion. This should make it easier for you to reproduce bugs in your code and help you in the debugging process.

Running your program with these flags will only affect outgoing segments. If you would like your incoming segments to be affected as well, you will need to run both instances of cTCP with the command-line flags.

Large/Binary Inputs

Don't assume that your implementation will always work correctly if you've only tested it by typing messages in the console. Try testing with large files (students have used novels from Project Gutenberg), and test with binary executable files as well (cTCP itself!). You can use input/output redirection to pass large files. MAKE SURE you use these options carefully as they overwrite the contents of the file.

```
ctcp-server> sudo ./ctcp [options] > newly_created_test_binary
ctcp-client> sudo ./ctcp [options] < original_binary
```

To verify the transfer is correct, use the cmp utility to check the two files match (read the manual by running man cmp). To look at their contents in detail, you can use od -h.

Memory leaks

Your submission at the end of Part 1b should be free of memory leaks. If there are leaks, we will deduct points for your submission. To test for memory leaks, you can run the following command:

```
sudo valgrind --leak-check=full --show-leak-kinds=all ./ctcp [params for cTCP]
```

Tester

We will provide a tester which will run your program against several tests. This is a Python script that must be run in the same directory as the reference binary and your code. It will assume your code compiles into a binary called **ctcp**, which should be the case if the `Makefile` was not modified. These tests are a subset of the ones we will use for grading.

They are not comprehensive and you should do your own extensive testing. However, passing all the tests we give you is a good indication of your program's robustness. MAKE SURE TO REMOVE ALL PRINT STATEMENTS BEFORE RUNNING THE TESTER!

```
sudo python ctcp_tests.py
sudo python ctcp_tests.py --tests [specific test numbers to run] --timeout [test timeout in secs]
sudo python ctcp_tests.py --tests 1 2 3
```

You can increase the test timeout if some tests are failing (as there is still some randomness and test results may differ slightly with each run). However, if your tests only pass consistently with a large timeout (longer than 20s), then there is a problem with your code.

To get a list of all tests:

```
sudo python ctcp_tests.py --list
```

The script will print out your score for these tests.

Testing on real topologies

The tests above do not run CTCP on a real network topology. After passing (most) of the tests in `ctcp_tests.py` you should try to run it on (a) a dumbbell topology on mininet, and (b) the mini-Internet you defined in Lab 2.

Dumbbell topology. A dumbell topology has two routers connected to each other in the middle, with 2 or more clients attached to one of the routers, and 2 or more servers attached to the other router. Then, we can test ctcp between each client server pair, or between two clients and the same server. If you pass the previous tests, then, your CTCP should pass the following test:

```
sudo python ../tester/generic_tester.py ctcp_mininet.xml
```

Take a look at the XML file, which describes what tests we have defined.

Mini-Internet topology. At this point, you should be able to run CTCP on the mini-Internet you set up in Lab 2. To try this, do the following:

```
cd ../lab2

sudo python ../tester/generic_tester.py ctcp.xml
```

This test runs a CTCP client on a node in the West AS, and a server on a node in the East AS, and downloads a large file across the network. The packets flow through the simple router that you developed. If you pass this test, you should feel proud: you have built your own version of the Internet!

FAQ

Most questions and answers can be found on Piazza. This is merely a list of logistical questions.

1. Should ACKs be piggybacked on top of outgoing data segments?
    - Piggybacking ACKs is preferable but not required and you will neither gain nor lose points either way. However, you do need to handle receiving acks + data from reference and webservers.
2. Can we assume that the len field in ctcp_receive() is correct?
    - No, you must examine the length field of the segment and should not assume the cTCP segment you receive is the correct length. The network might truncate or pad segments.
3. Why is a double pointer prev used for the ctcp_state_t linked list?
    - This is a clever trick that enables it to delete elements of a linked list efficiently with a bit less pointer rewiring than if prev were a single pointer.
4. What sliding window protocol are we supposed to implement for Lab 2?
    - You MUST use selective repeat.
5. Why do all commands have to be run with sudo?
    - cTCP uses raw sockets, which require root in order to open. You can avoid having to type in sudo each time if you run the shell as root (sudo bash).
6. What's the difference between input/receive and output/send?
    - Input/output is done locally, whereas send/receive is done over a connection.
        - Input is the term we use for text/binary received on a client via STDIN.
        - Receive is when a segment is received from a connection.
        - Output is the text/binary printed out to STDOUT.
        - Send is when a segment is transferred from one client to another.
7. What should I do if an error occurs (e.g. conn_output() returns -1)?
    - It's up to you. We suggest terminating the connection (via a call to

ctcp_destroy()).

