Protocol X
Revision 1.5

Ivan Beschastnikh

June 5, 2005

Abstract

Protocol X is a Transport Layer Protocol:

Over UDP
Bidirectional
Congestion controlled (TCP friendly)
Adjustable reliability

1 Introduction

X is connection based, bidirectional protocol that works on top of UDP. The protocol deals in datagrams that hereafter are referred to as frames. One intended design goal of the protocol is to transport data with as little connection tracking overhead as possible. Inspired by DCCP, the protocol is semi-reliable, a notion that requires a more in depth description of how exactly the protocol functions.

2 Protocol Overview

Under protocol X, a single connection requires two endpoints. The first is an initially passive endpoint we term as server, this endpoint listens for incoming connections. The second endpoint is initially active as it connects to the server, we term it the client.

The first exchange is a three way handshake. Thereafter the disctinction between server and client is lost due to the bidirectional nature of the protocol and we talk about the Sender and Receiver instead, or Endpoint if we don't wish to differentiate. Therefore, from part of the protocol responsible for data exchange, it doesn't matter who was the initiator or the accepter of the connection.

Data received by an endpoint from the application level is handed down discretely in the form of what we will refer to as frames, it is not streamed. The frames are then broken down into fragments which are then payloaded onto packets and sent off to the Receiver.

There are size limitations on frames and fragments. In particular, fragment size and other parameters particular to an endpoint must be exchanged during the handshake. The fragment size unless specified differently by the application is set to the MTU for the data path.

It is the receiver's duty to send back acknowledgements (Acks) within a specified time period which we call `TTAW` (Time to Ack Window). These packets carry information relevant to congestion control (CC), such

as the number of packets missed since the previous Ack. These Acks are not used to determine specific lost fragments so as to retransmit them, they only determine the Sender's sending rate and help establish Flow Control on the Receiver's side.

Having completed a frame, the Sender may initiate the transfer of another frame. Upon detecting that the Sender has transferred the whole frame, the Receiver computes the ratio of the number of bytes transferred to the total size of the frame and compares this number to the `RELIABILITY` constant that has been initialized at start up. If the value is below the reliability threshould, the frame is discarded. Otherwise, the Receiver sends an acknowledgement vector to the Sender. The coefficient is in the range [0, 1], where a 0 indicates a completely reliable connection and a 1 indicates that we never want to resend a fragment to recover from fragment losses. Note that in the last case, acks will still be sent back in order to maximize throughput with congestion control, but lost fragments will not be resent.

This will let the Sender know which packets are missing from the frame on the Receiver's side. The Sender will then proceed to resend only these packets in the same manner as before, receiving acks from the Receiver for CC. This process repeats until the frame is transferred completely, at which point the Receiver will pass the frame to the application level.

The protocol satisfies the invariant that every frame is either delivered in full or dropped completely depending on the comparison of percentage of bytes transferred in the initial attempt to the `RELIABILITY` constant.

At any given moment, there can be a number of incomplete frames. The sender prioritizes the frames based on their age, the older the frame, the higher its priority. A connection may be terminated immediately or by waiting on an acknowledgement by any one of the endpoints at any time.

The protocol also implements a tcp friendly congestion and flow controls.

The remainder of this document will describe protocol X in detail. The organization is structured around connection states, packet types, and congestion and flow controls.

3 Connection States

Table 1 shows the connection states for protocol X. The S is the Server and C is the Client. The connection states determine the types of packets that can be sent between the end points at any particular point of time. Their organization guides the connection and logically divides it into relatively independent units that are easier to analyze separately.

State ID	State Name	Description
1	Disconnected	Initial state for S and C
2	Listening	S's state while listening for incoming connections
3	CConnecting	C's state while trying to connect to a Listening S
4	SConnecting	S's state after a reply to a Connecting C
5	Connected	S's and C's state when both have received ack of the other's acceptance of connection
6	Disconnecting	S's or C's state while waiting on the other to ack a disconnect

Table 1: State names and descriptions

4 Packet Types

Protocol X has various packet types. The type of a packet determines the type of information it carries as well as the state of the sender and sometimes the state of the receiver as seen by the Sender. Some of

the packets specify transitions between connection states, others, such as Ack and Sync packets are used to facilitate the data exchange, whereas the data itself is transfered in the Data and DataAck packets. Table 2 lists all packet types with the state ids in which Receiving and Sending them makes sense.

