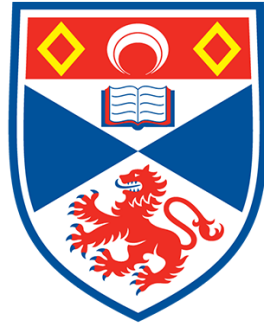


# CS3102 P2: Practical Report

Reliable Data Transfer Using UDP



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# 1 Introduction

This report cover the design and implementation of a connetion-orientated, reliable, uni-cast, transport protocol,built on top of UDP.

The protocol in question is called RDT - Reliable Data Transport

## 2 Design

Given the scope of the practical, simplicity was the main goal when considering the design of the RDT protocol. This section will discuss the design decisions and rationale behind them. The two attempted extension features, checksums and adaptive re-transmission timeouts, are also detailed here.

### 2.1 Packet Structure

RDT packets (see Figure 1) are composed of a constant 12 byte header and an optional data segment. The size of the data segment ranges from 0 to 1300 bytes, with 1300 bytes used as the maximum size so as not to interfere with the operation of `slurpe-3`, which was used for testing. **Theoretical maximum size?**

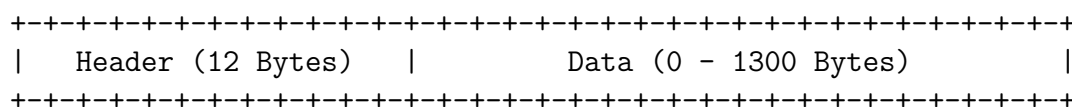


Figure 1: RDT Packet Structure

The RDT header (see Figure 2) is comprised of the following fields: a 32-bit **sequence** field, used for **what?**; a 16-bit **type** field, used to denote the packet function (see Figure); a 16-bit **checksum** field, calculated over the header and data segment to **what?**; a 16-bit **size** field denoting the size of the data segment (in bytes); and a 16-bit **padding** field to ensure 32-bit word alignment.

Several factors influenced the RDT header design. As the **C library function** `ftell` (used to calculating file sizes in `RdtClient.c`) returns 32-bit **long** values, a 32-bit **sequence** field was required to support the transmission of large files/amounts of data. The given implementation of the IPv4 header checksum used returns a `uint16_t` value, thus necessitating a 16-bit field. As a maximum data segment size of 1300 bytes was required, at least 11 bits were required for the **size** field, however 16 bits were used for alignment. For the remaining **type** and **padding** fields, there were no other considerations for field size other than 32-bit alignment.

A single type field was chosen, rather than a set TCP-style flags, for simplicity. Given the minimal nature of the RDT protocol, it was faster simpler to enumerate all packet types (see Figure), rather than testing multiple flags.

The type field supports the following types: **SYN** (0) and **SYN ACK** (1), used for the connection handshake; **DATA** (2) and **ACK** (3), used for sending and acknowledging data

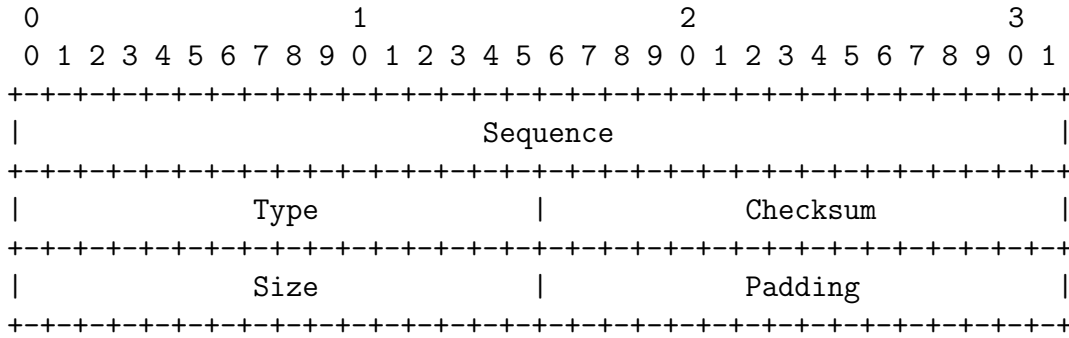


Figure 2: RDT Header

segments; **FIN** (4) and **FIN ACK** (5), used for graceful connection termination; and **RST** (6), used for abrupt connection termination.