8. How do you get the EOF to show up?
    - The reference solution does this by doing fprintf(stderr, ...) when it reads in an EOF (Ctrl + D) from STDIN. You can add this to your code too.
9. How come when connecting to a webserver, the data I send is 1 byte bigger than what I expect?
    - Webservers expect messages that end with \r\n; however, this is difficult to input into the terminal. When communicating with webservers, the library automatically converts all \ns to \r\ns (hence the 1 extra byte).
10. Will I need to handle overlapping segments?
    - No, you will not need to handle the case when segments overlap (i.e. a new segment has bytes that partially overlap with the bytes of the previous segment).
11. Why am I getting valgrind memory leaks from the starter code (e.g. in convert_to_ctcp() or start_client())?
    - Make sure you are freeing all segments passed in from ctcp_receive() and freeing the ctcp_config_t struct passed into ctcp_init().

Part 2: BBR Congestion Control

BBR is a new congestion control protocol developed for the Internet. It is described in this paper. The paper also contains sample pseudo-code for BBR which should help you implement BBR. Finally, the patches to the Linux kernel for BBR are also publicly available, which you can take a look at.

You are expected to implement add BBR to cTCP, both to the client and the server side.

Understanding BBR

Two main functions of traditional TCP: 1. Reliable packet transmission. 2. Congestion control. As we studied in class, TCP's native congestion control has some shortcomings:

1. Packet loss ≠ Congestion: TCP reduces its congestion window (cwnd) when packet loss is detected then TCP will assume that current network is congested. However, in reality, packet loss is not only caused by congestion but also network error (e.g. wrong checksum, noise, etc). Therefore, some packets lost doesn't mean current tunnel is crowded. But due to TCP's congestion avoidance mechanism, the cwnd will be decreased so TCP cannot fully utilizing the tunnel. Therefore TCP doesn't fit perfectly with today's data center networks, which usually have high bandwidth and frequent packet loss.

2. No packet loss ≠ No congestion: If there's few packet loss, traditional TCP's cwnd will fast increases then occupies all the buffer space on the receiver side. This indeed fully utilizes the bandwidth, but leaves a long queue for receiver to process, which increases RTT. When TCP was invented (1980s), the receiver buffer is quite small so filling up the buffer is not a big issue. However, today's hardware supports GB-level buffer. If we still use the aggressive policy to fill the buffer, RTT will be unacceptable.

BBR does congestion control differently:

1. Decoupling "reliable packet transmission" and "congestion control" in TCP. BBR lets TCP only work on the first part and BBR estimates the congestion control policy.
2. Using Bandwidth-Delay-Product (BDP) to estimate the traffic condition, i.e. the congestion condition, instead of separately estimating packet loss or delay.
3. Trying to fullying utilize the bandwidth while keeping buffer occupation low, which minimizes the RTT.
4. Besides cwnd, BBR uses a new concept, pacing, to control burstiness in sending packets.

BBR Logic

Since packet loss is a bad indicator of congestion condition, we need the "true" indicator, BDP. Therefore, we should accurately estimate max(bandwidth) and min(RTT). The first condition guarantees that TCP can run at 100 percent bandwidth utilization. The second guarantees there is enough data to prevent receiver starvation but not overfill the buffer. These two values change separately.

BBR computes RTT like this: When the sender received a ACK for a previous sent packet. Then the RTT will be estimated as:

min_RTT = min(min_RTT, cur_time - pkt_sent_time)

Similarly, BBR computes bandwidth as:

max_BW = max(max_BW, (data_delivered - data_before_pkt) / (cur_time - pkt_sent_time))

Note that if the data rate on the sender side is application-limited, the calculated bandwidth will be less than actual bandwidth.