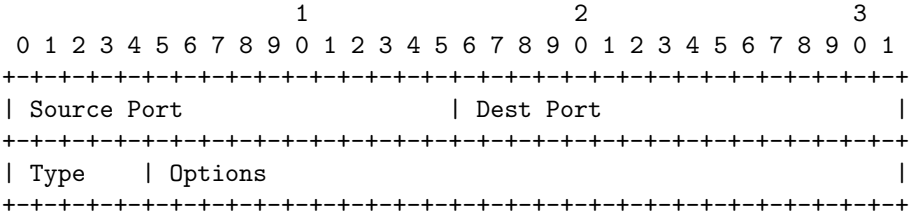
ID	Type	Receiving States	Sending States	Description
0	Request	2	1	Sent by C ; initializes a connection with S ; 1/3 Handshake
1	Response	3	4	Sent by S ; ack of the Request packet by C ; 2/3 Handshake
2	Data	4,5	4,3,5	Transmits application data
3	DataAck	5	4,5	Transmits data with acks
4	Ack	4,5	4,5	Transmits pure acks ; Might include a vector of acks
5	CloseReq	3,4,5	3,4,5	A request to disconnect ; sent by C or S ; elicits a Reset Packet
6	Reset	3,4,5	3,4,5	A reset (kill) of the connection
7	Sync	5	5	Used to resynchronize seq. numbers after bursts or loss

Table 2: Packet types and descriptions

Whenever an endpoint receives a packet that is not in the set specified in the table for the current state, they are to ignore it. Likewise endpoints should not send out packets types in a state other than those specified for each packet type.

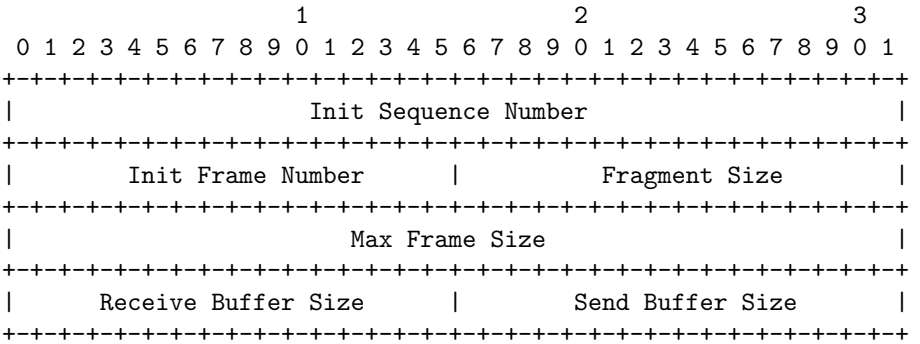
The rest of this section decribes the packet types and the information they carry in detail. It might be useful to skip this section and refer back to it when reading the rest of the document.

4.1 Generic Header



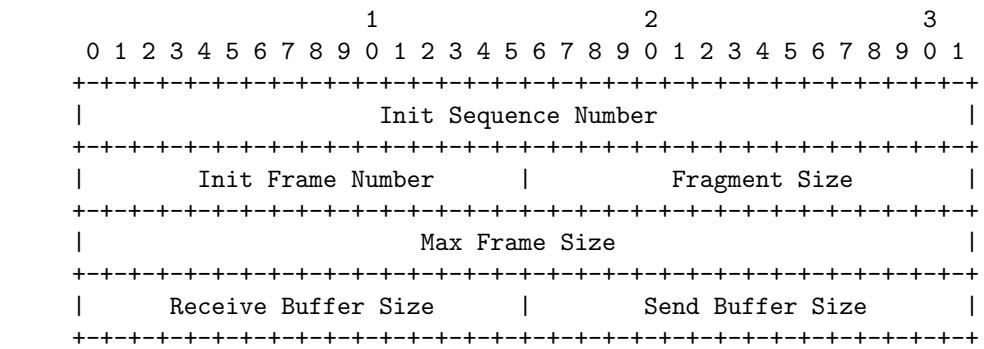
Every packet in protocol X has a generic header to which the formats outlined in the sections below are postfixed. With each packet we identify a packet type (see Table 2), the source and destination ports, and a space for options that may be interpreted differently depending on the type of the packet.

