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Performance and interoperability of Server Message Block implementations over QUIC

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Abstract

The Server Message Block (SMB) protocol, a widely adopted protocol for network file sharing, traditionally relies upon TCP, which faces limitations in modern networking environments. The QUIC protocol offers a modern alternative, allowing for a secure, reliable transport with native multiplexing capabilities. The SMB protocol has added support for QUIC, but performance analysis of its impact and of protocol implementations outside of Windows is limited.

In this thesis a QUIC transport layer for the Fusion SMB server was designed, implemented and evaluated in the Linux user space. A prototype was developed by integrating Microsoft's MsQuic library for QUIC capabilities. The thesis discusses the design of the transport layer and the architectural decisions made to interface with the existing server. The interoperability and performance of the implementation were benchmarked against Microsoft's implementation of SMB over QUIC in several different benchmarking scenarios. These results were compared to the performance of the standard encrypted TCP transport. The thesis concludes by discussing the performance results obtained, along with providing insights into the potential of leveraging advanced QUIC features in the SMB protocol, gained during the development process.

Keywords QUIC, Server Message Block, MsQuic, Linux

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Sammandrag

Server Message Block-protokollet (SMB), ett väletablerat protokoll för delning av filer över ett nätverk. Traditionellt förlitar sig SMB-protokollet på TCP, som medför begränsningar i moderna nätverksmiljöer. QUIC-protokollet erbjuder ett modernt alternativ som möjliggör säkert och tillförlitlig transport av information med inbyggt stöd för multiplexering. SMB-protokollet har utvidgats för att stöda QUIC, men prestandautvärderingar och implementeringar utanför Windows är begränsade.

I detta examensarbete har ett QUIC-transportlager för Fusion SMB servern designats, implementerats och utvärderas i Linux användarrymd. En prototyp har utvecklas genom att integrera Microsofts MsQuic bibliotek, som erbjuder stöd för QUIC funktionalitet. Examensarbetet diskuterar designen av transportlagret och de beslut som tagits ur ett arkitekturperspektiv, för att samverka med den existerande servern. Implementationens prestanda och interoperabilitet jämfördes mot Microsofts implementation av SMB över QUIC, genom ett flertal test scenarion. Resultaten jämfördes med referensvärden från standardtransporten med krypterad TCP. Examensarbetet avslutas med en diskussion angående resultaten, samt ger insikter till potentialen att bättre utnyttja de avancerade QUIC funktionerna i SMB-protokollet, med hjälp av insikter som erhållits under utvecklingsprocessen.

Nyckelord QUIC, Server Message Block, MsQuic, Linux

Preface

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David C. Enberg

Contents

Abbreviations

ACK Acknowledgment
ALPN Application Layer Protocol Negotiation
DFS Distributed File System
DTLS Datagram Transport Layer Security
HOL Head-of-line
HTTP Hypertext Transfer Protocol
IP Internet Protocol
ISN Initial Sequence Number
NAT Network Address Translator
NIC Network Interface Controller
OS Operating System
RDMA Remote Direct Memory Access
RFC Request for Comments
RTT Round-Tripe Time
SCTP Stream Control Transmission Protocol
SMB Server Message Block
SSL Secure Socket Layer
SST Structured Stream Transport
TCP Transmission Control Protocol
TFO TCP Fast Open
TLS Transport Layer Security
UDP User Datagram Protocol
VPN Virtual Private Network

1   Introduction

In modern communication networks, reliable, secure and high-performance internet transport protocols have become a cornerstone of communication over the internet, being essential for applications ranging from web browsing and multimedia sharing to enterprise level file sharing. The Transmission Control Protocol (TCP) [1] has served as the solution of choice for reliable communication for over four decades. Side-by-side with TCP, the User Datagram Protocol (UDP) [2] has been used for applications with high requirements on latency, with no guarantees of reliability. As the application demands move towards higher throughput, lower latency and increased security demands, the inherent limitations of TCP have become apparent.

One demanding use of internet transports is remote file access, present at both the enterprise and consumer level. The Server Message Block (SMB) [3] is a widely adopted protocol for sharing files over a network, with implementations available in Windows, Linux and embedded systems. Traditionally, the SMB protocol has used TCP to ensure reliable delivery [3]. However, the increasing demand of modern use cases, with mobile users and more security demanding applications, is increasingly highlighting the limitations of TCP.

Recently, a new transport protocol, QUIC, has been developed to address the issues and shortcomings of TCP. QUIC was initially developed by Google [4], and later standardized by the Internet Engineering Task Force (IETF) in Request for Comments (RFC) 9000 [5]. QUIC implements transport level multiplexing, encryption and authentication via TLS 1.3, and improved connection set up latency as parts of its design [5, 6]. QUIC has already been widely deployed as a foundation for HTTP/3 [7], supporting large-scale web applications. However, QUIC was not only developed as a transport protocol for web applications, but also as a general-purpose transport protocol [5]. One of these applications is to serve as a transport protocol for the SMB protocol.

Although TCP has served as the reliable transport protocol for SMB during the last decades, it has several well-known drawbacks in modern networking environments. First, TCP suffers from Head-of-Line (HOL) blocking, which can significantly impact performance in lossy environments. Second, TCP combined with TLS requires additional messages during the connection set up, the handshake, negatively affecting the latency of connections, especially for short communications. Finally, TCP heavily suffers from protocol ossification: different middleboxes, firewalls and operating system (OS) implementations make changes difficult and slow to propagate [4].

Even though the improvements brought on by QUIC have been widely studied in the context of web traffic [8–10], no studies have been done on the performance of SMB using the QUIC transport protocol. While the support for alternative transport protocols in the SMB protocol, mainly Remote Direct Memory Access (RDMA) and QUIC, has been defined and implemented by Microsoft [3], To the author's knowledge, there have been no formal efforts into researching and comparing the performance of these alternative transports. Additionally, there is no known implementation of SMB over QUIC for Linux based systems, with currently the only available implementation being in Windows machines. SMB suffers in the context of network reachability, as

many internet service providers choose to block the port used by SMB due to several exploits targeting this specific port, resulting in standard SMB traffic being blocked as well [11].

A possible solution to address the limitation present in TCP is using QUIC as an alternative transport protocol. Via combining QUIC's multiplexed transport, and SMB semantics for remote file access, this approach could mitigate the adverse effects of HOL blocking, improve latency of the connection and improve deployment to user space applications. Additionally, since QUIC is designed with mobile users in mind, it could improve the reliability of the SMB protocol in diverse mobile environments, especially for mobile users. By moving from the traditional TCP port 445 to UDP port 443, SMB over QUIC would essentially mimic HTTP/3 web traffic on the transport level, circumventing the blocking of SMB traffic that is commonly done for TCP.

The objective of this thesis is to design, implement and benchmark a QUIC transport layer for an SMB server. The aim is to develop a working prototype that can be used to test performance and interoperability and compare it against standard SMB over TCP solutions. In order to develop this solution, the thesis will implement a QUIC transport layer. This will be accomplished by using the MsQuic library [12]. This QUIC transport layer will then be integrated into Fusion SMB, an SMB server developed by Tuxera [13]. The thesis will conduct a series of performance benchmarks and interoperability tests, focusing on throughput and compatibility. The main research questions this thesis aims to answer are:

1. Is it feasible to implement SMB on top of QUIC and what are the hurdles in doing so?

2. What are the benefits of a QUIC-based implementation as compared to the traditional TCP-based implementation?

3. How does the performance of SMB over QUIC compare to the TCP-based implementation?

This thesis is limited to supporting a QUIC transport layer using MsQuic. Other potential QUIC libraries are beyond the scope of this thesis, as there is no unified interface between the libraries, meaning the work required to integrate one library is not transferable to another. The performance evaluations are limited to different throughput tests, including several different workloads, representative of real-world enterprise and customer use cases. Other tests that will not be considered are latency and benchmarking connection creation, as the only available client is the Microsoft Windows SMB over QUIC client, which is limited in the number of connections. Questions such as large-scale deployments, multi-node setups and integration into cloud architecture remain outside the scope of this thesis.

The rest of this thesis is structured as follows. Chapter 2 provides background information on internet transport protocols and network storage protocols. Chapter 3 introduces QUIC, the motivations for its creation and its architecture. Chapter 4 gives an overview of the SMB protocol. Chapter 5 describes the design and implementation of a QUIC transport layer for the Fusion SMB server. Chapter 6 presents the test

environment, workloads and the resulting performance numbers. Finally, Chapter 7 discusses the findings and possible future work.

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). Second, this section covers data streams and the Structured Stream Transport (SST), as well as the Stream Control Transmission Protocol (SCTP) which introduces multiplexed stream transport. An overview of HTTP and TLS is given to set the foundation for understanding the QUIC protocol. Finally, this section also covers distributed file systems (DFS), and compares the Server Message Block (SMB) and Network File System (NFS) protocols.

2.1 Internet transport protocols

2.1.1 Transmission Control Protocol

TCP, as outlined in RFC 793 [1] 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. 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, fitting into the protocol stack on top of the Internet Protocol (IP) [14]. 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 duplication. TCP works with the assumption that the underlying protocol is unreliable and may lose, fragment or reorder sent datagrams [1].

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 receive an ACK within a timeout period, the missing segment will be retransmitted. A checksum is included with each segment, ensuring that the datagram corruption during transport is detected. If data corruption is detected, the receiver will discard the damaged segment and rely on the retransmission mechanic to recover. 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 TCP demand that 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 4-tuple containing the source and destination network addresses and ports is used to identify the connection. Using this mechanic to uniquely identify connections, allows for multiple processes to simultaneously communicate in parallel using TCP [1].

The TCP header, Figure 1, encodes the functionality of TCP. It follows the IP header in a datagram. The header is usually 20 bytes long but can be extended using

Figure 1: The TCP header

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 refers to the first data byte in the data segment. If the SYN flag is set, this is the beginning of a new connection and the sequence number in this case refers 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 were 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 [15].

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 [1].

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 client's ISN. The server responds with a packet with the SYN and ACK bit set, acknowledging the client's 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 its side [1].

2.1.2 User Datagram Protocol

UDP, which was defined by RFC 768, is designed to enable programs to transmit self-contained messages, known as datagrams, over a packet-switched network. UDP is designed to run on top of IP [14], using IP addresses and port numbers for addressing. 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

Source Port 2 Bytes	Destination Port 2 Bytes
Length 2 Bytes	Checksum 2 Bytes
Data	

Figure 2: The UDP header

14

port, as well as the length of the datagram and a checksum to verify the received datagram [2]. 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 [16]. The 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 Streams

A data stream is defined as a set of digital signals that is used to transmit information [17]. In practice, when talking about internet transport protocols, a data stream refers to an ordered series of bytes that is used to transmit information. For TCP this holds true, even though TCP splits the transmitted data into packets, from the application point of view the information is a stream of bytes. In comparison, UDP does not offer this same abstraction, as each datagram exists independently of one another, but a series of these datagrams could be seen as a type of data stream, albeit an unordered one. Having only one stream per connection can lead to issues in the transport layer, such as HOL-blocking, which is discussed in Section 3.1.1. The solution of opening multiple connections comes with the drawbacks of large overhead and resource consumption for the peers, with each connection taking up a separate socket on the client machine. As a result, protocols have emerged that aim to extend the data stream capabilities of TCP and UDP, and in the process solve some of the issues present in these legacy transport protocols.

2.2.1 SST

SST was introduced in [18] with the aim of extending TCP semantics by introducing a hierarchical structure, allowing child streams to be created on top of an existing stream. These streams can be created without having to set up new connections, making them much more lightweight than the TCP equivalent of opening multiple connections. SST supports multiplexed streams, allowing the streams to exist in parallel, alleviating the issues of HOL blocking as each stream uses independent flow control. SST also supports independent stream prioritization as well as reliable and best effort (unreliable) delivery [18].

There were three primary reasons for creating the new protocol, SST, that splits a single connection into multiple data streams. First, some protocols provide multiplexing on top of TCP but suffer from HOL blocking as loss will block all data transfers until retransmission. This issue arises as reliable communication protocols guarantee that all bytes will be delivered in order, without any missing chunks. Retransmissions, especially in high latency environments, causes delivery delays for all of the data in the affected stream, or in the case of TCP, the whole connection. Secondly, while TCP is better suited for long connections and large data transfers, UDP excels in short

communications where the overhead of setting up connection via TCP's three-way handshake provides significant overhead. Finally, some protocols such as HTTP/1.0 have bad utilization of network resources, as they are designed to run on top of TCP and open a new TCP connection for each resource that is loaded. This was somewhat remedied in HTTP/1.1 with the serializing of requests over one connection, but at the time of the SST's creation it had yet to be widely implemented [18].

