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Performance of Server Message Block implementations over QUIC

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Abstract

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Preface

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Contents

Abbreviations

ACK	acknowledgment
HOL	Head-of-line
IP	Internet Protocol
ISN	Initial Sequence Number
NAT	Network Address Translator
OS	Operating System
RFC	Request For Comment
RTT	Round-Tripe Time
SMB	Server Message Block
TCP	Transmission Control Protocol
TLS	Transport Layer Security
UDP	User Datagram Protocol

1 Introduction

1.1 Research questions and objectives

1.2 Thesis structure

2 Background

This section of the thesis will give an overview of the two most common transport protocols, the Transmission Control Protocol (TCP) and the User Datagram Protocol (UDP). This section also covers the Server Message Block (SMB) protocol, outlining the file-sharing protocol.

2.1 Internet transport protocols

2.1.1 Transmission Control Protocol

The TCP, as outlined in Request For Comment (RFC) 793 is a foundational internet transport protocol. It was originally published in September 1981, focusing primarily on solving military communication challenges. It is intended to be a highly reliable transport protocol between hosts in a packet switched network. The TCP is connection-oriented, providing reliable, end-to-end, bi-directional communication between a pair of processes, in the form of a continuous stream of bytes. The TCP protocol is designed to fit into a layered hierarchy of protocols, slotting in on top of the internet protocol (IP)[1]. IP handles the addressing and routing of datagrams between the hosts, while TCP aims to ensure that information is delivered correctly, in order, and without duplications, without any reliability guarantees needed from the underlying protocol, which may lose, fragment or reorder the datagrams[2].

TCP ensures reliable communication by using a system of sequence numbers and acknowledgments (ACKs). Each transmitted byte of data is assigned a sequence number, and the peer is required to send an ACK to acknowledge that the data was received. On the receiver side the sequence numbers are used to reconstruct the data, ensuring that the data is received in order. If the sender does not received an ACK within a timeout period, the missing segment will be retransmitted. A checksum is included with each segment, ensuring that the datagram was not corrupted during transport. If data corruption is detected, the receiver will discard the damaged segment, and rely on the retransmission mechanic to recover. The TCP uses a receive window for flow control, allowing the receiver to decide the amount of data that the sender may send before waiting for further ACKs. The reliability and flow control aspects of the TCP demands that the TCP store some information about the transmission. The data stored about the data stream, sockets, sequence numbers and windows sizes, is referred to as a connection. The network address and port tuple is referred to as a socket, and a pair of sockets is used in identifying the connection. Using this mechanic to uniquely identify connections, allows for multiple processes to simultaneously communicate using the TCP[2].

The TCP header, figure 1, encodes the functionality of the TCP. It follows the IP header in a datagram. The header is usually 20 bytes long, but can be extended using options. It begins with the source and destination port, which together with the source and destination addresses, are used to identify the connection. The next two fields in the header are the sequence and acknowledgement numbers. The sequence number is the sequence number of the first data byte in the data segment. If the SYN

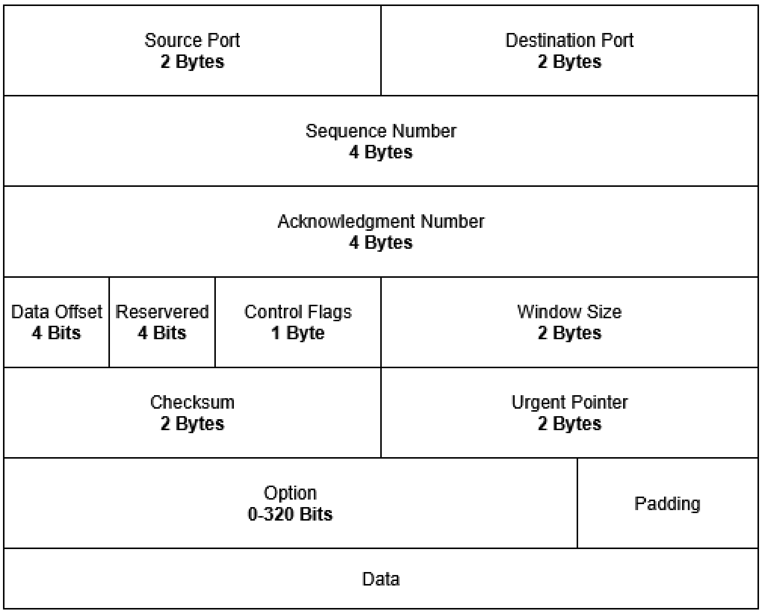

Figure 1: The TCP header

flag is set the sequence number is referring to the initial sequence number (ISN). The acknowledgement refers to the next sequence number the receiver is expecting to receive, at the same time acknowledging that all sequence numbers up to this point was received. Next is the data offset field, indicating where the data begins. The reserved field following this must be 0. After this is the 1 byte flags field