4.2 Request Packet



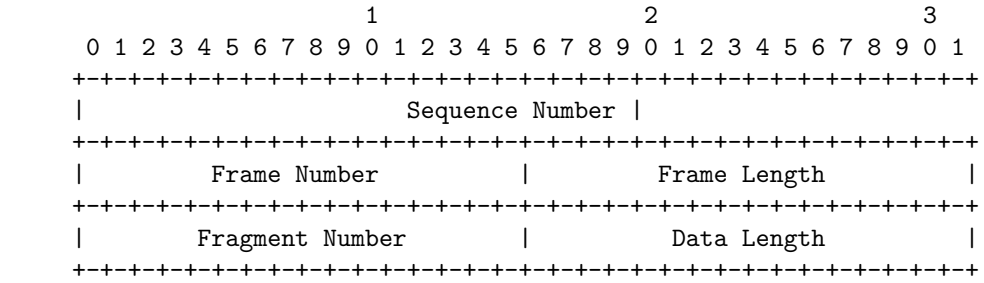
A request packet is sent by the client and initiates connection with the server endpoint. This packet parametrizes the connection to follow. Table summarizes the parameters sent by the client on a Request packet

4.3 Response Packet



Sent by the server endpoint the Response packet is a reply to the Request packet. It must be sent back as soon as possible as the client's initial `RTT` estimate will come from the pair of Request and Response packets. The Response packet contains the same set of parameters as the Request packet and is identical in format.

4.4 Data Packet



Data packets carry frame fragments. With each Data packet we associate three numbers: sequence number, frame number, and a fragment number.

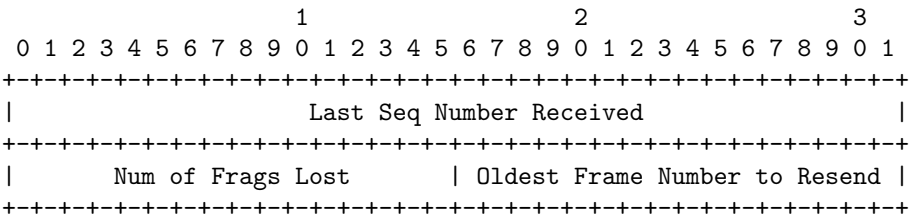
The fragment number cannot exceed `MAX_FRAME_SIZE/FRAGMENT_SIZE` and identifies the fragments location within a frame. Frame numbers and sequence numbers are incremented modulo the maximum number of frames and packets respectively. The sequene numbers are only used to detect losses, and each endpoint maintains its own sequence number independent of the other endpoint, the initial values for these are exchanged with the Request and Response packets.

4.5 Ack and DataAck Packets

There are three distinct forms of acknowledgement. The first is congestion control acknowledgements, which we will term CC_Acks. The second type of acknowledgement is a negative vector acknowledgement, which we will term Vec_Ack. The last is a synchronization acknowledgement which we term Sync_Ack. The three forms play very different roles in the protocol. CC_Acks are acknowledgements sent every so often for congestion control. Vec_Acks are sent after an incomplete frame is transferred, Sync_Acks are responses to Sync packets that synchronize the connection. The DataAcks are nothing more than Data packets with prefixed CC_Ack information. All of this is summarized in the following table.