As can be seen in Figure 3, the SST architecture consists of three parts, the Stream, Channel and Negotiation Protocols. The Channel Protocol is responsible for the delivery channel providing the best-effort delivery service, congestion control and packet sequencing. The Negotiation protocol is responsible for setting up the channel and negotiating protocol details. Most interesting is the Stream Protocol, building on top of the other protocols to implement the Stream structure that SST is based upon. SST implements the concept of child streams. Child streams are created on top of a parent stream but are otherwise equivalent to the primary stream in all but name. Child streams allow multiple streams to exist on top of one connection, and they are designed to be lightweight. This mitigates the issue of HOL blocking between streams, as loss on one stream still enables the other streams to keep transmitting while the erroneous stream independently recovers. SST also implements a second type of stream, while the normal stream is semantically like TCP, the second stream type, the ephemeral stream, is closer to UDP. Ephemeral streams allow the application to have short communications as part of a larger connection, without any additional overhead [18].
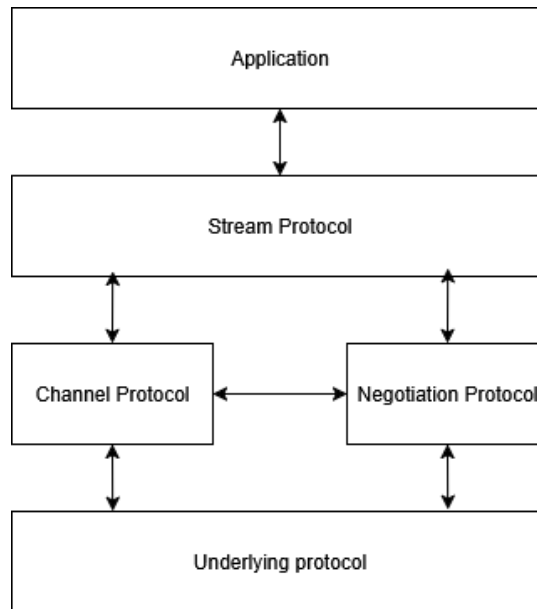

Figure 3: The SST architecture

2.2.2 SCTP

The Stream Control Transmission Protocol (SCTP), was first defined in RFC 4960 [19], which was then later updated by RFC 9260 [19]. SCTP was originally implemented to carry Public Switched Telephone Network (PSTN) messages but was later standardized into a general-purpose transport protocol. SCTP is a connection-based protocol, with connections referred to as associations, and is designed to be run on top of IP. As a contrast to TCP, which is designed to transport a byte stream, SCTP focuses on delivering individual messages between applications. A message is sent on one end in one operation, and the whole message is received on the other end, in one operation. A series of these messages then form the stream inside SCTP. SCTP natively supports multiplexing of these streams, addressing TCP's issue of HOL blocking. This is a significant improvement for PSTN messaging, where control messages must not be delayed by data loss in other streams [20].

In SCTP user messages are split into SCTP DATA chunks, which are bundled together with SCTP Control chunks, together with a common header to form the SCTP packet. While there may be multiple chunks inside one SCTP packet, each chunk only contains the data of one message. Multiple sequences, streams, of these chunks may be sent in parallel over one association, enabling multiplexed streams. SCTP supports both ordered and unordered message streams, allowing messages to potentially be delivered out-of-order, which may be preferable for certain kinds of data, such as control data [20].

SCTP supports multihoming, allowing the protocol to simultaneously bind to multiple different addresses and ports. This improves resilience, allowing the traffic to migrate to a backup path in case the primary network path is experiencing issues. This migration allows continued communication without breaking the association [20].

2.3 HTTP and its evolutions

The Hypertext Transfer Protocol (HTTP), is a protocol stemming from the early days of the World Wide Web, defining a protocol that would allow a server and a client to talk to each other. HTTP allows a client to request different resources from a web server, such as JavaScript, CSS, images, multimedia files or any number of different resources. The high-level idea of HTTP is that a client requests a resource from the server, and the server, if possible and permitted, responds with the requested resource. HTTP defines the formats of these messages, which are written in American Standard Code for Information Interchange (ASCII), making the message human readable directly on the wire, without having to interpret them. There are a set of HTTP request methods defined for the message, such as the GET method for requesting a resource, and POST for sending data to the server [16].

A comparison of HTTP version can be seen in Table 1. The first version of HTTP, HTTP/1.0 was defined in RFC 1945 [21]. HTTP/1.0 used non-persistent connections and had no concept of multiplexing over a single connection. It was a basic protocol, working on a request-response principle, with each request opening its own TCP connection. HTTP/1.1 defines three request methods. GET to request data from the

server, POST to post data to the server and HEAD, which is similar to GET, but requests that only the HTTP header be returned in the response, omitting the body from the response [21]. What is notable for HTTP/1.0 is that it did not suffer from the HOL-blocking of TCP, unlike most of its successors. Because each request opened a new TCP connection, there was no concept of HOL blocking between requests as the requests were entirely independent of one another. However, this was not very efficient resource usage, as perform multiple three-way TCP handshakes to load a simple web page added significant overhead.

Table 1: Comparison of HTTP Versions

Version	HTTP/1.1	SPDY	HTTP/2	HTTP/3
Transport	TCP	TCP	TCP	QUIC
Multiplexing	Pipelining	Over TCP	Over TCP	QUIC multiplexing
Compression	No	SPDY specific	HPACK	QPACK
Server Push	No	Yes	Yes	Yes

HTTP/1.0 was iterated upon and became HTTP/1.1, which was initially defined in RFC 2616 [22]. HTTP/1.1 improved upon HTTP/1.0, adding support for several new features. First, persistent connections allow multiple requests to be sent over one connection, cutting down on the number of connections that need to be opened to load a set of resources. Second, pipelining allows multiple requests to be made at one time, without having to wait for the response to the previous request. This was not quite yet true multiplexing, as the responses need to be returned in the same order as the request were made, causing HOL blocking of subsequent requests in case of loss [22]. Third, with the Introduction of HTTP over TLS, HTTPS, it was now possible to set up more secure, encrypted communications, instead of transmitting data in cleartext [23].

2.3.1 SPDY

The SPDY protocol was designed by Google to improve latency of web pages, as compared to HTTP/1.1. The SPDY protocol was designed to work as an extension to HTTP/1.1, slotting into the session layer and largely defining how HTTP traffic is sent over the wire, with some additional improvements to the HTTP protocol. In practice SPDY introduced three improvements, multiplexed streams, header compression and server push capabilities [24].

SPDY introduced multiplexing of HTTP streams, allowing for an unlimited number of multiplexed streams over one TCP connection. The multiplexing works by splitting the data into frames, which are interleaved and transmitted over a single TCP connection, as can be seen in Figure 4. As compared to HTTP/1.1's pipelining, the true multiplexing of SPDY allows the server to respond to requests in any order, instead of being forced to follow the specific request order. This allowed for a second improvement, where a client could attach a priority value to the requests, signaling to the server which resources should be prioritized, especially in congested channels. To limit the number of bytes that must be sent over the wire SPDY implemented HTTP

header compression, sending the header in a compressed format instead of ASCII. Additionally, SPDY added server push and server hint features. Server push allows the server to send a resource to the client, without the client having request the resource first. Server hint is similar to server push, but instead of sending the resource, the server sends a hint to the client with a suggestion that the client should request a specific resource [24].
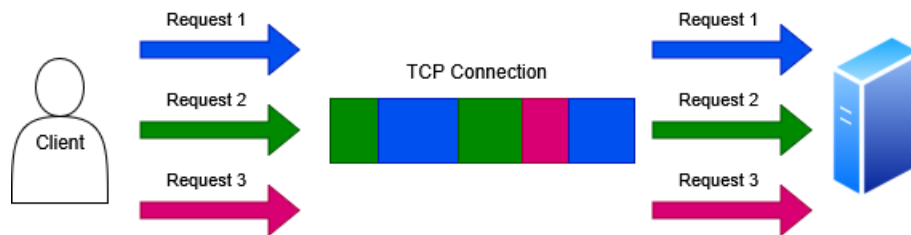

Figure 4: HTTP multiplexing

2.3.2 HTTP/2

HTTP/2 is standardized in RFC 9113 [25], is the evolution from HTTP/1.1. It implements some of the ideas from SPDY, with the original specification for HTTP/2, RFC 7540 [26] containing acknowledging for the contributions made by the SPDY team. The HTTP/2 protocol uses the same HTTP semantics as HTTP/1.1, but like SPDY improves on the transport of the HTTP data between endpoints. HTTP/2 implements streams and stream multiplexing, stream prioritization, header compression and server push capabilities. Like SPDY, HTTP/2 supports multiple independently multiplexed streams, allowing multiple parallel requests over one TCP connection. Stream priority allows the client to signal which requests are more important to be filled, and in case no priority is given, the server may choose to determine priority based on other information [25]. For header compression, HTTP/2 implements a dedicated compression algorithm called HPACK, defined in RFC 7541 [27]. HPACK not only implements compression, but the algorithm also helps in reducing redundant header fields, by preventing the same static header fields from being sent repeatedly [27]. HTTP/2 supports server push capabilities, allowing the server to send unprompted resources to a client, usually by estimating the requests that the client likely will make in the future, in that way improving latency [7].

Even though HTTP/2 implements stream multiplexing, it still suffers from HOL blocking since HTTP/2 streams are created on top of a single TCP connection. This means that the requests won't block each other, but in case of packet loss, the TCP HOL blocking will cause all requests to stall while waiting for [25]. This highlights the fact that despite improvements in the HTTP protocol, the HOL blocking issue may not be solved as long as the protocol is running on top of TCP.

2.3.3 HTTP/3

HTTP/3, as defined in RFC 9114 [7], is as of writing the latest evolution of the HTTP protocol. It was specifically designed to be run on top of QUIC, taking advantage of the improvements offered by the QUIC transport protocol. Like HTTP/2, the HTTP semantics stay largely the same, but improvements are made to take advantage of the new transport. HTTP/3 includes support for transport level multiplexing, transport level encryption and connection migration. The most significant advantage offered by HTTP/3 is the improvements to transport level HOL-blocking. Because HTTP/3 takes advantage of native QUIC streams for its multiplexing, which are entirely independent on the transport level, packet loss won't affect unrelated streams. This allows HTTP/3 to take full advantage of multiplexing requests to load resources, without bottlenecking from the transport layer. QUIC offers an improved connection set up time as compared to HTTP/2, as HTTP/2 uses TCP + TLS, requiring multiple RTTs to establish a secure connection. HTTP/3 utilizes the fact that QUIC has the TLS handshake integrated into the QUIC transport handshake, potentially reducing connection set up time to 1 RTT. In some cases, it is even possible to use 0-RTT data transfer to instantaneously transfer data without waiting for the connection to be set up. Connection migration of QUIC allows connection to stay active for mobile users, even in the face of network changes [7]. As with HTTP/2, HTTP/3 uses header compression, but HPACK has been replace by QPACK, which improves on HPACK by better taking advantage of QUIC's features, improving HOL blocking issues with only a slight degradation in compression rate [28].

2.4 TLS

The Transport Layer Security (TLS) protocol is an evolution from the Secure Socket Layer (SSL) [29] protocol. TLS aims to secure communication between two endpoints. The TLS protocol provides three properties to data communication. First, authentication: the server side of the communication is always authenticated, and the client side may optionally be authenticated. Second, confidentiality: TLS ensures that data sent between the endpoints is encrypted, ensuring that only the intended recipient can decrypt and read the data. Finally, integrity: data transmitted through a TLS tunnel cannot be tampered with, without alerting the recipient [30].

The TLS protocol consists of two components, First, the handshake protocol is used to establish the connection, being responsible for authentication, negotiation and creating the key material used for encryption. Second, the record protocol is responsible for encrypting and transmitting the data. The name comes from the fact that the data to be transmitted is split into a series of records, all of which are individually encrypted and transmitted. The TLS protocol is designed to be a general-purpose security protocol, as it only requires to be ran on a reliable transport, and in principle any application protocol may be run on top of TLS [30].