## 2.2 Connection Management

The operation of the RDT protocol can be modelled by the FSM in Figure 3. For connection management, a two-way handshake is used. As RDT only supports uni-directional communication, a two-way handshake (see Figure ) is adequate for establishing and terminating connections. Adaptive re-transmission timeouts are used in both the handshakes and transmission of data segments.

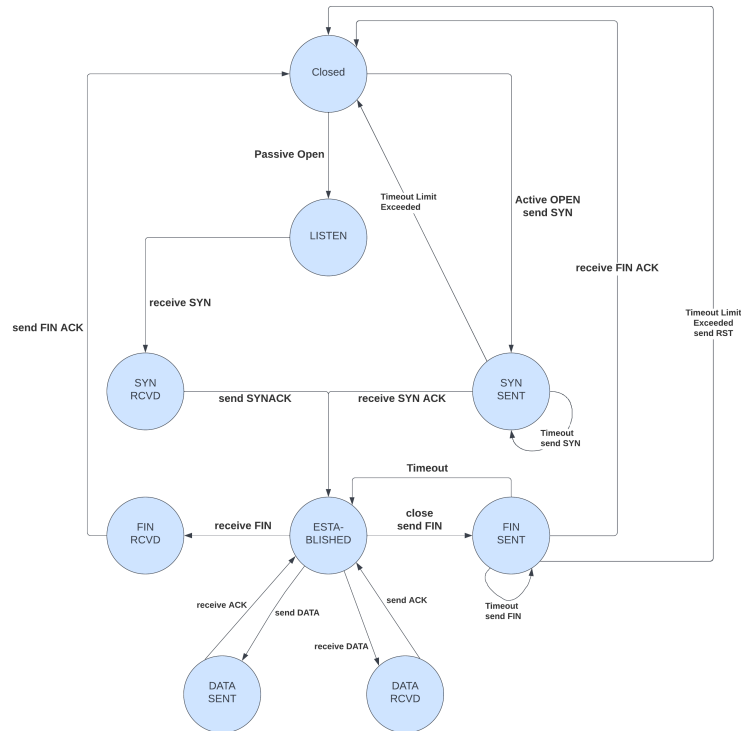


Figure 3: RDT Finite State Machine (see also A.1)

An initial RTO value of 200ms is used for handshakes, and this doubles until 5 attempts have been made before the connection is abruptly terminated with an RST.

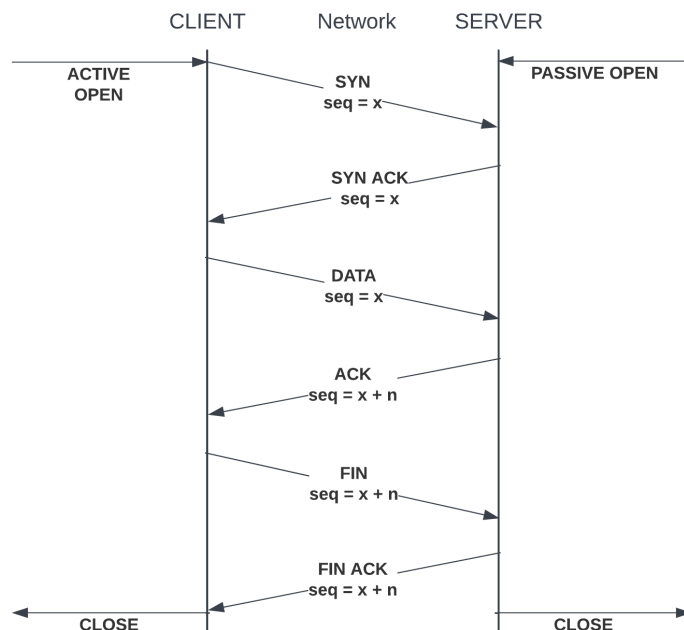


Figure 4: Connection Timeline Diagram (see also A.2)

## 2.3 Adaptive RTO

In an extension to the base specification, adaptive re-transmission timeouts using measured RTT has been implemented. RDT's adaptive RDT is modelled on TCP's Retransmission Timer [3]<sup>1</sup>. RDT uses the same initial RTO of 1s and maximum of 60s.

## 2.4 Checksum

The RDT `checksum` field is calculated over the entire segment/packet (i.e. header and data) using the IPv4 Header Checksum Algorithm [1]. The implementation of the original source [2] to support the use of 8-bit byte values in the data segment. During checksum calculating, the `checksum` field itself is set to 0 for consistency.

# 3 Testing

To test and verify the correct operation of RDT, two test programs `RdtClient.c` and `RdtServer.c` were created to send an arbitrary file between two lab PCs using RDT.