Ack Packet Type	Description
CC_Ack	Sent every TTAW seconds so that the sender may adjust its sending rate
Data_Ack	Carries Data and a CC_Ack
Vec_Ack	Acknowledges a frame and lets the sender know which pieces the receive needs resent
Sync_Ack	Prompted by a Sync packet, used for estimation of RTT by the receiver

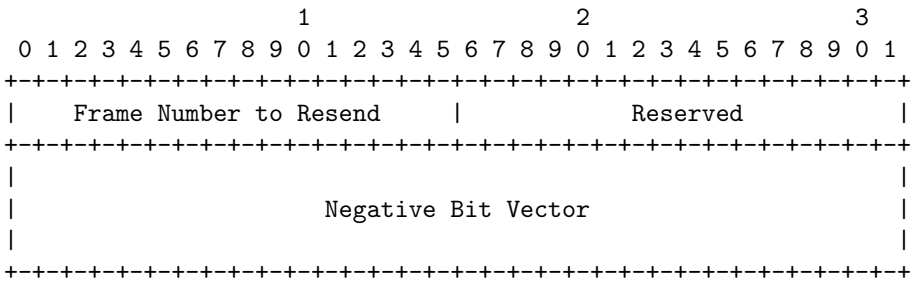
CC_Ack



Congestion control acknowledgements must be sent once per `TTAW`. Their role is to signal the other endpoint of possible congestion in the network. For more see the Congestion Control section.

CC_Ack caries three pieces of information: the sequence number of the most recent data carrying packet, the number of packets this endpoint has estimated as having been lost since the previous transmission of a CC_Ack, and the frame number of the oldest incomplete frame which this endpoint expects to be resent.

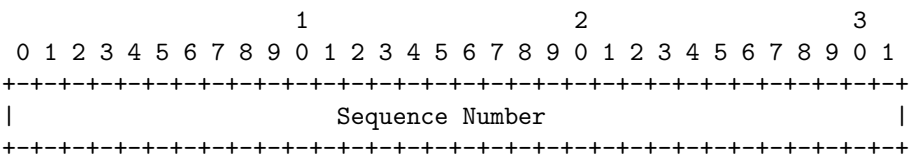
Vec_Ack



The negative vector acknowledgements are a way to tell the sender that the fragments of the frame in question have been transferred at a high enough reliability that the receiver would like the rest of the missing pieces.

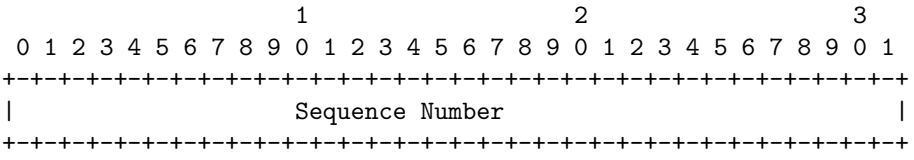
This packet effects the sending priority of the endpoint receiving this packet. If the current frame the Sender is sending is older or the same then it ignores the packet. If the current frame is newer, the Sender must stop sending it and begin resending the frame specified in the Vec_Ack packet by resending only those fragments that are specified by the Negative Bit Vector field in the Vec_Ack.

Sync_Ack



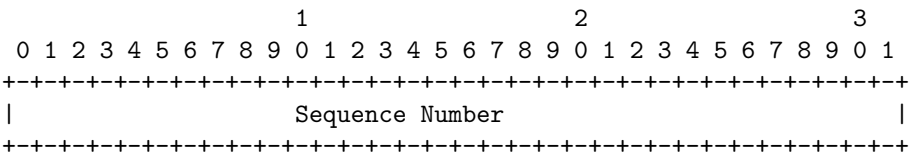
The Sync_Ack packet is sent as a response to the Sync packet and is used to recompute the `RTT` estimate on the receiver side. In the future it might be extended to buddle a set of other options that might need to change in the middle of a connection.

4.6 CloseReq Packet



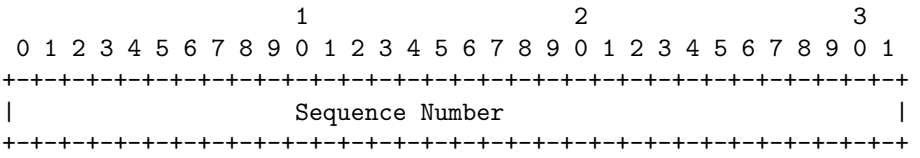
A close req packet is sent with only a sequence number as its payload and is a nice way to close a connection.

4.7 Reset Packet



This packet is sent with only a sequence number as its payload and is a last resort method to close a connection.

4.8 Sync Packet



Sync packets are meant to stabilize the congestion control. Paired with a Sync_Ack acknoledgement, `RTT` can be estimated.