The TLS handshake is the most important part when establishing a TLS session. In TLS 1.2 and earlier the handshake required 2 RTTs before application data could be sent securely, but since TLS 1.3 the handshake has been streamlined into 1-RTT,

with the possibility of 0-RTT data transfer, at the cost of perfect forward secrecy. The TLS 1.3 handshake is outlined in Figure 5 where the asterisk indicated optional or alternative parts of the message, curly brackets indicate handshake secret protection and square brackets indicate full application protection of data. The handshake begins with a ClientHello message, containing a nonce, the protocol version and either a pair of Diffie-Hellman key shares or a set of pre-shared key labels. The server responds with a ServerHello, containing key shares, certificate if it is used in authentication of the server and possible extensions, such as a CertificateVerify request in case the server requests that the client be authenticated via a certificate as well. The certificate and CertificateRequest as well as Extensions are protected with keys derived during the handshake process. These are different from the secrets used to protect application data and do not guarantee perfect forward secrecy. The combination of ClientHello and ServerHello is used to create the shared keys. Finally, the client responds, acknowledging that the process is finished [30].
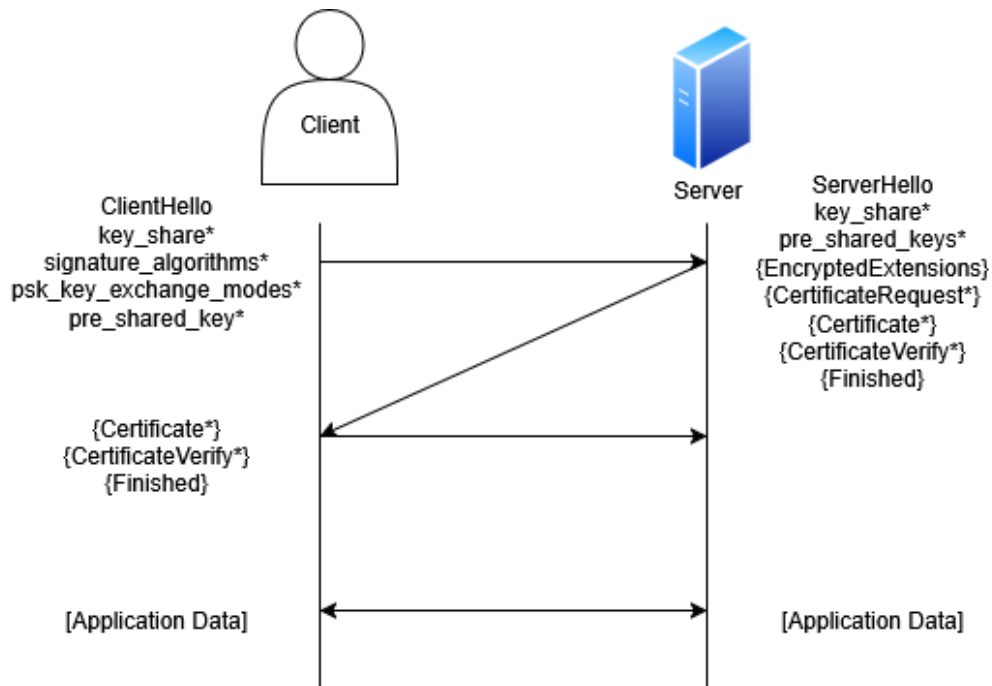

Figure 5: The TLS 1.3 Handshake

Normally the handshake adds additional overhead to connection set up. With TLS 1.2 and lower adding an additional 2+ RTTs of latency, and TLS 1.3 adding 1-RTT of latency. However, the QUIC protocol integrates the TLS 1.3 handshake into the transport layer protocol, allowing both handshakes to be made in parallel, negating the latency impact, but more on this in Chapter 3. Once the handshake is finished and the TLS session set up, the record protocol takes over. The record layer splits the application data into appropriate chunks, which are then encrypted and integrity protected. Finally, the record layer transmits these records via the underlying, reliable transport.

The first versions of TLS, TLS 1.0 and 1.1, defined in RFC 2246 [31] and RFC

4346 [32], improved upon SSL but were deprecated in 2021 via RFC 8996 [33], due to the fact that these algorithms rely on weak cryptographic algorithms. The TLS 1.2 protocol improved upon its predecessor by introducing more secure cryptographic algorithms and authenticated encryption [34], and removing backwards compatibility with SSL via RFC 6176 [35]. Finally, as mentioned earlier in this section, TLS 1.3 is the latest version of the TLS protocol. It removed support for legacy cryptographic algorithms, added the 1-RTT handshake and 0-RTT data transfer, among other cryptographic improvements [30].

2.4.1 DTLS

As mentioned in Section 2.4, the TLS protocol is designed to run on top of a reliable transport protocol. However, it is quite common for applications to run on top of UDP, which as mentioned in Section 2.1.2, makes no guarantees for reliability. To solve this problem and provide protection for datagram-based applications, Datagram Transport Layer Security (DTLS) is designed to overcome the challenges of securing an unreliable transport. There are four main reasons the TLS cannot be used directly on top of UDP. First, TLS uses an implicit sequence number as a nonce, meaning that TLS records being delivered out of order causes the decryption process to fail. Second, the TLS handshake is rigid, the right messages must arrive in the right order for the handshake to succeed. Third, handshake messages may be larger than one datagram, causing fragmentation. Finally, UDP is susceptible to Denial-of-Service (DoS) [36].

To handle packet loss, DTLS uses a retransmission timer to enable retransmission of lost packets, such as part of the handshake. Reordering is mitigated by assigning an explicit packet number, enabling the receiving party to process the received messages in the correct order. DTLS adds support for fragmentation of handshake messages, the sending party can split the handshake message into multiple parts that all fit inside single datagrams, and the receiving party can then reassemble the handshake message from the fragments. To combat DoS attacks, DTLS implements a HelloRetryRequest message, which contains a cookie, that then must be included in the ClientHello message to be accepted by the server. Optionally, DTLS includes replay detection and protection [36].

DTLS is designed to be as close to TLS as possible in capabilities and functionalities. DTLS follows TLS versioning, with the latest, DTLS 1.3, being defined in RFC 9147 [36]. Like TLS 1.3, DTLS 1.3 implements the improved 1-RTT handshake, enforces stronger cryptographic algorithms than earlier versions and adds support for 0-RTT data transfer, however 0-RTT data transfer requires that replay protection is enabled on the connection [36].

2.5 DFS

Distributed File Systems (DFS) allow clients to communicate with servers, enabling the clients to manipulate the files as if they were stored on their local machine. DFS's abstract away the complexities of distributed data access, exposing a set of file operations to the clients, allowing them to create, delete, read and write files among

other operations. The DFS ensures consistency, reliability and access between different clients and access points [37]. Currently, the two most widely adopted distributed file systems are the Server Message Block (SMB) [3] and the Network File System (NFS) [38]. Both protocols provide file system access over a network, but they differ in design, environment and features.

2.5.1 Server Message Block

The Server Message Block (SMB) protocol is a stateful protocol, and is a protocol designed for sharing files, printers and other resources over a network. It was originally created in the early 1980s and later adopted and extended by Microsoft [39]. The protocol has undergone several iterations since that time. While the underlying concepts of SMB have stayed the same, later revisions, that is SMB version 2.0 and forwards, differ heavily from earlier versions. SMB 2.0 introduced different packet formats, and reduced the chattiness of the protocol, among other improvements. SMB 3.0 introduced encrypted traffic, multichannel support and integration with RDMA in the form of SMB Direct. Since SMB 3.1.1 the protocol has added support for using the QUIC protocol as an alternative transport [3].

The main distinguishing characteristics of the SMB protocol is its stateful design. The server maintains information about the state of a session, such as file handles, locks and leases, over the lifetime of a session. This allows the SMB protocol to guarantee consistency and helps in coordinating multiple, concurrent clients. As the development of the SMB protocol has largely been a result of Microsoft's effort, the protocol is deeply integrated into the Windows ecosystem, supporting NTLM and Kerberos authentication, and integrates as part of Active Directory Domain services [3].

2.5.2 Network File System

The NFS protocol was introduced by Sun Microsystems in 1984, aiming to be a protocol for enabling remote access to filesystems between different UNIX machines. NFS was designed to be a simple and open filesystem, making it easy to port to different machines. In contrast to the SMB protocol, the NFS protocol was designed to be a stateless protocol. This simplifies processing, as each request contains all necessary information to complete the request, and if the request fails the client may simply try again. Recovery from error states is simpler as no sessions have to be re-established. The stateless design come with drawbacks, for example two concurrent clients trying to write the same file could cause the writes to be interleaved, causing unintended outcomes, as the writes are simply processed in the order they arrived [40].

Over time the NFS protocol has evolved, with the latest version NFSv4.2 being defined in RFC 7862 [41]. NFSv4 introduced stateful capabilities into the protocol, and includes features such as file locking, multipath and compound operations, alongside performance improvements and improved authentication support for stronger security [38]. NFSv4.1 introduced NFS sessions and parallel access to data, as well as improving support for clustered solutions [42], while NFSv4.2 introduces support for server-side operations, and sparse files [41].

Table 2: Comparison of SMB and NFS

Protocol	SMB	NFS
Environment	Windows	UNIX
Resources	Files, directories, printers, network resources	Files and directories
Sessions	Stateful	Stateless v2/v3, Stateful v4

Table 2 gives an overview of the major differences between SMB and NFS. SMB provides many different features, a natively stateful protocol for sharing all kinds of resources, and a solution that is very Windows centric, with UNIX based system requiring a third-party solution for support, such as Fusion SMB or Samba. In comparison, NFS is a lightweight, easy to set up protocol that is purely focused on file and directory sharing, and which is integrated into many different UNIX based operating systems.

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, TCP has been the de-facto solution for reliable and secure (when combined with TLS) communications. However, TCP was developed in a time when security and latency were not primary considerations, at least not in the same way as in today's landscape. Over the years there have been efforts to enhance TCP, such as multipath TCP [15] and using both TCP and TLS in HTTPS [23] to improve security. This section of the thesis will cover QUIC, a transport protocol developed to overcome the limitations of TCP and improve performance [4].

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 [5]. QUIC is designed to address the issues experienced by TCP, with a focus on optimizing for web traffic, with the development being largely done hand-in-hand with HTTP/3. The main issue of TCP that needs to be overcome is the HOL blocking, but QUIC also aims to improve on other aspects of the protocol, 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 [4].

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

3.1 The motivation for a new transport protocol

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

3.1.1 Head-of-Line Blocking

As discussed in Section 2.2, TCP is negatively affected by HOL blocking. To get around this limitation, modern network protocols, such as HTTP/2 [25], have introduced measures to combat this 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 issue in earlier versions of HTTP. However, HTTP/2 still suffers from transport level HOL blocking, as the multiplexed stream is sent over a single TCP connection. As a result, a single lost packet in the TCP stream still causes all other unrelated streams over the same connection to stall, even if their packets were successfully delivered, until the offending packet is retransmitted and received. This in practice means that many of the improvements made by HTTP/2 in this regard are negated by the issues of TCP, particularly in lossy or high latency environments [43].

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 applications use TLS for security, and historically the TLS 1.2 handshake and set up add two additional RTTs of delay. While network bandwidth is ever increasing, much of the communication done on the internet consists of short dialogues, that are significantly impacted by the additional latency brought on by the TCP plus TLS handshake [4].

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 [30]. Even with these enhancements, the TCP plus TLS handshake takes a minimum of 1.5 RTTs before data can be transmitted, due to the separation of the connection and security handshake. There is an extension to TCP called TCP Fast Open (TFO) that enables data to be sent during the TCP handshake, in the SYN and SYN-ACK packets. For TLS this means that the ClientHello can be carried in the initial SYN packet [44]. The main problem with TFO is that a significant amount of middleboxes will drop TCP packets with unknown SYN data or options. If any of these devices are on the path between the endpoints, using TFO is simply not possible [45].

3.1.3 Protocol Ossification

A big hurdle in deploying new protocols and extensions to existing ones is the protocol ossification of protocols on the internet. There exists a large number of different middleboxes, such as Network Address Translators (NATs) and firewalls that are all part of the network. These devices may be overly conservative, dropping or modifying packets that do not conform to their assumptions. 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 the middleboxes on the path. To get around this, protocol designers must design their protocols from the ground up to be resistant against middlebox interference, such as is the case with QUIC encapsulating its protocol inside UDP as an anti-ossification measure [46].

A related issue of rolling out enhancements to existing protocols is that the network stack tends to be part of the OS kernel. The networking stack is tightly coupled with the OS, requiring OS updates or upgrades to implement changes to existing protocols.

With today's upgrade frequency, it can take years to roll out simple changes to the networking stack. QUIC moves the implementation of the protocol into 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 [4].

Additionally, the TCP header is limited to 60 bytes, of which 20 bytes is taken up by the mandatory fields of the header, which leaves 40 bytes of space for the options. However, all existing options consume space, and seeing as the space is so limited, it is not difficult to exceed the available space. For example, SACK can use as much space as is available, or timestamps that take up 10 bytes, so one quarter of the total available space. In practice this leaves very little space for extending the protocol, further increasing the difficulty of implementing new features [47].