With min_RTT and max_BW, BBR is able to calculate the BDP, the remaining space in the tunnel:

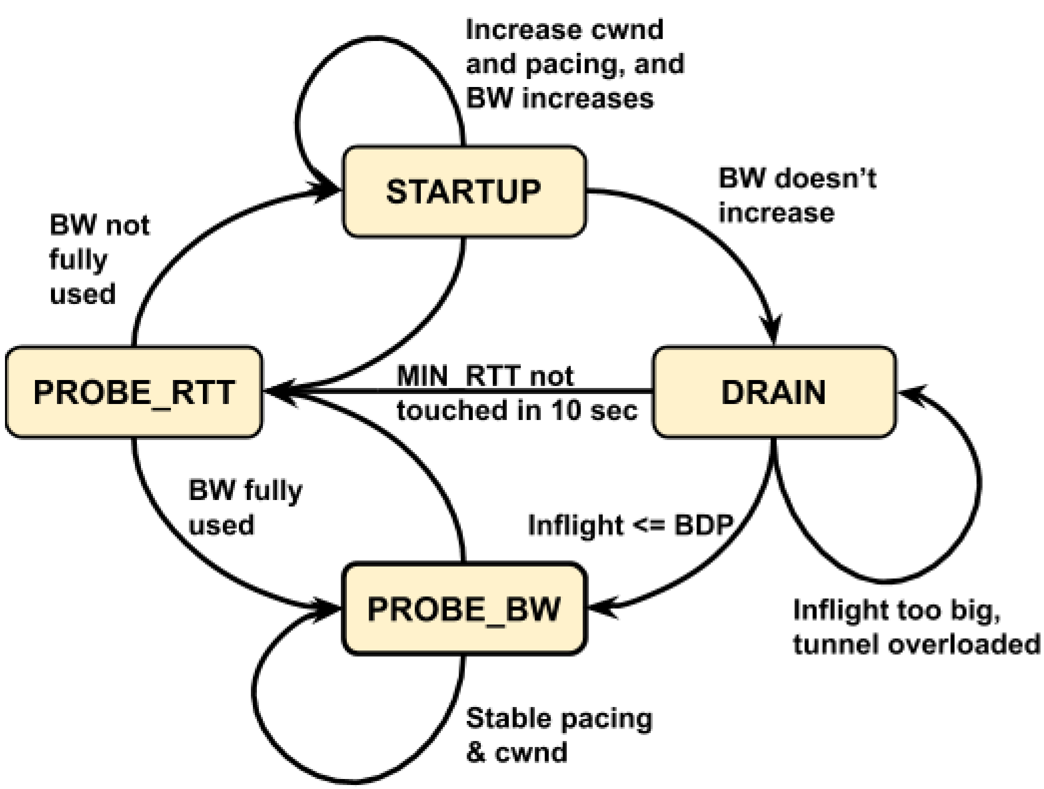
BDP = min_RTT * max_BW

By comparing BDP with inflight (the data sent but not acked) and monitoring min_RTT and max_BW, BBR will know which state it is currently in so it can adjust the sending strategy, which is controlled by cwnd and pacing rate. Cwnd defines the maximum data sender can send within a period of time. Pacing, added by BBR, is used to tell the sender how it should send this piece of data. This is important because there may be burst if the cwnd is big and the sender sends the data altogether. The details of choosing cwnd and pacing are well explained in the BBR code patch.

The paper doesn't discuss fairness between standard TCP and BBR in different situations. You can test in your code if you are interested.

BBR Code Organization

You should first look at the main function of BBR patch: ***bbr_main***, which operates a state machine that can be described as the graph below:



Four states explained:

- **STARTUP**: exponential growth to quickly fill pipe (like slow-start), will stop growth

when bw estimate plateaus, not on loss or delay
- **DRAIN**: drain the queue created in STARTUP stage
- **PROBE_BW**: cycle pacing_gain to explore and fairly share bandwidth
- **PROBE_RTT**: occasionally send slower to probe min RTT

All the implementation details are well explained in the inline comments of the patch code.

Incorporating BBR to your CTCP implementation

After understanding BBR code, you can add it to your ctcp implementation. While you are encouraged to look at the BBR kernel code patches, please **do not** simply copy the code patches. Rather, try to rewrite the code yourself, so you can understand better how BBR works.

We require you to write your BBR code in separate files. To do this, create two files ***ctcp_bbr.h*** and ***ctcp_bbr.c*** under the same directory as the rest of the CTCP code. Add all your BBR code to these files. By including ctcp_bbr.h in ctcp.c, you can use BBR functions.