5 Connection

Figure 1 is a summary of the first two tables in the paper, depicting a deterministic finite state machine where the states are Connection States from Table 1 (identified by their ID from the table), and the transitions (edges) are packet types from Table 2. All the transitions which are missing from the picture are taken to be invalid and are ignored, i.e. they are identity transitions, so for example a Request packet would be ignored if received in a Connected state (ID 5).

Note that the transitions for some of the states depend on factors other than packet types, timeouts might also trigger state transitions for example.

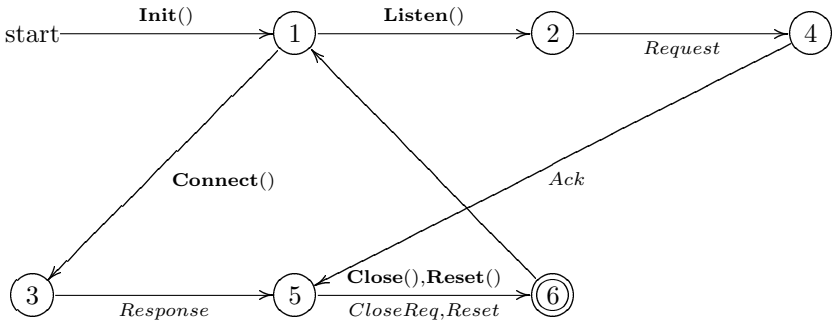


Figure 1: DFA representation of the protocol connection states

Note that the details of the source network address and port numbers for each packet have been left out below. During the handshake, these are cached and then validated for each incoming packet.

The Following sections will break up the connection into three separate parts- the Set Up, the Handshake, Data Transfer and Termination. Terms and their explanations relevant to each step may be found in the tables relegated to page 10.

5.1 Set Up

Initially both endpoints are in the Disconnected state. To transition into the next state, the protocols at both endpoints must have defined the connection variables summarized in Table **??**. Just as with TCP, an endpoint may choose to become a server or a client depending on whether it chooses to listen for a connection or send a Request packet to a listening connection.

5.2 Handshake

Client The client initiates the connetion by sending a Request packet. The packet must include the parameters in Table 4.2. The client then enters CConnecting state. If the server does not reply within a `CONNECTING_RESPONSE_TIMEOUT` seconds threshold, the connection packet is resent. After `CONNECT_ATTEMPTS_THRESHOLD`, the client will stop retrying and signal an error to the application.

Upon receiving a Request packet, a server must reply as soon as it can with a Response packet. This pair of packets (Request and Response) are used to compute the initial `RTT` (round trip time) estimate by the client.

Server After responding with a Response packet, the server enters the SConnecting state. If the Response packet is lost, the server will wait for the client to retry with the Request packet before sending another Response packet. After `CONNECT_TIMEOUT` second, the server will drop the connection and return to Listening state.

Once the client receives a Response packet from the server, it enters the Connected state. The next packet to come from the client, unless it is a Reset packet, must be an Ack packet and it will place the server into Connected state.

The bidirectional nature of the connection allows us to deal with a "half connection," and associate with it a Sender and a Receiver both of which may be either of the endhosts.

To disambiguate the attributes of a half connection as seen by an endpoint in Table **??**, we prepend "R_" for remote and "L_" for local. As an example, the local `FRAGMENT_SIZE` as seen by an endhost is denoted `L_FRAGMENT_SIZE` and the remote attribute becomes `R_FRAGMENT_SIZE`.

5.3 Data Transfer

Data Transfer for protocol X can be viewed at two different scales. At the fragment or packet level, we are sending fragments of a constant size, not caring about reliability, determined only to send as much as possible under the rate controls.

At the frame level, we are filling up variable sized frames with fragments (all of a constant size except perhaps the last one). After we see that the Sender has moved on to the next frame, we compute the percentage (in bytes) of the frame received (by deducing the number of missed fragments) and either keep the frame around or throw it away depending on the `RELIABILITY_COEF`.

If we keep the frame then we ask the Sender to resend the missing pieces of the frame using a bit vector to represent the missing fragments in the frame and do not let the sender move on to newer frames until this particular one has been completed.