3.2   Background and evolution

As discussed in Section 3.1, the combinations of TCP, TLS and HTTP/2 are plagued by issues that are difficult to circumvent without major overhauls or extensions to the individual protocols. Due to protocol ossification, this is increasingly difficult. With this in mind, a new protocol was created, aiming to solve the issues of HOL blocking, improve latency and circumvent protocol ossification. The result was the protocol that would later be standardized into QUIC, early on known as gQUIC. QUIC began development back in 2012, by Jim Roskind, an engineer at google. Initially the motivation for developing a new transport protocol was to improve support for the now deprecated SPDY protocol, which had initially been designed to run on top of TCP [48]. QUIC was designed to run over UDP, by encapsulating the protocol frames into UDP datagrams, and encrypting the contents. This way the protocol could effectively sidestep the issues of middlebox interference, allowing for rapid deployment without any necessary modifications to existing infrastructure. To combat the issue of HOL blocking, QUIC implements transport level multiplexing of data streams, allowing multiple independent streams to exist over one connection. Packet loss in any of the data streams would not affect any of the others, blocking only one stream, while waiting for retransmission. QUIC uses a combined connection and cryptographic handshake, minimizing the latency of establishing a new connection. While TCP uses IP-port tuples to identify connections, this does not allow for changes in the underlying network, such as might be the case for mobile users. If the IP or port of the user changes during the lifetime of the connection, the connection is dropped and must be reestablished. To combat this QUIC uses Connection IDs to identify the connection, allowing the connection to resume when there is some change in network. QUIC was widely deployed on Google's front-end servers, and by 2017 it was already estimated that QUIC represented 7% of global internet traffic [4]. Since then, the adoption of QUIC has been rapid and according to statistics by Cloudflare, QUIC makes up more than 30% of web traffic [49].

QUIC was submitted to the IETF for consideration in 2016, and a working group was created for the purposes of standardizing QUIC. The goal of the working group was to create a general-purpose transport protocol that contained the benefits of gQUIC. The custom cryptographic handshake was replaced with TLS 1.3, the packet header

was reworked into two types, a long and a short header. The header is mostly encrypted to prevent middlebox interference. Loss detection and congestion control mechanics were updated, flow control semantics were separated into per stream and per connection limits and version negotiation was introduced to improve forward compatibility of the protocol [5]. In the end the protocol was standardized in several RFCs. RFC 9000 [5] defines QUICs core transport mechanics, RFC 9001 [6] defines the use of TLS 1.3 and RFC 9002 [50] describes the loss detection and congestion control algorithms used by the protocol. In addition, RFC 8999 [51] defines some version-independent properties of QUIC, aligning QUICK packets, headers and versioning between different versions of the protocol. Following the standardization the adoption of QUIC has been quick. Chromium, and by extensions all chromium-based browsers has supported QUIC since before it was standardized [52]. Both Firefox [53] and Safari [54] added support for QUIC soon after the standards were published. QUIC has shown the viability and potential of deploying network protocols to the user space, enabling rapid adoption without slow OS kernel updates.

3.3   Architectural Overview

The QUIC protocol is designed to be a general-purpose, secure and multiplexed transport protocol, working on top of UDP. In comparison to TCP, which usually is part of the OS kernel, QUIC is typically implemented in the user space, enabling quick iteration and deployment of protocol enhancements. This section describes QUIC's position in the networking stack, the different architectural parts of the protocol and the basic elements of the protocol's operations.

QUIC packets are directly encapsulated inside UDP datagrams. This design has both practical and implementational advantages. When deploying QUIC, the fact that the wire image of QUIC packets is identical to that of UDP datagrams, means that they pass seamlessly through middleboxes and firewalls, without suffering the adverse effects of protocol ossification as is often the case in TCP. As discussed in Section 2.1.2, UDP is a barebones protocol, with the minimum overhead needed to transmit datagrams. This works to the advantage of the designer when building a protocol on top of UDP, as this give the freedom needed to implement entirely custom semantics, without risking interference from the underlying protocol. UDP provides the basic datagram services that are then enhanced by the QUIC protocol, ensuring a secure, reliable and performant protocol [4].

From an architectural perspective, QUIC incorporates three main layers, a transport layer, a security layer and an application interface. From the bottom up, UDP provides minimal mechanism for transmitting datagrams, without any guarantees. On top of this QUIC implements its own transport layer, handling the multiplexing of data streams, ensuring reliable and in-order delivery of data as well as connection management mechanics [5]. QUIC's security layer is fully integrated into the protocol, using TLS 1.3 for encryption of on-wire traffic, as well as authentication and authorization of peers, ensuring secure communication between endpoints. The security layer also protects most of the packet header, leaving only the necessary info for routing and version control visible on the wire [6]. The final layer exposes a standardized

application interface that can support virtually any application-layer protocol, the most prominent one being HTTP/3 [25].

One of the main innovations made by the QUIC protocol is the concept of transport-level, multiplexed and independent data streams. Any QUIC connection may contain one or multiple data streams, handled entirely as their own independent object. This helps mitigate the HOL blocking issues, as if the application is using multiple streams, when loss occurs on a stream, only the stream on which the loss occurred will be blocked. While the lossy stream is stalling and waiting for retransmission, the other streams can continue sending as normal. QUIC uses per connection limits for the number of streams and per stream flow control limits [5].

Connection management and identification in QUIC differs from the IP-port tuple combination that is traditionally used in TCP for identifying connections. QUIC connections are identified via a connection ID, which is independently selected by the endpoints. The purpose of the connection ID is to allow the connection to survive changes in the underlying network, for example when a mobile user moves from a local network to cellular. The migration is done transparently and securely, allowing the application to continue operations without interruption or having to reestablish the connection [5].

3.4   Packet and Frame Structure

QUIC packets are contained inside UDP datagrams, and together with the encryption and integrity protection provided this gives robust protection against protocol ossification. As compared to TCP, where segmentation and reassembly of packets are transparently done as part of the transport protocol and not accessible to the application, QUIC defines a large set of different frame types for different, distinct roles in connection establishment, data transfer and protocol logic management. Depending on the packet type QUIC uses either a long header or short header.

QUIC packets are divided into two general categories, those that use the long header and those that use the short header. The long header is used during the establishment of the connection, being used in the Version Negotiation, Initial, 0-RTT, Handshake and Retry packets. The long header, as seen in Figure 6, contains the necessary data for these functions. The first bit of the long header is set to 1 to indicate that this is a long header. The fixed bit following this is always set to 1, except for Version Negotiation packets. The Long Packet type indicates the type of packet, and the 4 type specific bits are dependent on the packet type. Following this, the version ID field indicates the specific QUIC version this packet is using, and finally the destination and source ID length and IDs contain the identifiers for the source and destination of this packet. Following the header is, if relevant, the specific packet type payload. In addition, the Initial, 0-RTT and Handshake packet types include a packet number length and packet number field. In comparison, the short header is more compact, reducing per-packet overhead. The Header Form bit (set to 0 in the short header case) and the Fixed bit is the same as for the long header. The Spin Bit that follows may be used to make on path estimations of the RTT. Following the two reserved bits is the key phase bit, used to keep track of the keys that are used to protect the packet. The

packet number length is the length of the packet number field minus one. Lastly is the destination ID again indicating the recipient, the packet number used for protocols semantics and finally the packet payload [5].

QUIC Long Header

Header Form 1 bit	Fixed Bit 1 bit	Long Packet Type 2 bits	Type Specific Bits 4 bits	Version ID 32 bits
DCID Len 8 bits	DCID 0-160 bits		SCID Len 8 bits	SCID 0-160 bits

QUIC Short Header

Header Form 1 bit	Fixed Bit 1 bit	Spin Bit 1 bit	Reserved 2 bits	Key Phase 1 bit	Packet Number Length 2 bits
DCID 0-160 bits	Packet Number 8-32 bits	Payload 8+ bits			

Figure 6: The QUIC header formats

The long header packet type numbers differ between QUIC version 1 and version 2, as is outlined in Table 3. Initial packets are used to initiate the connection and transmit the initial TLS handshake. The Handshake packet carries the subsequent cryptographic messages, after the initial exchange. In case the connection attempt is a resumption, the client may use the 0-RTT packet type to transmit encrypted application data as part of the handshake, basing the encryption on previously established keys. The Retry packet type may be used by the server to force the client to validate its address [5].

Table 3: Table of QUIC long header types

Packet Type	QUIC v1	QUIC v2
Initial	0x00	0b01
0-RTT	0x01	0b10
Handshake	0x02	0b11
Retry	0x03	0b00

In QUIC there is only one short-header packet type, known as the 1-RTT packet. It is used for the bulk of communication, taking advantage of the more compact short header to reduce overhead on the connection, which is useful for high-throughput applications. The header is in part protected by header protection, preventing interference by middleboxes on the wire.

QUIC uses the concept of frames to enable protocol functionality. The payload of QUIC packets consists of one or more frames, and the frames can be of different

types, enabling the transmission of application data and control data in the same packet. Frames allow the QUIC protocol to signal transport events between the peers. Arguably the most important frame type is the data transmission frame, known as the STREAM frame. The stream frame carries stream data associated with a specific stream. The STREAM frame contains a stream identifier, byte offset and flags to track the state of the stream. The STREAM frame implicitly creates a new stream in case the identifier is previously unknown. The ACK frame is then used to acknowledge received packets. The ACK frame contains one or more ranges of packet numbers that have been received, with possible gap values indicating missing packets. This higher granularity enables the sender to only retransmit the missing packets, lowering retransmission overhead in reordering scenarios. Other important frame types include MAX_DATA and MAX_STREAM_DATA for flow control on the connection and stream level, PING frames to probe the peer and ensure that the connection stays alive, CRYPTO frames for carrying TLS handshake data and CONNECTION_CLOSE frames to terminate the connection. These are the most important frame types, with an extensive list being available in RFC 9000 chapter 19 Frame Types and Formats. Together, these frames allow for great control of transport functionality, allowing for explicit connection management [5].

3.5 Streams

The QUIC protocol is designed around the concept of streams. Streams are an abstraction of a byte-streams, that are used to transmit data between applications. Streams may be either bidirectional or unidirectional, that is, from the perspective of one peer, the streams may be read-only, write-only or read and write. Any single connection may contain one or more streams, with the streams being multiplexed and entirely independent from one another. Streams are designed to be lightweight, with minimal overhead. A stream can open, close and transmit data in a single frame, or alternatively the streams can persist for longer lengths of time, up to the lifetime of the connection. The number of streams and the amount of data that may be sent over a single stream is constrained by the flow control. The different streams are identified via a unique identifier inside a connection, the stream ID, which is a 62-bit integer. The reasoning for using a 62-bit integer is that the two remaining bits are used to distinguish between unidirectional and bidirectional streams, as well as identify the initiator of the stream, which may be either the client or server [5].

The lifetime of a stream is explicitly managed by the peers. The stream is first created by sending a stream frame, containing the stream ID of the stream that is requested to be opened. Data transfer is done via STREAM frames, that carry the application data that is to be transmitted. Once data transfer is complete, and the application is done with the stream, it may be closed by sending a stream frame with the FIN bit set or alternatively by sending a RESET_STREAM to abruptly end the stream, signaling that no further data will be sent. At the receiving end the receiver can read data from the stream and may end the stream by sending a STOP_SENDING frame, signaling that no more data will be accepted on the stream [5].

The design of QUIC streams is such that it directly addresses TCP's HOL blocking

issue. In TCP, the in-order delivery guarantee is implemented on an entire connection, resulting in the whole connection stalling in case of a lost packet until the packet is retransmitted and received. In comparison, QUIC building its streams on top of UDP, has implemented an in-order guarantee per stream. That is, any lost packet only blocks the specific stream that packet belonged to, allowing the other streams that may exist on the same connection to keep transmitting while the erroneous stream retransmits. This feature also works in the favor for control data, as control frames are transmitted separately from the stream frames, they are not affected by loss on any of the data streams [5]. An example of this advantage in action is in the HTTP/3 protocol. Here each client request is mapped to its own bidirectional QUIC stream, and unidirectional streams are used for control and push streams. HTTP/3 implementations should support at least 100 concurrent streams taking advantage of the multiplexing introduced by QUIC [7]. When a client visits a web page, it may open tens of streams simultaneously, to fetch HTML, CSS, JavaScript, images and fonts. If, for example, a packet containing part of an image is lost, QUIC enables the HTML and CSS streams to continue delivering while the image stream recovers, improving web page render times in lossy environments.

The lightweight design of QUIC streams makes them suitable for short request-response communications in other domains as well. DNS over QUIC takes advantage of the stream semantics by creating a new QUIC stream for each request. Besides preventing HOL blocking, DNS over QUIC takes advantage of QUIC's stream semantics by allowing each DNS query to use a separate stream for communication. This allows the protocol to cheaply process each request-response, forgoing the overhead of setting up a connection, with its full handshake, instead creating a lightweight stream for the specific communication, while still guaranteeing the reliable delivery of requests and responses. The multiplexed streams also allow the server to process and respond to the queries out-of-order, allowing multiple, simultaneous, outstanding queries at any one time [55].