- **URG** Urgent pointer field is set
- **ACK** acknowledgement field is set
- **PSH** Push function, requesting that buffered data is sent immediately to the receiver
- **RST** Reset the connection
- **SYN** Synchronize sequence numbers
- **FIN** Sender is done sending data

The 2 byte window field specifies the number of bytes that may be in-flight at any one time. This is the specified size of the sliding window that is used for flow control purposes. Following the window field is a 2 byte checksum field, used for detecting corruption of the TCP-header, data payload as well as a pseudo IP header, containing information about the source and destination addresses, as well as the protocol number and tcp packet length. In case the URG bit is set in the flags field, the 2 byte urgent pointer header field indicates where the urgent data ends. Finally the options field contains extension to the normal TCP header, containing among other, options for maximum segment size and multipath TCP[3].

To ensure reliable delivery of TCP segments, each segment is assigned a sequence number. This allows the receiver to reconstruct segments delivered out of order, and additionally detect missing segments. The receiver sends acknowledgments, containing the next expected sequence number, to signal to the sender that the data was successfully received. The sequence number is a 32 bit number, with the initial sequence number (ISN) selected randomly at the time when the TCP connection is established. This ensures that sequence numbers from stale connections have a low probability of overlapping with any active connection[2].

The TCP connection is established via a three-way handshake. The client sends a TCP packet with the SYN bit set in the flags field, this packet contains the clients ISN. The server responds with a packet with the SYN and ACK bit set, acknowledging the clients sequence number as well as providing the servers ISN. Finally the client responds with an ACK, acknowledging the servers ISN. Following this the client and server are synchronized, and communication may begin. A peer may terminate its side of the connection by sending a FIN packet, signaling to the other endpoint that one side has closed its side of the communication. The closed endpoint may continue receiving data until the other endpoint also closes it side[2].

2.1.2 User Datagram Protocol

The UDP, which was defined by RFC 768, is designed to enable programs to transmit self-contained messages, know as datagrams, over a packet-switched network. The UDP is designed to run on top of the IP[1], using IP addresses and port numbers for addressing. The UDP is by design connectionless, providing no guarantees for datagram delivery, duplicate datagrams or in-order delivery. In exchange, the UDP aims to minimize the overhead present in the protocol. As UDP is connectionless there is no need to establish a connection via a handshake, instead datagrams can be transmitted directly, and they should be designed in such a way that they can stand on their own. The UDP header, as seen in figure 2 is only 8 bytes long, consisting of the source and destination port, as well as the length of the datagram and a checksum to verify the received datagram[4]. Even though UDP has a checksum field its use varies depending on the implementation. Some implementations may discard the datagram, or alternatively pass it along to the application with a warning, as UDP provides no way to recover from broken datagrams[**computer_networking**]. The

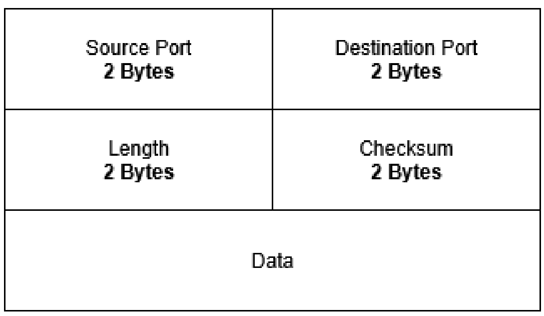

Figure 2: The UDP header

minimal UDP header (8 bytes) combined with the lack of a handshake makes UDP a protocol with the bare minimum needed for datagram transfer. Generally, UDP is used for applications where low latency is a requirement and some amount of packet loss is deemed acceptable. It is then up to the application layer to handle missing, reordered or duplicate datagrams.

2.2 The Server Message Block protocol

3 QUIC

The Internet's underlying infrastructure is in a state of constant evolution, driven by a demand for decreased latency, increased throughput requirements and a need for improved security. For many years now, the TCP has been the de-facto solution for reliable and secure, when combined with Transport Layer Security (TLS), communications. However, TCP was developed in a time where security and latency where not considerations, at least not in the same way as in todays landscape. Over the years there have been efforts to enhance TCP, such as multipath TCP[3] and combining TCP and TLS in HTTPS[5] to improve security. This section of the thesis will cover QUIC, a transport protocol developed to overcome the limitations and improve the performance as compared to the TCP[6].