The retransmission of missing fragments happens as before, only now we are determined to receive the full frame and if the frame is incomplete after all the fragments have been retransmitted, we repeat the process as necessary until we received the full frame. Once the frame is complete, we let the application have the frame and allow the Sender to send us new frames.

5.3.1 Receiver

Let the oldest frame the Receiver is interested in be Oldest_Frame, the newest frame which we have seen so far in our `FRAME_WINDOW` to be Newest_Frame, and let the sequence number of the last data packet received be Last_Seq_Number.

Every data carrying packet (a Data or a DataAck packet) has a sequence number, a frame number and a fragment number. Initial sequence and frame numbers are established during the handshake. Fragment numbers are only relevant when talking about a specific frame number. Fragment numbers start at 0 and increment with each data fragment, also each subsequent data packet increments the sequence number independently of the frame number- this lets us determine how many packets were sent or were lost or delayed between the reception of any two data carrying packets. These three attributes determine the actions of the Receiver.

Given a data packet D with attributes Seq_Number, Frame_Number, Frame_Len, Frag_Number, Data_Len, the Reeiver processes D as follows.

First we run the attributes through a set of common validation rules and ignore D if any of the following are true.

- Seq_Number $\leq$ Last_Seq_Number
- Frame_Len $\neq$ stored frame length for the frame Frame_Number
  (only if we have a record of this frame already)
- (Frame_Len $>$ `MAX_FRAME_SIZE`) $\vee$ (Frame_Len $<$ `FRAGMENT_SIZE`)
- (Frag_Number + 1) $\times$ `FRAGMENT_SIZE` $>$ Frame_Len
  (note that Frag_Number starts at 0)
- Data_Len $>$ `FRAGMENT_SIZE` $\vee$ Data_Len $= 0$

Note that by this point, the packets have passed the UDP data checksum, therefore a bad attribute is not due to data corruption.

Define the $R(X)$ to be the boolean result of the following inequality:
(Bytes Received for a Frame X) / (Total Number of Bytes in a Frame X) $\geq$ `RELIABILITY_COEF`

A New Frame Number (Oldest_Frame $\leq$ Frame_Number $\leq$ Newest_Frame)

Let $M = min($`FRAMES_WINDOW` $-$ (Newest_Frame - Oldest_Frame), (Frame_Number - Newest_Frame))

$M = 0$: D doesn't fit into our `FRAMES_WINDOW` and we therefore ignore D.

$M > 0$: D has a frame number in the future. This causes the Sender to allocate a buffer of size Frame_Length to accomodate the new frame specified by Frame_Number.

$(M > 1) \wedge (R(0) = true)$: Whole frames were lost or delayed. We send $M - 1$ negative vector acknowledgements filled with 1s to indicate that the Sender received no fragments for any of the $M - 1$ frames between Newest_Frame and Frame_Number. Each of them will carry a Frame_Number attributed of (Newest_Frame + k) for $k \in \{1, .., M - 1\}$.

The data portion of the packet is then placed within the newly allocated buffer for the frame Frame_Number at a location offset by Frag_Number * `L_FRAGMENT_SIZE`.

Note that the *min* in the definition of M tells us to discard the frames that are outside of our scope of frame vision controled by `FRAMES_WINDOW`.

Urgency

For each incomplete frame with frame_number $\in$ {Oldest_Frame, ...,Newest_Frame}

- Urgency[frame_number] is incremented by one. Initially Urgency[i] = 0 for all i. The value for each frame_number determines when to send another Ack Vector to the Sender urging the Sender to resend the missing fragments of the incomplete frame in which the Receiver is still interested.
- If Urgency[frame_number] $\geq$ `URGENCY_THRESHOLD` then a Vec_Ack packet is sent with the `ACK_VECTOR_OPT` option set to 1, and a compressed bitmap describing the state of the incomplete frame. Each of these packets will carry the attribute Frame_Number set to the frame_number.

The Vec_Acks tell the Sender that there are incomplete frames to be dealt with, and that they should be attended to first. The missing fragments details are sent so that the Sender will be able to resend them. The Sender must prioritize the sending of fragments based on the age of their frames: the oldest receiving the highest priority.