3.6   Flow and Congestion Control

To prevent fast senders or malicious actors from overwhelming the receive buffer, the QUIC protocol implements flow control on two levels, a per-stream flow limitation and a per-connection limitation. The receiver controls the amount of data that can be sent on any one QUIC stream at a time, as well as over the connection as a whole. The receiver also limits the number of concurrent streams that the sender may open, preventing the overwhelming of the receiver with a large number of unique streams. The limits are set during the handshake process, but they may be updated by sending MAX_STREAM_DATA and MAX_DATA frames to signal that the flow control limits have been increased. A receiver may not reduce the flow control limits during the connection. If the sender is faster than the receiver it may reach a state where it is unable to transmit data due to either one of the limits. In this case the sender should send a STREAM_DATA_BLOCK or DATA_BLOCK frame to signal to the receiver that there is outstanding application data waiting to be sent, but it is currently unable to send due to the flow control limits. Similarly, the limit on the number of

streams that may be opened is set during the connection initialization phase. The limit may be increased by sending a MAX_STREAMS frame, and similarly to the data flow control limits, the streams limits may only be increased, not decreased. The sender can signal to the receiver that it would like to open more streams by sending a STREAMS_BLOCKED frame [5]. This granular control allows the protocol implementations to adapt the flow control according to the environment, enabling improved performance especially in dynamic environments.

The QUIC protocol uses three different packet number spaces, separate for initial, handshake and application data spaces. This is due to the corresponding packets using different levels of encryption, ensuring that acknowledgements sent in one packet number space does not cause retransmissions of packets of a different encryption level. The packet numbers of QUIC are monotonically increasing, signaling the order in which the packets were sent. That is, when retransmission occurs the retransmitted packet uses the next available packet number instead of the same packet number. To reconstruct reordered packets QUIC instead uses a stream offset field inside the stream frames, to keep track of the correct order of data. Loss detection in the QUIC protocol is then based on these packet numbers. Loss can either be detected via acknowledgement-based detection, if a packet that is considered in-flight and is unacknowledged, but a packet sent after the potentially lost packet has been acknowledged, and enough time has passed since the first packet was sent. Alternatively, a Probe Timeout (PTO) causes one or two datagrams to be sent out, expecting a response to test the reachability of the peer [50].

The QUIC protocol provides a set of generic signals that are designed to support multiple different sender-side congestion control algorithms. A congestion algorithm similar to TCP NewReno [56] is outlined in RFC 9002. However, a sender is free to choose any one algorithm they feel suits their use case, such as CUBIC, given that they conform to the congestion control algorithms outlined in RFC 8085 [57]. One option is to use QUIC packets which contain only ACK frames, as these do not count towards the flow control limits, but the loss of these packets can be detected by the QUIC algorithm [50].

The algorithm outlined in RFC 9002 begins in slow start, setting the initial congestion window to 10 times the maximum datagram size. When in slow start the sender increases the congestion number by the number of acknowledged bytes, until packet loss is detected. This means that the congestion window can double every RTT as long as all inflight packets are acknowledged. When packet loss is detected, the sender enters recovery period. During the recovery period the congestion window is reduced to half its current size, and this reduction may be made instantly when entering the recovery period or gradually using some other mechanism. Any additional loss detected during this period does not affect the congestion window size. The recovery period ends once the congestion window has been reduced to the new size, and the sender has sent a packet and that packet has been acknowledged. From here the sender enters congestion avoidance. During the congestion avoidance period, the sender may at maximum increase the congestion window by one maximum datagram size for each complete congestion window that has been acknowledged. In case loss is detected the sender goes back into recovery mode. In case persistent congestion is detected, that is

the continuous loss of all sent packets, the sender reenters slow start mode, and the process starts over [50].

In the QUIC algorithm there may be a situation where reordering of packets causes the receiver to receive packets to which it does not have the necessary keys to decrypt, such as handshake or 0-RTT packets arriving before the initial packets. Loss of these packets may be ignored, unless an earlier packet from the same packet space has been acknowledged, in which case the loss must not be ignored. When a probe packet is sent it should not be blocked by the congestion controller, even if they might exceed the current congestion limits, and these packets should also be accounted for as being in-flight. The sender should pace the sending of its packets, as large bursts of packets may induce congestion or packet loss. A lack of application data or pacing on the part of the sender may cause the congestion window to be underutilized. In this case the congestion window should not be increased [50].

3.7 Connections

The QUIC protocol is a stateful, connection-oriented protocol where the QUIC connection functions as the shared state between peers. Connections are created via a handshake, during which both the transport and security handshake is performed, with the security handshake being based in TLS 1.3 [6]. The handshake establishes the connection, and the parameters for the connection are declared. The parameters are individually decided by the peers, and the opposite peer needs to comply with the conditions set. It is possible via 0-RTT data transfer to transfer application data as part of the initial message, before the server has responded, and the server may already start sending data before the final handshake message from the client. This allows the protocol to exchange some security guarantees for improved latency when establishing a connection [5].

3.7.1 Connections and Identifiers

From a connection standpoint one of the innovations made by the QUIC protocol, as compared to TCP, which uses port-IP tuples to identify connections, is the Introduction of connection IDs. The point of connection IDs is to make the protocol more resilient to changes in the lower-level protocols, such as IP addresses changing when a client moves from one network to another. In case of connection migration from one network to another the connection ID used is also changed at the same time. This is to prevent passive listeners from tracking peers from one network to another, which could be done via correlating connection IDs [5].

In a QUIC connection each peer decides the connection ID for the other peer. Initially the connection IDs are issued during the handshake phase, and the connection ID is associated with a sequence number, starting from 0. A peer may use NEW_CONNECTION_ID frames to assign a new connection ID to its peer, at the same time increasing the sequence number by one. Both peers should issue multiple connection IDs to each other and should accept packets originating from any of these connection IDs. This gives both peers a sufficient number of unused connection

IDs that are available for later use during the connection. Packets from the assigned connection IDs should be accepted until they are explicitly retired, either using a RETIRE_CONNECTION_ID frame or by sending a NEW_CONNECTION_ID frame with the Retire Prior To value increased, which forces the peer to retire the related connection IDs. Sending a RETIRE_CONNECTION_ID frame implicitly requests the peer to issue a new connection ID to replace the one that has just been removed. A connection ID must not be retired without alerting the peer, and the number of connection IDs that have been locally retired, that is the connection ID is retired and a RETIRE_CONNECTION_ID frame has been sent, but no acknowledgement has yet been received, should be kept to a limited number [5].

When a QUIC packet is received, it is attempted to match this packet to an existing connection. If the recipient is a server, it is also possible that the incoming packet is part of a new connection. If it is not possible to associate the packet with any connection ID of any existing connection, and the packet is not part of initializing a new connection, then a Stateless Reset may be sent in response. If a packet is received, but it is not consistent with the connection, the packets are discarded. Such a scenario could be incorrect versioning or inability to decrypt the packet due to various reasons [5].

3.7.2   Connection Lifetime

On of the main innovations made by the QUIC protocol is the integration of the TLS 1.3 cryptographic handshake into the transport handshake, improving the latency of connection set up and encrypting the connection from the beginning. As can be seen from Figure 7, in a 1-RTT handshake the connection is initiated by the client sending an Initial packet, which contains a TLS ClientHello message contained inside a CRYPTO frame. The server responds with an Initial Packet containing the TLS ServerHello message, and a Handshake Packet containing EncryptedExtensions, as well as its certificate and possibly the CeritifcateVerify message in case the client should be authenticated as well. The client then responds with a Handshake message containing the TLS Finished message, which signals the completion of the handshake, and application data can now flow securely [5, 6].

It is possible to transmit data before the cryptographic handshake has been completed, albeit with some drawbacks. In Figure 7 the opportunities for early data transmission have been marked with an asterisk. Firstly, 0-RTT data transfer is possible, in case the client is previously known to the server and has received a TLS resumption ticket. With 0-RTT data transfer the initial data may be sent during the Initial Packet from the client. 0-RTT data transfer comes with the caveat that it is susceptible to replay attacks. It is also possible for the server to derive the encryption keys early and transmit data alongside its initial handshake message. This mechanic is referred to as "0.5-RTT", but the data is carried inside a 1-RTT packet. This has the drawback of transmitting data before the client has authenticated, which means that the server should avoid transmitting sensitive data during this transfer. Similarly, once the client has completed the cryptographic handshake it may already start transmitting data alongside the TLS Finished message [5, 6].
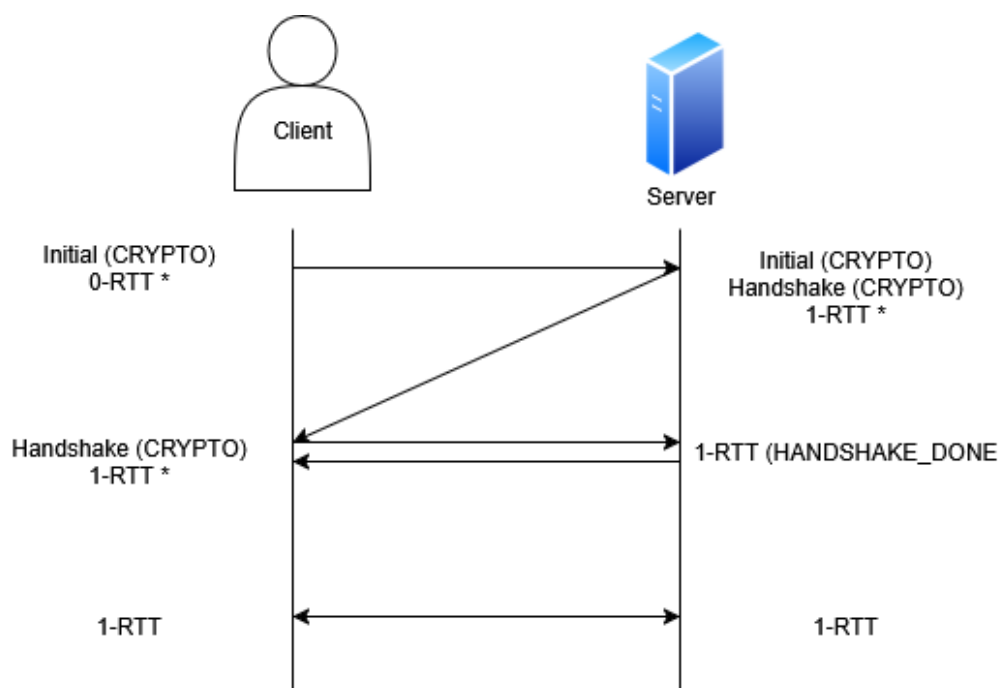
Figure 7: The QUIC Handshake

There are three ways in which a QUIC connection may be terminated. First via an idle timeout. The idle timeout is configured in the transport parameters, via the max_idle_timeout. If both endpoints announce a max_idle_timeout, the lower of the two will be used. The idle timeout is reset when receiving a packet from the peer or alternatively sending a packet to the peer. This means that an endpoint may defer the idle timeout, by sending a packet to the peer if it expects that data is still outstanding. If the idle timeout runs out, the connection will be silently closed, and the state discarded. Second, an endpoint may explicitly shut down a connection by sending a CONNECTION_CLOSE frame. This also immediately closes all open streams, putting the peer that sent the CONNECTION_CLOSE frame into the closing state, and the peer that received the CONNECTION_CLOSE into the draining state. This is to enable the peers to cleanly shut down their connection and properly handle packets that arrive out of order. Finally, a connection may be shut down via a stateless reset. This is used in case the peer does not have access to the state of the connection, and as a result is unable to process incoming packets. This causes a Stateless Reset Token to be sent, which is associated with the connection ID, causing the peer to immediately reset the connection.

3.7.3 Connection Migration

The QUIC protocol adds support for connection migration. In legacy protocols such as TCP, the connection is identified by the tuple pair of source and destination IP addresses and ports. If something happens that causes any of this information to change, for example changing from a local network to a mobile network, the connection will be

closed and the state discarded. As discussed in Section 3.7.1 one of the motivations for adding connection IDs was to improve the resilience of connections, allowing the connections to survive changes in network. The connection IDs are selected by the endpoints, and allow the packets to be correctly routed, even if the IP address or port of either endpoint changes for any reason [5].

As changes in network port or address may be involuntary, which can happen as a result from NAT rebinding or a network change, there may be a situation where either endpoint detects a change to its peer's address or port. The endpoint must then perform path validation. Path validation is done via sending a PATH_CHALLENGE frame, containing a random or hard to guess payload. The peer receives the PATH_CHALLENGE, and responds with a PATH_RESPONSE frame, echoing the payload from the PATH_CHALLENGE frame. In this way the path is validated one way. The peer may include its own PATH_CHALLENGE frame alongside the PATH_RESPONCE, to validate reachability the other way [5].

Connection migration may also be initialized explicitly by sending packets from a new local address. In this case both peers must ensure reachability on the new path, by using path validation as described earlier in this section. When an endpoint receives packets from a new address, it must validate the path and subsequently send packets to the new address, using a new connection ID, to prevent on-wire tracking of the connection migration. In practice, on the receiving end the process is the same for explicit and implicit migration. A server will not initialize a connection migration, this is exclusively done by the client. What a server may do is use a preferred address. This allows servers to accept incoming connections on one address, and then quickly transfer the connection to a separate, preferred address, performing path validation against the client from the new address [5].