<sup>1</sup>Clock granularity is not considered, however RTO values are calculated in microseconds and the School Lab PCs have a granularity of 1 nanosecond.

### 3.1 Methodology

To test that RDT packets were delivered reliably and in-order, a sufficiently large JPEG file (*dog.jpg*) was used as test file. This provided visible feedback as to the integrity of the data received by **RdtServer**. This integrity was also verified by calculating SHA1 checksums of both the sent and received files. **slurpe-3** was used to provide an emulated path with loss, delay and restricted data rate to test the reliability of RDT in degraded network conditions.

### 3.2 Results

From the results in Table 1 and Appendix A.3 we can see the RDT performed as expected in a variety of degraded network conditions.

Scenario	In file	Out File	Match	Time
Control	<i>dog.jpg</i>	<i>dog-control.jpg</i>	✓	0.002336s
Delay	<i>dog.jpg</i>	<i>dog-d.jpg</i>	✓	12.037570s
Loss	<i>dog.jpg</i>	<i>dog-l.jpg</i>	✓	57.010318s
Constrained Rate	<i>dog.jpg</i>	<i>dog-cr.jpg</i>	✓	0.930083s
Loss and Delay	<i>dog.jpg</i>	<i>dog-ld.jpg</i>	✓	92.969945
Loss and Constrained Rate	<i>dog.jpg</i>	<i>dog-lcr.jpg</i>	✓	64.960451s
Delay and Constrained Rate	<i>dog.jpg</i>	<i>dog-dcr.jpg</i>	✓	13.948315s
Delay, Loss and Constrained Rate	<i>dog.jpg</i>	<i>dog-ldcr.jpg</i>	✓	64.960451s

Table 1: Results for varying network conditions

## 4 Analysis

### 4.1 Performance in Different Network Scenarios

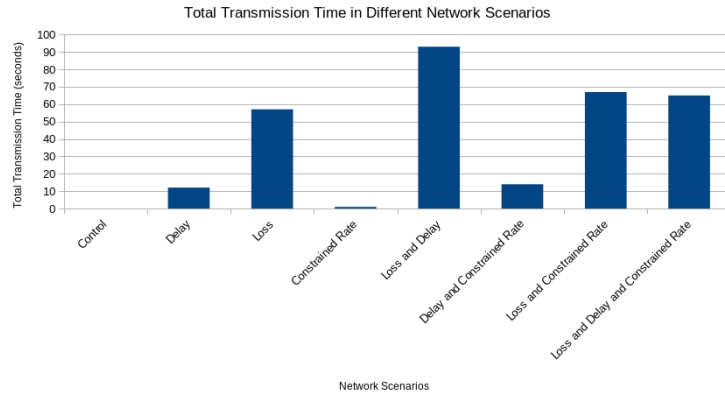


Figure 5: Total Transmission Time in Different Network Scenarios for 230 Kb file.

- Bandwidth utilization

## 4.2 RDT Packet Data Size

## 4.3 Wastage due to Control Information

$$\rho = \frac{i - c}{i + a} = \frac{1312 - 12}{1312 + 12} = 0.982 \text{ (3 s.f.)}$$

# 5 Evaluation

## 5.1 Extension Features

Given that RDT is implemented using UDP and uses the same checksum algorithm used by UDP, it is unlikely that RDT will ever encounter an incorrect checksum. This makes the use of a checksum in RDT mostly redundant. However the inclusion of a checksum provided some utility. Firstly, it helped to catch several errors in the initial implementation of RDT, and secondly it provided a useful opportunity to understand the function of the Internet Checksum.

The inclusion of adaptive RTO was more useful however. G

## 5.2 Further Extension

While the implementation of bi-directional communication and Continuous-RQ was not implemented, it is useful to consider these features in evaluating the desing of RDT.

In it's current design, RDT would be unable to support simulatenous bi-directional communication, due to its use of a two-way handshake. For bi-directional communication to occur, both parties are required to choose and synchronise an 'Initial Sequence Number', which is not possible with only a two-way handshake. Therefore, to support bi-directional communication RDT would require a significant re-design.

However, RDT would not require a fundamental re-design to support Continous-RQ. Continous-RQ with Go-Back-N would solve RDT's fundamental issue of low transmission rate due to link under-utilisation (see **section**).