The importance of the QUIC protocol is not to be underestimated. It represents a substantial change to the internet's transport layer, the first one in over two decades. QUIC was initially developed by Google, and then later standardized in RFC 9000[7]. QUIC is designed to address the issues experienced by TCP, with a focus on optimizing for web traffic. The main issues being the Head-of-line (HOL) blocking, but also aiming to improve on other aspects, such as integrating the TLS handshake into the transport handshake. A decision that was made for QUIC specifically was to move the protocol out of the kernel space and into the user space, allowing for rapid development and innovation[6].

This chapter of the thesis will outline the limitations of TCP that led to the development of QUIC, give an overview of the architecture and logic that drives QUIC.

3.1 The motivation for a new transport protocol

The TCP is a cornerstone of modern internet infrastructure. It has been a reliable work horse for more than 40 years, ensuring communications between users and hosts since its inception. However, as the design of TCP is largely colored by the landscape when it was created, much of the improvements that have been made to TCP, such as security, has had to be built on top of TCP, as these were not considerations at the time of the TCP's inception. Todays internet landscape, with real-time content, hyper mobile users and increased demands on latency, but also privacy and security, have exposed some of the limitation imposed by the TCP stack. This section will outlined the key issues that prompted the development of a new, modern protocol: QUIC.

3.1.1 Head-of-Line Blocking

The TCP guarantees that all frames will be delivered, reliably in-order. Seeing as the TCP works as a single byte-stream the creates a phenomenon know as Head-of-Line (HOL) blocking, where any lost packet will block the delivery of all subsequent packets until the missing packet has been retransmitted. This can potentially amplify issues in the network, increasing delays, decreasing throughput and worsening the user experience. To combat this limitation, modern network protocols, such as HTTP/2[8], have introduced measures to combat the issue. HTTP/2 introduced multiplexing of multiple requests over one connection, allowing multiple application-level streams, for example for different resources such as images or javascript, to be multiplexed over a single TCP connection. This means that HTTP/2 managed to mitigate application-level HOL blocking, which was an issues in earlier versions of HTTP, it still potentially suffered from transport level HOL blocking, as the multiplexed streams was still sent over a single TCP connection. As a result a single lost packet in the TCP stream still caused all other unrelated streams over the same connection to stall, even if their packets were successfully delivered, until the offending packet could be retransmitted. This in practice means that much of the improvements made by HTTP/2 in this regard was negated by the issues of TCP, particularly in lossy environments[9].

3.1.2 Handshake Delay

A limitation of the TCP stack is the delay caused by the TCP handshake. As discussed in earlier chapters, establishing a TCP connection is done via a three-way handshake (SYN, SYN-ACK, ACK). This handshake incurs one Round-Trip Time (RTT) of delay. In addition, most application use TLS for security, and historically the TLS 1.2 handshake and setup adds two additional RTTs of delay. While network bandwidth is ever increasing, much of the communication done on the internet consist of short dialogues, that are significantly impacted by the additional overhead brought on by the TCP plus TLS handshake[6].

Some of the latency brought on by the TLS handshake is addressed by TLS 1.3, adding support for 1-RTT and 0-RTT handshakes, at the cost of perfect forward secrecy[10]. Even with these enhancements, the TCP plus TLS handshake takes a minimum of 1,5 RTTs, due to the separation of connection and security handshake.

3.1.3 Protocol Ossification

A big hurdle in deploying new protocols and extensions to existing ones on the internet, is the protocol ossification of existing protocols on the internet. There exists a heap of middleboxes, such as Network Address Translators (NATs) and firewalls that are part of the network. These devices may be overly conservative, dropping or modifying packets that do not conform to their assumption. This is already an issue for TCP enhancements, and entirely new protocols have no chance of reaching their destination, without explicitly adding support in all necessary middleboxes. To get around this, protocol designer have to design their protocols from the ground up to be middlebox proof, such as QUIC encapsulating its protocol inside UDP as an anti-ossification measure[11].

A related issue of rolling out enhancements to existing protocols is that the network stack tends to be part of the Operating System (OS) kernel. The networking stack is tightly coupled to the OS, requiring OS updates or upgrades to implement changes to existing protocols. With todays upgrade frequency it can take years to roll out simple changes to the networking stack. QUIC moves the deployment of the protocol to the user space, improving the speed of development and deployment, and opening up the space for multiple actors to create their own implementations of the protocol[6].

4 The Server Message Block protocol

4.1 Information about the SMB protocol

5 Implementing QUIC as transport for SMB server

5.1 MsQuic architecture and API

5.2 Fusion SMB server QUIC transport layer design

6 Performance and interoperability benchmarking

6.1 Test environment

6.1.1 Hardware environment

6.1.2 SMB over QUIC implementations analyzed

Windows SMB client/server

Fusion SMB server

6.2 Test scenarios

6.2.1 interoperability tests

6.2.2 Becnhmarking workloads

6.3 Results

7 Conclusions

7.1 Discussion

7.2 Future work

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