A Known Frame Number (Oldest_Frame $\leq$ Frame_Number $\leq$ Newest_Frame)

$\forall$ frame_number $\in$ {Oldest_Frame, ...,Frame_Number}

- Urgency[frame_number] increments by one and if Urgency[frame_number] $\geq$ `URGENCY_THRESHOLD` then we send a Vec_Ack back to the Sender urging the resending of the missing fragments for each incomplete frame older than the one we've received.

$\forall$ frame_number $\in$ {Newest_Frame , ..., Frame_Number}

- Urgency[frame_number] = 0

Notes

- Vec_Ack for an incomplete frame will be repeated until a fragment destined for this or older frame is received.
- An `URGENCY_THRESHOLD` of 1 will squeeze the frame window to as little as two frames, as an incomplete frame will request its missing fragments immediately after a fragment for a new frame has been received.
- Upon receiving the last fragment for a frame. The Receiver delivers the frame to the application.
- To make sure that the frame is removed from the Sender's buffer space, the frame needs to be implicitly acknolwdged by a CC_Ack which will specify a frame number attribute that is newer than the frame that has been completed, thus allowing the Sender to free its buffer space of this frame.

5.3.2 Sender

Note that the Sender's set of variables is independent from its other half connection in which this Sender is a Receiver.

Each frame sent by a Sender is associated with an identifier that is initialized and exchanged during the handshake. The initial value of this identifier which from now on we call the frame number is initially generated by each endpoint and exchanged during the handshake. The recommended procedure to generate the initial frame number is use a randomizer modulo $2^{\texttt{FRAME\_ID\_LEN}}$, as the purpose of this identifier is two fold: to map packets to frames and to prevent spoofing of the connection. The latter is effective because the Receiver is aware of only a few frames at a time and drops packets for which the frame number deviates too much from working set of values. Every successive frame increments the working frame number by one.

Let the newest frame that Sender is sending have the frame number Last_Frame.

The sender maintains two queues for allocated frames. The first is the Current_Queue, the second is the UnAcked_Queue.

The Current_Queue is a priority queue, prioritized by the age of the frames, the older the frame, the higher its priority (Note the problem with wrap around of Frame Numbers!) This queue contains frames who still need some or all of their fragments sent. Each frame in this queue has an associated fragment bitmap to let the Sending mechanism know exactly which fragments need to be resent. After being resent, the frame moves to the UnAcked_Queue.

The UnAcked_Queue contains frames that have been already sent once, but might need some or all of their fragments resent in the future. Thus, depending on a Receiver's Ack pattern, a frame from this queue may move to the Current_Queue.

A New Frame Before allowing a new frame to enter the queueing system, the Sender discards the frame if it has an illegal frame length.

Once the frame has been validified, it receives a frame number Last_Frame + 1, and it is fragmented into `FRAGMENT_SIZE` sized fragments, except perhaps the last one. Each one of them is enumerated starting from 0. A vector map filled with 0s is associated with the new frame. The frame is then enqueued into the Current_Queue.

Sending a Frame

Taken from the top of the Current_Queue

- Sent in order based on the bitmap associated with the frame
- Moved from the Current_Queue to the UnAcked_Queue

Asynchronous events

Receiving a Vec_Ack

- If frame is in the Current_Queue, update its vector (but have to somehow check that the new Vec_Ack has fresh information)
- If frame is in the UnAcked_Queue, move it to the Current_Queue

Receiving a CC_Ack

- Deallocate frames from Current_Queue and UnAcked_Queue that are older than the older working frame number on the CC_Ack
- Check for data loss (CC job ...)

5.4 Termination

Termination can be initiated in two ways by either endpoint. The first way is to send a CloseReq packet. This packet will transition the Receiving endpoint into Closing state and will alert the application of the new state. An Ack with a `CLOSING_OPT` option set to one is the only valid acknowldgement of the CloseReq packet. After `CLOSING_TIMEOUT` seconds the Sender resets the connection and all packets drops all packets coming from the other endpoint.