# 6 Conclusion

As demonstrated in this report RDT provides reliable, ordered, uni-directional data transfer on top of UDP. RDT provides reliable performance in heavily degraded network environments. However in its current implementation RDT is extrememly inefficient due to its Idle-RQ transmission mechanicsm. RDT performance suffers particularly when there is heavy packet loss that causes retransmission. This efficiency could have been addressed, if given more time, by replacing Idle-RQ with Continuous-RQ with Go-Back-N.

Overall this practical provided an interesting opportunity to design and implement a protocol and analyse protocol performance.

## References

- [1] Internet Protocol. RFC 791, September 1981.
- [2] Saleem Bhatti. Simple IPv4 Checksum Calculation Example, January 2022.
- [3] Matt Sargent, Jerry Chu, Dr. Vern Paxson, and Mark Allman. Computing TCP's Retransmission Timer. RFC 6298, June 2011.



# A Appendix

## A.1 Finite State Machine

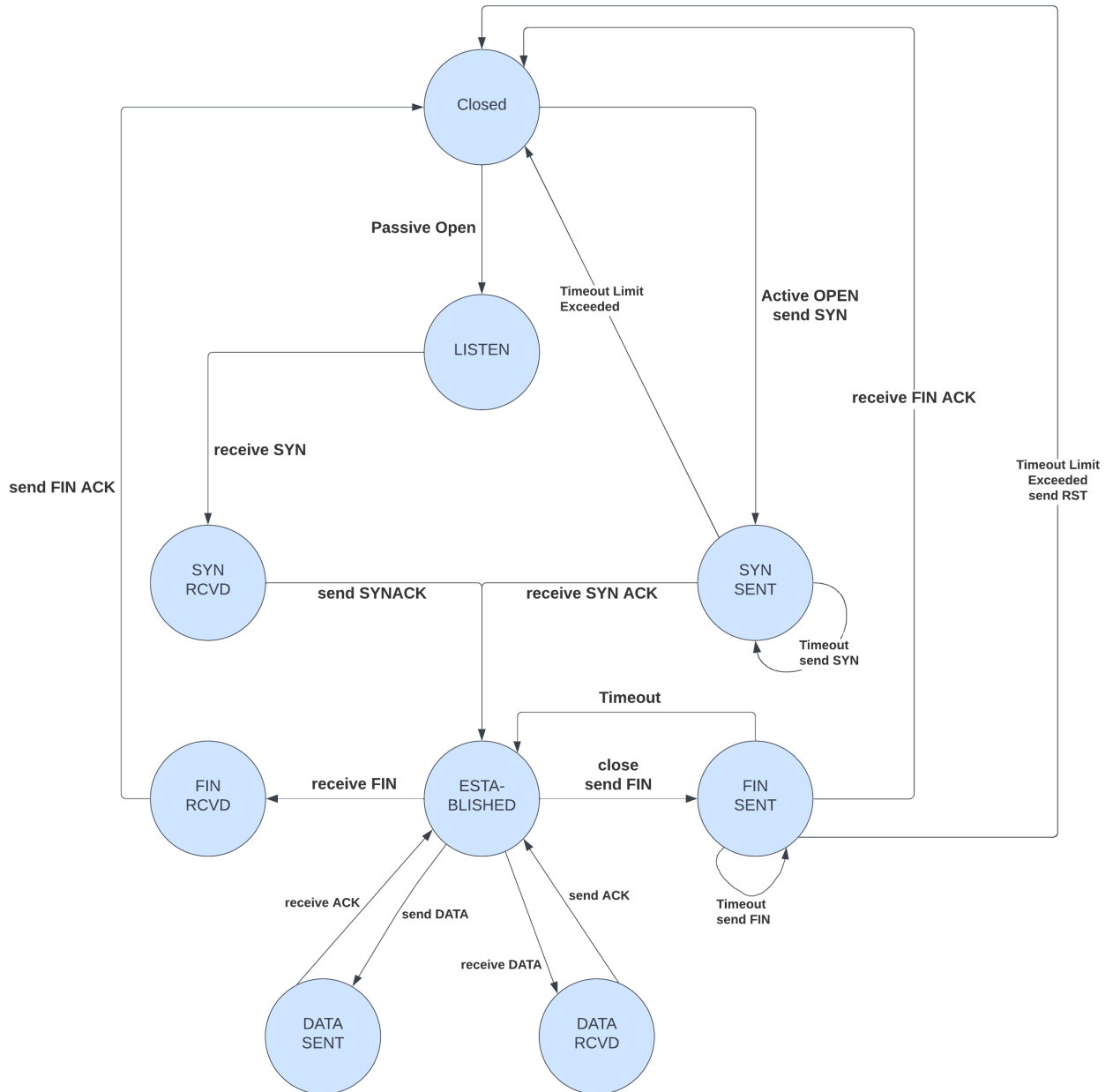


Figure 3: RDT Finite State Machine

## A.2 Connection Management Timeline Diagram

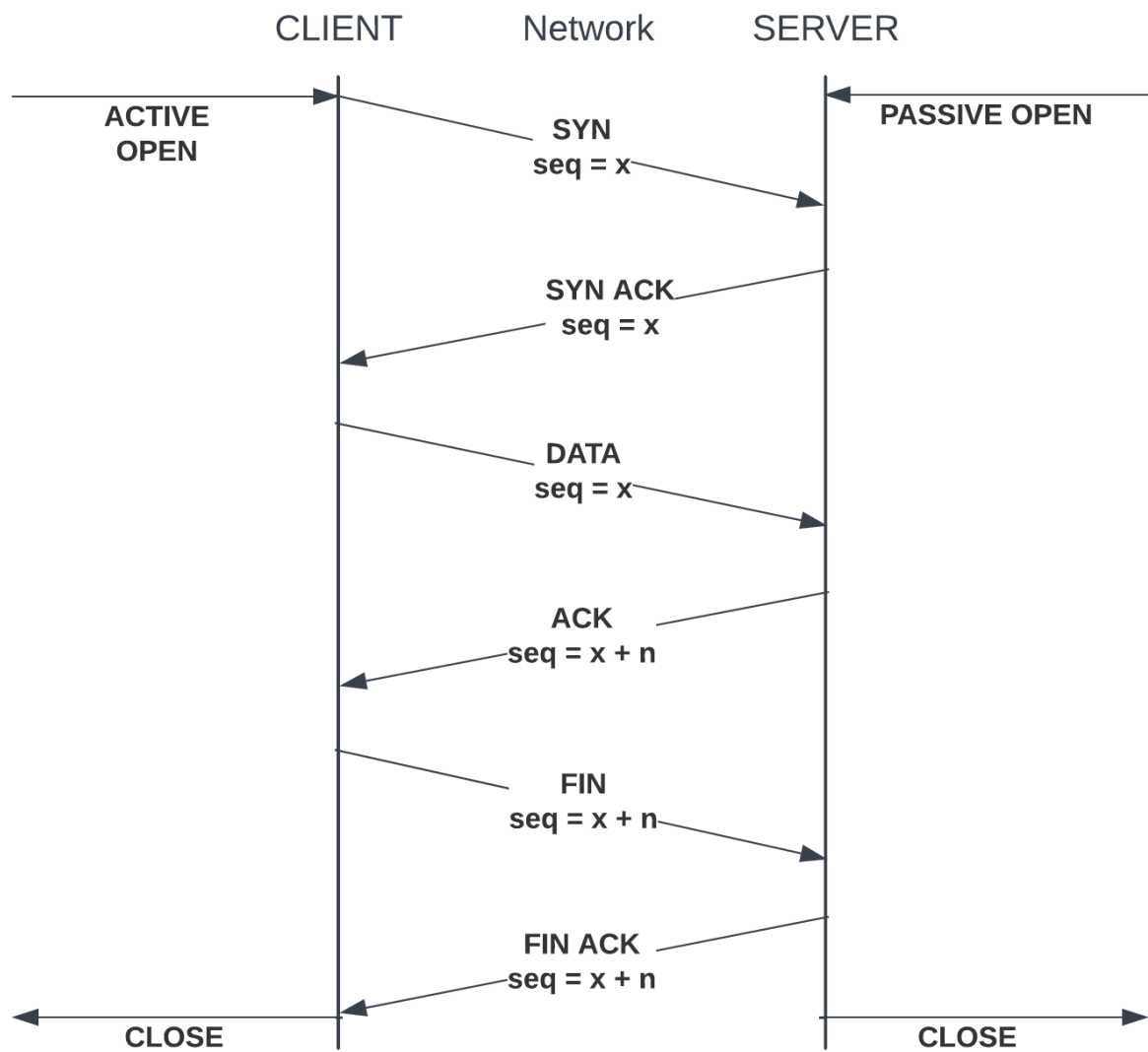


Figure 2: RDT Finite State Machine

### A.3 Testing Screenshots

Testing screenshots can be found inside the `data/screenshots` directory.