4 The Server Message Block Protocol

The main purpose of the SMB protocol is to share files and directories over a network. The SMB protocol is a stateful protocol, where clients initiate connections, an authenticated session is created and requests are sent over the connection, allowing the client to perform file operations. In addition to sharing files, the SMB protocol enables access to other network resources, such as printers. There currently exist two major versions of the SMB protocol, the original SMB protocol, sometimes referred to as the CIFS protocols, which later evolved into the SMB 2 Protocol, which encapsulates SMB Versions 2 and 3 [58]. The SMB 2 protocol will from this point forwards be referred to as the SMB protocol. The legacy SMB protocol will be referenced as SMB 1 for clarity, however it will not be covered in depth by this thesis. This section of the thesis will give a technical overview of the SMB protocol.

Within the larger versions of the SMB protocol there exists minor versions, referred to as dialects. Within the SMB protocol the following dialects exist, SMB 2, SMB 2.1, SMB 3.0, SMB 3.0.2 and SMB 3.1.1. The basic functionality of the SMB protocol was already defined in SMB 1, allowing clients to connect to shares and perform file operations. As can be seen in the overview given by Table 4, the protocol has been continuously improved by then, adding support for, among other features, encryption and alternative transports [58].

Table 4: Comparison of SMB protocol dialects

Dialect	Introduced in	Key Features
SMB 1.0	2000	Basic file and printer sharing
SMB 2.0	2006	Reduced chattiness, new packet format, support of symbolic links
SMB 2.1	2009	Opportunistic locks, minor performance enhancements
SMB 3.0	2012	Encryption of traffic, multi-channel support, SMB Direct (RDMA), transparent failover, Directory Leasing
SMB 3.0.2	2013	Unbuffered read and write
SMB 3.1.1	2016	Support QUIC as a transport, RDMA encryption, Improved cryotography support

The SMB protocol places itself in the application layer and it does not stand on its own but relies on other protocols for much of its functionality. To enable authentication of clients, the SMB protocol uses the Simple and Protected GSS-API Negotiation (SPNEGO), as defined in RFC 4178 [59]. The SPNEGO protocol exposes a common interface that may be used for authentication with the help of other authentication protocols. In the case of the SMB protocol the SPNEGO protocols rely on Kerberos

Authentication, as defined in RFC 4120 [60], or the New Technology LAN Manager (NTLM) protocol, as defined in [MS-NLMP] [61] for authentication purposes.

The SMB protocol is designed to be run on top of a reliable transport. Historically this was NetBIOS over TCP, later standardized to standard TCP. As outlined earlier in this section, later versions of the SMB protocol support alternative transports. The SMB dialects >= 3.0 supports use of RDMA via SMB direct, for improved performance. The latest version of the SMB protocol, 3.1.1 adds support for the QUIC protocol, which comes with increased security guarantees and potentially improved performance over TCP [58].

4.1   SMB message structure

The SMB protocol uses the Direct TCP packet header, as shown in Figure 8. It begins with a zero byte, followed by a 3-byte StreamProtocolLength field which indicates the length of the SMB2 message. Following this is the actual SMB2 message. The first part of the SMB2 Message contains the SMB2 packet header. There are 2 versions of the header, an asynchronous and synchronous header, for asynchronous and synchronous requests respectively. The first 32 bytes of the SMB2 packet header is common for both request types, as may be seen from Figure 9. The SMB2 Transport Header begins with the 4-byte Protocol Id field, which is always set to 0x424D53FE, or when converted to ASCII and in network byte order, 0xFE, S, M, B. The following field is the StructureSize, which is set to 64, which is the total size of the header. Next is the CreditCharge field, this indicates how many credits the request consumes, more on the SMB credit system later in this chapter. Following this is a field that is interpreted differently for the client and the server and depending on dialect. On the client side, if the dialect is at least SMB 3.0, this field indicates Channel, for dialects < 3.0, this field is reserved. From the Server side this field indicates the status in response to any request. The 2-byte command field is used to indicate the command of the request or response, Table 5 contains a list of all possible SMB2 commands. Following this is the 2-byte credit request and response field, which contains information about requested credits. The flags field contains flags for determining how the packet should be processed. In the SMB protocol it is possible to concatenate multiple requests into one packet. This is then indicated via the NextCommand field, that indicates the offset to the next request. Finally, the MessageId field is the unique identifier that identifies the specific request and its corresponding response. In addition, the asynchronous header contains an AsyncId for identifying the asynchronous operation, a SessionId that identifies the session, and in case the message is signed, the signature. The synchronous header likewise contains the signature and MessageId, but instead of the AsyncId the synchronous header contains the TreeId. Following the SMB2 packet header is the SMB2 Protocol Message, which has a variable length depending on the command code and if it is a request or a response [58].

Figure 8: The SMB Direct TCP header

Figure 9: The SMB protocol header

4.2 SMB connection lifetime

The SMB connection begins by establishing a connection via the transport protocol of choice. As discussed in Section 4, this may be TCP, RDMA or QUIC. Once the connection is established, the first message sent from the client is the SMB2 NEGOTIATE message. The client sends an array of supported dialects, and in

Table 5: SMB Commands

Name	Value
SMB2 NEGOTIATE	0x0000
SMB2 SESSION_SETUP	0x0001
SMB2 LOGOFF	0x0002
SMB2 TREE_CONNECT	0x0003
SMB2 TREE_DISCONNECT	0x0004
SMB2 CREATE	0x0005
SMB2 CLOSE	0x0006
SMB2 FLUSH	0x0007
SMB2 READ	0x0008
SMB2 WRITE	0x0009
SMB2 LOCK	0x000A
SMB2 IOCTL	0x000B
SMB2 CANCEL	0x000C
SMB2 ECHO	0x000D
SMB2 QUERY_DIRECTORY	0x000E
SMB2 CHANGE_NOTIFY	0x000F
SMB2 QUERY_INFO	0x0010
SMB2 SET_INFO	0x0011
SMB2 OPLOCK_BREAK	0x0012
SMB2 SERVER_TO_CLIENT_NOTIFICATION (ASYNC only)	0x0013

response the server selects the greatest common dialect that is supported by both parties. In the SMB2 NEGOTIATE response the server also indicates its capabilities, such as transport encryption in the case of QUIC transport. Once the dialect and capabilities have been negotiated, the next step in the process is authenticating the user. This is done via the SMB2 SESSION_SETUP message. As discussed in Section 4, this is done via the SPNEGO protocol, and the underlying protocol used for authentication is either NTLM or Kerberos. For Kerberos authentication the client authenticates against a domain controller, which issues a ticket to the client that can be used for authentication. In case Kerberos authentication is not available, the client may authenticate via NTLM. This works by the server sending a challenge to the client, and using the client password as a shared secret, the response to the challenge is calculated, and sent back to the server, which verifies the response and authenticates the client [58].

Once the SMB connection is established and the client is authenticated, it may now connect to a share exported by the server. This is done via the SMB2 TREE_CONNECT request. The client requests to connect to a specific share, identified with the path name in the form \\server\share. The server tries to find the requested share, and if successful and the client has appropriate access rights, allocates a specific tree connect object to store information about the session. This object is identified via a TreeId that is sent back to the client, alongside the share type: disk, pipe or printer. In addition, the

server returns information about the share capabilities, such as continuous availability or scaleout capabilities [58].

Having connected to the share the client is now able to perform the main part of the transaction, file operations. The client can use SMB2 CREATE to create or open files, SMB2 CLOSE to close or delete files, SMB2 WRITE and READ to write and read to and from a file correspondingly, or SMB2 QUERY_DIRECTORY to list the available files inside a directory. Once a file is opened it is identified by an opaque file handle, called an open, returned by the server. Subsequent file operations are then done via this handle, and the server and client store the associated information inside a table. As the SMB protocol is stateful, all the requests are associated with a certain session, and the requests are handled in accordance with the client's capabilities and authorization. To limit the number of outstanding requests any client has at any one time, the SMB protocol uses a credit system. Every client has a certain number of credits that are consumed upon sending any request. The number of credits is equal to the size of the request, or the expected size of the response, whichever is larger, divided by 65536 plus 1. It is up to the client to request credits as they are consumed, and the server should grant credits when requested, to avoid a situation where the client reaches 0 credits, as without any credits the client cannot request any more credits and is then blocked from performing any more operations [58].

When the client is finished with a share it is time to disconnect and log off. The disconnection is performed via a SMB2 TREE_DISCONNECT message, using the TreeId provided earlier when connection was set up to the share. If the client is done with its operations, it can log off from the server via the SMB2 LOGOFF message, allowing the client to cut all connections to the server. This section only covers the most basic process and possibilities of using the SMB protocol to access a shared resource. For further reference the Microsoft Technical documentation [58] outlines all possible scenarios and is available for further reading into the technical specifics of the SMB protocol messages and functionalities.

4.3   SMB over alternative transports

The SMB protocol currently supports 4 transports, TCP, RDMA, QUIC and the deprecated TCP over NetBIOS. The standard transport in most scenarios is TCP, but the protocol is moving towards the use of alternative transports to improve performance and security. The SMB protocol supports RDMA via SMB Direct. The goal of RDMA is to allow remote direct memory access, that is reading from and writing to a remote memory buffer, without having to copy the data. This copy-less operation saves CPU cycles, and in that way requires less processing power and improves throughput by using specific network adapters with support for RDMA [62]. The goal of SMB Direct is to improve performance by utilizing RDMA and multichannel. SMB multichannel allows the client to open multiple transport connections to the server, improving performance by transferring data in parallel. Encryption over RDMA is supported since SMB 3.1.1 by encrypting the payloads before they are sent over the wire [63].

From the perspective of this thesis, the most relevant alternative transport is SMB over QUIC. As discussed in Section 3, the QUIC protocol brings many advantages

over legacy TCP transports. QUIC enables transport-level encryption that leverages TLS 1.3 for more secure cryptography, multiple logical streams to combat HOL blocking, moving the traffic to port 443 to avoid the common port blocking of SMB port 445 and supports connection migration. The most notable of the improvements to the SMB protocol is the inclusion of transport level encryption. While SMB version >= 3 supports per share-based encryption, the cryptographic analysis is outside the scope of this thesis, but benefits of TLS 1.3, such as perfect forward secrecy is not part of standard SMB encryption [64]. SMB over QUIC supports negotiating transport level encryption in the transport capabilities of the negotiate protocol request. This allows the client and server to skip SMB level encryption, even though it is enabled on a share, allowing the communication to rely entirely on QUIC for securing communications [58]. Additionally, SMB over QUIC supports what is known as client access control which forces the client to provide its own authentication via a certificate. As outlined in earlier sections, in TLS 1.3, during the handshake, there is the possibility for the server to request that the client provides its own certificate for authentication purposes. The server can then use that certificate to determine if the client should be able to connect to the share [65].

5 Implementing QUIC as transport for SMB server

The QUIC protocol, as outlined in Chapter 3, currently has no kernel level implementation in Linux, instead relying on user space libraries for support. This was a conscious design decision for the protocol, allowing for different implementations and easier iterations as well as switching between different stacks. There exist many different implementations of QUIC, such as *lsquic* developed by LiteSpeed, *quiche* developed by Cloudflare and *MsQuic* developed by Microsoft [66]. There is also an active effort to develop a version of QUIC for the Linux kernel, driven by Xin Long [67]. For the purposes of this thesis the MsQuic library was chosen as the library that is used to implement the QUIC transport layer. The reasoning behind this choice was twofold. Firstly, the MsQuic library shows promising performance numbers when compared to other libraries [68]. Secondly, as the only SMB over QUIC client currently available is the Microsoft client, it is advantageous to use the same QUIC implementation both in the server and the client, to ensure maximum compatibility. This section of the thesis will outline the architecture and API of the MsQuic library, as well as describe the design of a QUIC transport layer using the MsQuic library. Additionally the process of integrating the QUIC transport layer into the Fusion SMB server will be presented.

5.1 MsQuic architecture and API

The MsQuic library is a cross-platform implementation of the QUIC transport protocol, implemented as a shared library written in C, with support for development in C++, Rust and C#. It is open source and licensed under the MIT-license. The library utilizes the Windows Secure Channel (Schannel) suite for cryptography functionality in Windows, and for Linux based OSs the library uses a custom fork of the OpenSSL library to provide support for QUIC specific cryptography [12]. The library uses an asynchronous processing model, where the caller creates callback handlers and registers the handlers with different MsQuic objects, such as listeners and connections. These callback handlers are then invoked to handle events that occur, such as receiving a new connection request, a stream being opened or closed, or receiving data on a stream [69].

5.1.1 High level architecture

The MsQuic library abstracts away the concept of networking sockets, instead opting for an object-based model, which can be seen in Figure 10. The MsQuic library model includes 5 main types of objects: the Registration, Configuration, Listener, Connection and Stream object. The hierarchy of these objects can be seen in Figure 10. All function calls are done in relation to one of these objects with the exception of the call to initialize the library, which returns a function table of MsQuic functions, and the corresponding call to close the library [69].