A second way to initialize termination is to send a Reset packet. Packets of this type are not acknowledgeable. A Reset packet initiates immediate termination of the connection by both the Receiver of the packet and the Sender.

6 Sync

Sync packets are meant to estimate `RTT` (round trip time). Upon receiving a Sync packet, an endpoint is required to respond with a SyncAck packet as soon as possible. `RTT` is used to determine the `WINDOW` size, these are specific to each endpoint.

7 Rate Controls

Rate controls ensure that the Sender does not overrun the Receiver with its sending rate, and that the protocol is TCP-friendly towards other connections that are coexisting in the network at the time.

Congestion and flow controls control only the rate of Data or DataAck packets, Acks and other packet types are not counted towards the rate. The reason for this is because the overhead is not large enough to make for a substantial contribution- Acks are sent only every `TTAW` seconds, and are only a small fraction of the full MTU of the connection, even on relatively slow links. Therefore their impact on the friendliness of the connection is minimal. Likewise, Acks are processed on the Receiver's side without being buffered, thus there is no danger of overrunning any buffer.

Note: that the initial implementation is a user level implementation and therefore does not take into account Kernel space buffers which may indeed be overrun but over which the implementation has no control.

7.1 Congestion Control

The CC_Ack acknowledgment packets determine most of the congestion control mechanism.

A CC_Ack packet (or a DataAck packet) must be sent by both endpoints at least once every `TTAW` (Time to Ack Window) seconds. It provides information to the Congestion Control mechanism so that it may determine whether the current `CW` (congestion window) is too large and should be shrunk or if the network may allow for a higher load. This packet type includes three pieces of information: seq id of the first packet received since the last packet acknowledged, the number of packets received since the last acknowledgment, the seq id of the last packet received.

Both sender and receiver will keep windows (number of packets in flight).

If an endpoint misses an Ack within a `TTAW` timeperiod, an endpoint halves its sending rate by halving the `WINDOW`. Note that the rate cannot go below 1 packet.

If an endpoint receives the Ack within a `TTAW` timeperiod, this indicates that the data link may support more throughput. The endpoint thus increments `WINDOW` by one.

7.2 Flow Control

Flow Control works alongside Congestion Control to limit the rate of the sender. Flow Control is established by the Receiver and depends on the available buffer size. The two parameters that are prerequisite for this are Sender Buffer Size - `SBS`, and Receiver Buffer Size - `RBS`. Since each endpoint maintains a local and a remote copy of both, they will be denoted as `L_SBS`, `L_RBS` and `R_SBS`, `R_RBS` respectively.

The buffer sizes determine how many packets each endpoint may hold. They are exchanged during the handshake as `MAX_FRAME_SIZE` is known to each endpoint before a connection is established. The client's `RBS` and `SBS` are send with the Request packet, and the server's `RBS` and `SBS` are part of the Response packet.

8 Security

Name	Units	Description
MAX_FRAME_SIZE	bytes	Maximum fragment size for this connection
FRAGMENT_SIZE	bytes	Fragment size for connection
SEND_BUFFER_SIZE	bytes	Sending buffer size
RECEIVE_BUFFER_SIZE	bytes	Receiving buffer size
RELIABILITY_COEF	float $\in$ [0,1]	Controls the discarding of frames after an initial tra
CONNECTING_RESPONSE_TIMEOUT	sec	Timeout for a Response packet
CONNECT_ATTEMPTS_THRESHOLD	int	Number of attempts to connect to a server
FRAMES_WINDOW	int	Maximum number of frames to maintain simultaneo
URGENCY_THRESHOLD	int	Number of packet to receive for a newer frame before sending a Vec_A
FRAME_ID_LEN	bits	Length of the frame number

Table 3: Connection Parameters

Parameter	Units	Description
INIT_SEQ	int	Initial sequence number for outbound (data) packets(integer)
INIT_FRAME	int	Initial frame number for outbound frames(integer)
FRAGMENT_SIZE	bytes	Fragment size for local bound packets (in bytes)
MAX_FRAME_SIZE	bytes	Maximum frame size for local bound packets (in bytes)
BUFFER_SIZE	bytes	Buffer size of the local buffer (in frames)

Table 4: Request Packet Parameters