Testing BBR

We do not provide a tester for BBR. You should try to create your own test cases for BBR to make sure your submission works correctly. In particular you should try the ctcp tests described above: with packet losses, and for large binary inputs. You should so try to draw graphs similar to the ones in the BBR paper (e.g., BBR's estimated BDP vs time) to understand if your implementation works correctly. You can also create simple topologies in Mininet, such as the dumb-bell topology discussed in class, to test whether BBR behaves correctly in the presence of multiple flows.

Every time your code runs, it should output a log file that contains the timestamp and BDP. The file name should be "bdp.txt" and the format of each line is "timestamp BDP". When your ctcp sends a packet, it should append a new line to the file with current timestamp (get it from ***current_time()*** in ***ctcp_utils.h***). You only need to log BDP for sending so you don't need to log for re-sending. The BDP should be **expressed in units of bits.** This will help us generate graphs to test if your implementation is working correctly. Here is what your output should look like:

...
1508350555362 95820
1508350555389 96181
1508350555408 96374
...

Sample Topology and Scripts

Topology

(This part should be updated, please wait) To test your BBR logic, we provide a script, topo_test.py, to build simple topology, including dumbbell and line networks, in Mininet. Another script, simple_server.py, creates a regular TCP server that keeps receiving from a cTCP client. Your cTCP+BBR will act as a client and keep sending data to the server. You can send a large file so there is chances to get congestion. Below are the steps to use the scripts:

1. Disable networking and wifi of your Ubuntu VM (this is important)
2. Download topo_test.py and simple_server.py into your ctcp folder
3. In ctcp folder, run
   `sudo mn --custom ./topo_test.py --topo simple`
   Then you should go into mininet screen (If you cannot see mininet screen after this command, restart your VM).
4. In mininet: `xterm h1 h2`
5. In h2, start the server using
   `python simple_server.py 10.0.0.2 12345 outputfile`
6. In h1, check the ip address of h1-eth0 using `ifconfig`. If it is not 10.0.0.1, manually set it with `sudo ifconfig h1-eth0`. Then run the client using
   `sudo ./ctcp -c 10.0.0.2:12345 -p 11111`

You should see that cTCP can connect to the server and you can transmit data. Note that topo_test.py contains 3 networks (simple, dumbbell, line). You can select a network by specifying the –topo argument. Further, you can set link bandwidth and delay by adding link argument --link tc,bw=10,delay=10ms when you start Mininet. Note that a cTCP client must be run on an odd host has, and a simple server must be run on an event host.

Checksum issue

Mininet may cause checksum error when it is used to test cTCP. If you experience the checksum error at the client side, you can disable checksum in your cTCP code and report the issue in your report.

Plotting BDP

A log file, bdp.txt, can be used to evaluate your BBR implementation. We provide a simple script, [lab4_plot_bdp.py](), to plot BDP of your BBR implementation.

Submission

You will submit Part 1 and Part 2 separately. For each part, you will need to prepare a Report.pdf file and submit your code by committing to your git repository. Please name these files Report_1.pdf and Report_2.pdf.

Report_1.pdf and Report_2.pdf

Each report should be 2-3 page document (a page being 30-40 lines) with no more than 80 characters per column to make it easier to read. This file should be included with your submission. It should contain the following sections:

- Program Structure and Design - Describe the high-level structure of your code, by insisting on what you actually added to the code. You do not need to discuss in detail the structures that you inherited from the starter code. This should be the longest and most important part of the README. Use this as an opportunity to highlight important design aspects (data structures, algorithms, networking principles) and provide greater detail on those areas for your grading TA to understand. You are strongly encouraged to make this README as readable as possible by using subheadings and outlines. Do NOT simply translate your program into an paragraph of English.
- Implementation Challenges - Describe the parts of code that you found most troublesome and explain why. Reflect on how you overcame those challenges and what helped you finally understand the concept that was giving you trouble.
- Testing - Describe the tests you performed on your code to evaluate functionality and robustness. Enumerate possible edge cases and explain how you tested for them. Additionally, talk about general testing strategies you used to test systems code.
- Remaining Bugs - Point out and explain as best you can any bugs that remain in the code.

Submission

When you are ready to submit Part 1, you should add Report_1.pdf to the **`lab3`** folder, then do the following:

```
git commit -a -m 'Lab 3 Part 1 submission'
```

```
git push
```

When you are ready to submit Part 2, you should add Report_2.pdf to the **lab3** folder, then do the following:

```
git commit -a -m 'Lab 3 Part 2 submission'
git push
```