The Registration object represents the execution context for the transport, being responsible for the connection logic and creating the threads that are used for processing. All other objects are created under the registration, and each registration is completely
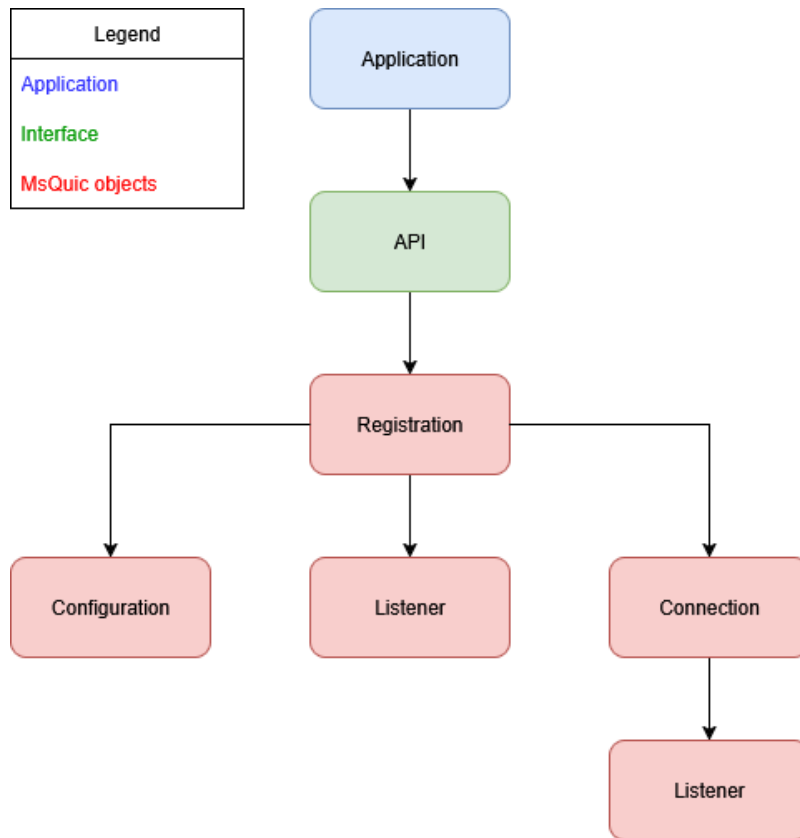
Figure 10: High Level MsQuic Object Model

independent from one another, with resources and scope being unique to each registration. In practice this means that one application generally uses only one registration [69].

The configuration object stores the settings that are used in configuring the MsQuic library. These include the QUIC transport settings, as well as security configurations. Everything from initial receive windows, idle timeout timers to the congestion algorithm used can be configured. The comprehensive list of configuration parameters can be found in the MsQuic documentation. In addition, each of the MsQuic objects have specific parameters that can be configured, and the certificate as well as its parameters is separately configured, for example configuring requesting a certificate from the client and allowed cipher suites. The configuration object also stores a list of Application Layer Protocol Negotiation (ALPN) IDs that the server supports [69]. ALPN is a TLS feature, that is designed for a situation where it is not clear which service a client is trying to access. This may arise when a server is running multiple network services that all use the same port, for example the SMB over QUIC protocol uses the same UDP port as HTTP/3. ALPN allows the server and client to negotiate which protocol will be used inside the TLS tunnel, removing the need for an additional round trip to negotiate the used protocol separately [70]. For reference SMB over QUIC simply uses the ALPN ID "smb".

The listener object is responsible for listening for and accepting incoming con-

nections. The listener object uses a callback function that is responsible for handling incoming connections. The callback function decides if the incoming connections is accepted, and if it is a connection object is created in response [69].

The Connection object represents the QUIC connection between two peers. On the client side this connection is explicitly created via a function call, while on the server side it is created in the listener callback. The connection object contains information about the connection, and it is expected to handle events related to connection set up, stream creation and connection teardown, either as a result of an error or from one of the peers closing the connection [69].

The most important object type is the Stream object, as it is responsible for the data transfer between the peers. As discussed in Section 3, there can be a theoretical unlimited number of streams on top of a connection, but in practice the maximum number is configured in the MsQuic library, with separate limits for unidirectional and bidirectional streams. New streams can be created either by the client or the server, with the other party being notified of the event. When sending data the MsQuic library takes temporary ownership of the buffer, copying the data internally and queuing the data to be sent. This allows the MsQuic library to pace the sending and "keep the pipe full", aiming to not let the stream sit unnecessarily idle. It is also possible to disable this internal buffering, instead making the caller responsible for queuing enough sends to prevent idling the connection [69].

Receiving data happens via a receive event, that gets passed to the callback function. This includes references to buffers that contain the received data. The data can then either be processed inside the callback or passed along to a separate thread to be processed. Either way, the caller needs to signal to the library when the data has been processed, giving back ownership of the buffers to the library. This also indicates that the caller is ready to receive more data. Normally only one receive event can be active at one time, i.e. the library will not indicate that more data is available before the application has signaled that the previous set of data has been processed. Alternatively, the MsQuic library supports Multi-receive mode, where the library will keep creating receive events when data is available, and it is up to the caller to keep track of the number of bytes processed, and report this back to the library. This enables more efficient processing of data when the processing is done asynchronously in a separate thread [69].

5.2 Fusion SMB server QUIC transport layer design

As discussed in earlier section, network communication using TCP experiences some drawbacks related to the fundamental design of the protocol, drawbacks which the QUIC protocol aims to solve. This section describes the design of a QUIC transport layer for the Fusion SMB server, utilizing the MsQuic library for implementing QUIC functionality. The decision to use the MsQuic library was discussed earlier in Section 5. The main goal of this project is to create a QUIC transport layer that supports all the necessary QUIC features such that it is able to communicate with and establish a connection with the Microsoft SMB over QUIC client. A secondary goal is to design the QUIC transport layer in such a way that the performance, measured via throughput

during various workloads, is comparable to the standard SMB stack using TCP on an encrypted share. This allows the QUIC transport layer to work as an improvement to the standard TCP transport layer, with improved security via TLS 1.3, resilience to SMB port blocking from ISPs and connection migration support, relevant especially for mobile users.

5.2.1 MsQuic Integration into Fusion SMB

The generic transport layer interface of the Fusion SMB server uses a set of functions that is called by the server to interact with the transport, such as for creating and destroying the transport, listening for incoming connection and receiving and transmitting data. This requires a little bit of work around when integrating with the MsQuic library, as it does not conform to standard BSD-style socket semantics. Figure 11 gives a high-level overview of the interface between the QUIC transport layer and the rest of the server. The processing begins by creating the specific transport, which is configured inside the Fusion SMB configuration. This causes the server to create one of two new types of transport, IPV4_QUIC or IPV6_QUIC. During the creation the MsQuic library is initialized, and the specific configuration options are read in and stored, such as listening interface and TLS credentials, which are needed later. A listener object is created and starts listening for incoming connections. When a connection arrives via a QUIC_LISTENER_EVENT_NEW_CONNECTION event, the listener accepts the connection and passes it along to the connection callback handler. The connection callback continues the handshake by using the TLS credentials stored earlier and creates an internal connection object storing the connection information that is necessary for the protocol. This internal connection object is passed back to the Fusion SMB server, that stores the connection internally as an active connection. The connection is now set up, and the connection object continues waiting until a QUIC_CONNECTION_EVENT_PEER_STREAM_STARTED event arrives, indicating that a new stream, with associated data has been opened. This causes a stream object to be created, with its associated callback function set. The stream object works as a middle layer between the Fusion SMB server and the QUIC transport, receiving data into a receive buffer for the server, and transmitting data from the server to the client, enabling two-way communication between the server and client.

5.2.2 QUIC transport layer data path

When implementing the QUIC layer, the multi-receive mode of the MsQuic library, as outlined in Section 5.1, will be used due to the improved performance this offers the application. In practice this means that as soon as data is available, the stream callback is invoked with a QUIC_STREAM_EVENT_RECEIVE event, containing the received data. This data should then be processed in a separate thread, that is not in the callback handler inside the MsQuic worker thread. This is accomplished by passing along the data to the Fusion SMB server's internal threads. To enable this, when data is received, the reference to the received buffers is copied into a separate
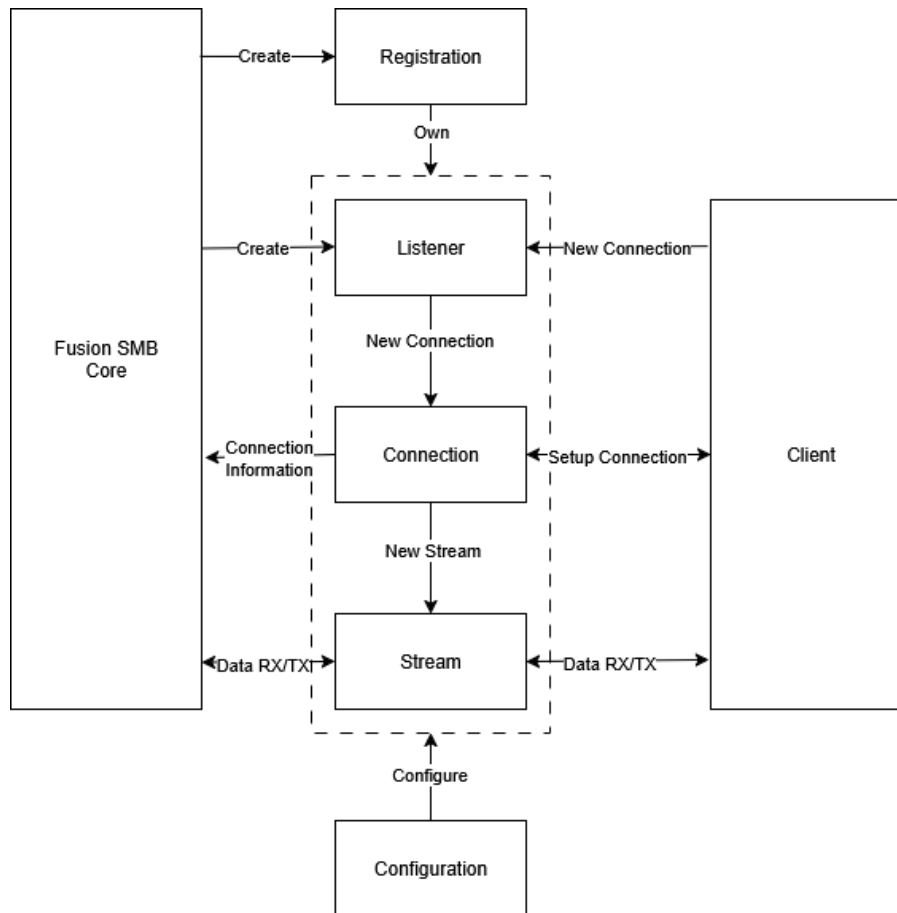
Figure 11: MsQuic integration with Fusion SMB

data structure, as seen in Figure 12, consisting of a linked list of buffers. This choice was made as it cannot be guaranteed that the servers is able to drain the incoming data faster than it arrives, and the linked list structure allows the total buffer size to grow dynamically as needed. This method of handling incoming network buffers is similar to the method used in the Linux kernel, albeit a simplified version, using a single-linked list instead as compared to the doubly linked list used in the Linux kernel [71]. As can be seen in Figure 13 the data is not copied until it reaches the internal receiving thread, with only the reference to the received buffer being passed around. This allows the processing in the callback function to be inexpensive, quickly freeing up the worker thread for additional processing. This solution of storing the received buffer temporarily in a linked list simplifies integration with the existing receive interface, as the receiving threads expect BSD-style socket semantics, that is issuing read calls to read in data, instead of the event driven model that is used by the MsQuic library. The linked list then works as a middle layer, allowing the receive interface to work with the QUIC transport layer without modification. When draining the receive buffer the caller begins from the head, reading in as much data as necessary, updating the offset, and if the head is completely drained, moves onto the next element in the list, until the requested amount of data has been drained.
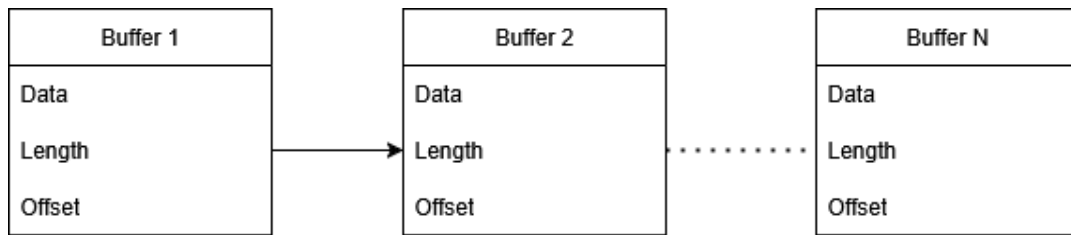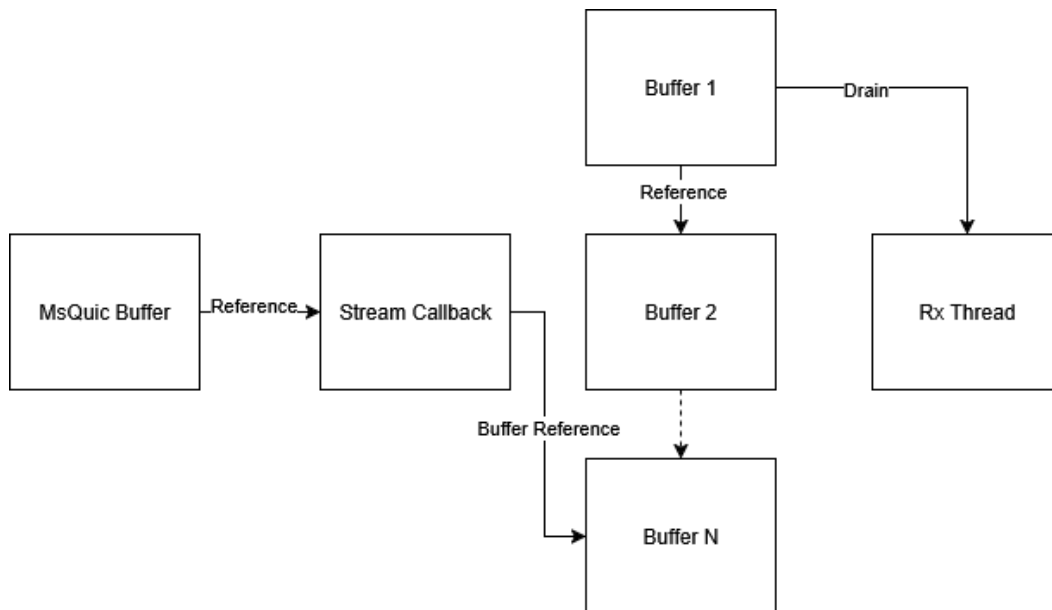
Figure 12: QUIC transport layer receive buffer

Figure 13: Data flow from MsQuic buffer to RX thread

The steps of receiving data and getting it into the Fusion SMB core can be seen in Figure 14. When data has arrived, and the buffer information has been copied into the receive buffer list, the next step is checking if the buffer is currently actively being drained. If it is, the callback signals to the draining thread that there is more data available to be read. Otherwise, the callback sends an internal IPC message to the receiving threads, signaling that there is data available on this connection and for a receiver thread to start processing. From the callback function STATUS_PENDING is returned to the MsQuic library, signaling that the data is being processed in a separate thread. Once the receiving thread has finished draining the buffers, it calls StreamReceiveComplete() with the number of bytes drained, signaling to the MsQuic library that the processing of the received data is at least partially complete.

The transmission of data is less involved than receiving data, as can be seen in Figure 15. When data is to be transmitted it is stored in an array of buffers in the iovec format[1]. When transmitting the data the MsQuic library expects the buffers to be in its own internal QUIC_BUFFER format, so the iovecs could simply be cast to the QUIC_BUFFER struct. However, as the MsQuic call to transmit data,

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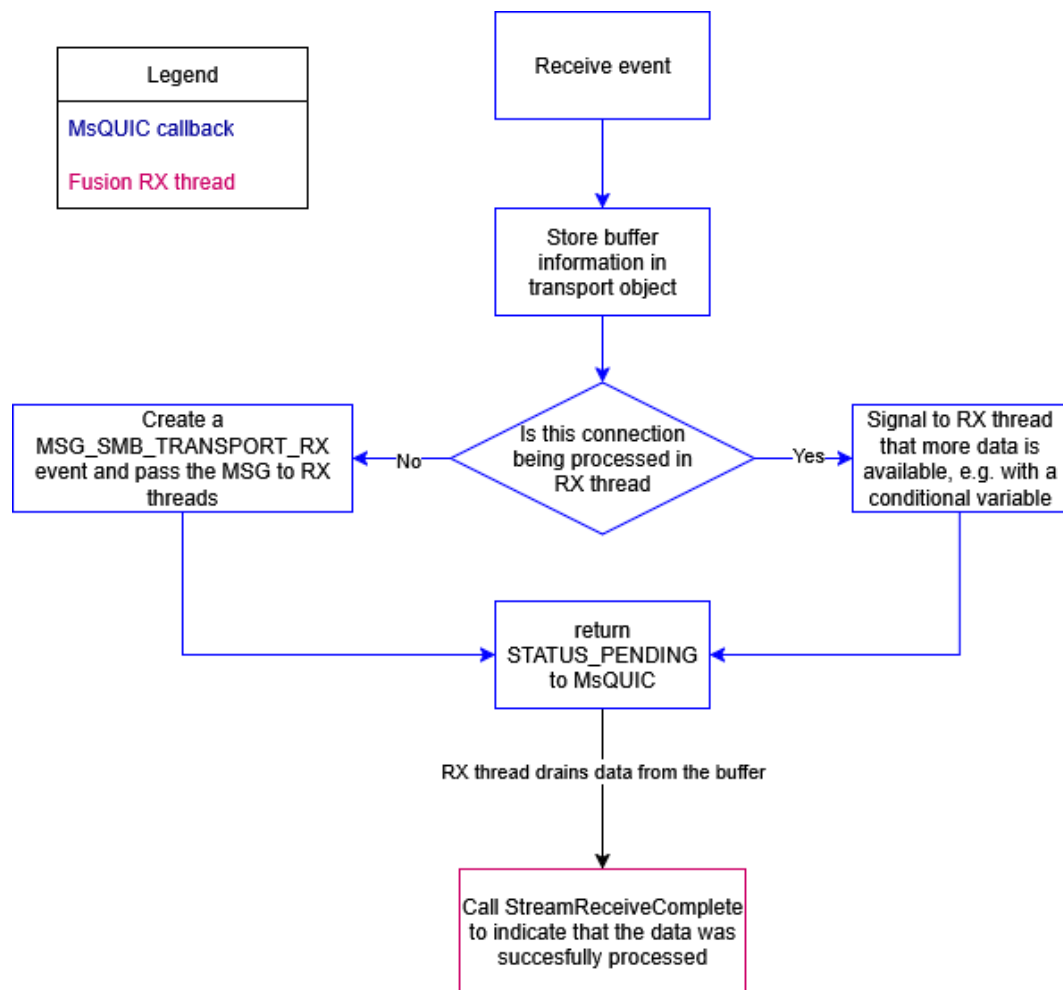[1] https://man7.org/linux/man-pages/man3/iovec.3type.html

Figure 14: QUIC transport layer receive flow

StreamSend(), is non-blocking, it is not guaranteed that by the time the buffers are freed in the transmitting thread, that the MsQuic library has had time to transmit them yet. So, to simplify lifetime tracking of the buffers we instead copy them into new QUIC_BUFFERS, adding one additional copy. The original buffers are freed as soon as the StreamSend() call returns, while the new buffers get freed in the callback handler once transmission is complete, as signaled by a STREAM_SEND_COMPLETE event in the MsQuic stream callback thread.

Testing against Microsoft's SMB over QUIC client shows that the client utilizes the QUIC transport in the same way that it would use a TCP transport, that is it only open a single stream per connection. This greatly simplifies processing of received and transmitted data, as there is no need to track an arbitrary number of streams per connection.
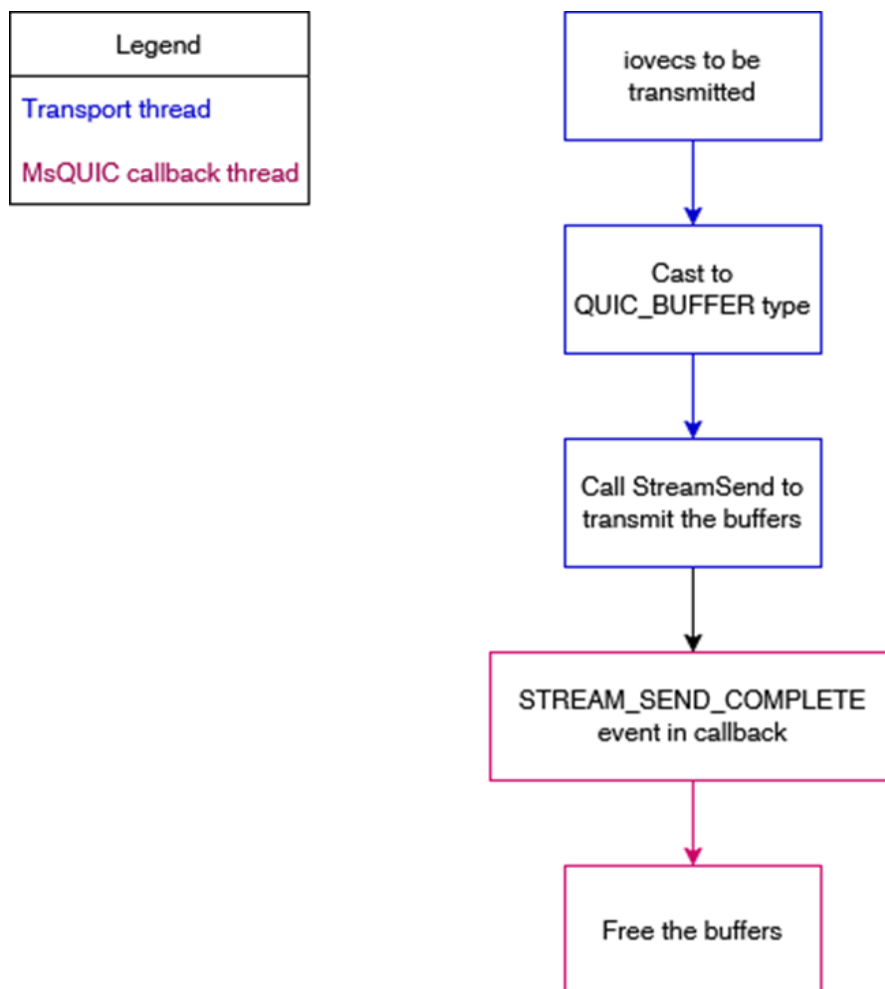
Figure 15: QUIC transport layer transmission flow

6 Benchmarking

6.1 Test environment

Hardware environment   As illustrated in Figure 16 the hardware environment for the test setup consists of a server and a client, connected directly via a 100 GbE switch. The server is a Supermicro SYS-2029BT-HNR, with dual Intel Xeon Gold 622R CPUs running at a clock speed of 2.9 GHz. The server has 192 GB of RAM, and a 3 TB Micron 9300 NVMe drive supporting read and write speeds of 3.5 GB. The client machine is a Gigabyte R281-3C1, with dual Intel Xeon Silver 4114 CPUs, running at a clock speed of 2.2GHz. Similarly to the server the client has 192 GB of RAM.
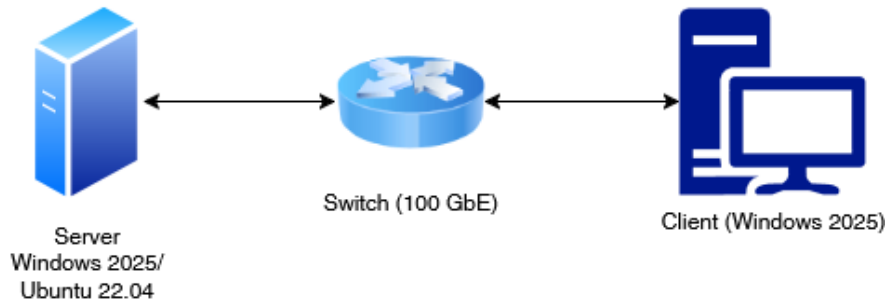
Figure 16: Test environment hardware setup

Software environment   The server is set up to dual boot Windows 2025 and Ubuntu 22.04 LTS, allowing for the benchmarks to be run against the same hardware for both the Fusion SMB and Windows SMB implementations. The client is running Windows 2025 to support SMB over QUIC client capabilities.

6.1.1 SMB over QUIC implementations analyzed

As part of this thesis two different SMB over QUIC implementations were considered for comparison in the benchmark.

Fusion SMB server: The Fusion SMB server, with the prototype QUIC transport layer as described in Chapter 5, will be the first item of comparison. The performance of the Fusion SMB server with the MsQuic library for QUIC transport capabilities will be compared to the Windows SMB over QUIC implementation, as well as the traditional TCP-based transport.

Windows SMB over QUIC server: The Windows SMB over QUIC server implementation provided by Microsoft is currently the only production-ready and widely deployed implementation of SMB over QUIC. The Windows server will be the main target of comparison against Fusion SMB, comparing the direct performance of QUIC as well as the performance impact of using QUIC in place of TCP for SMB traffic.

Windows SMB over QUIC client: The Windows SMB over QUIC client provided by Microsoft is, as with the server implementation, the only production-ready and widely deployed implementation of SMB over QUIC. The client will be used for testing both against Fusion SMB and Windows SMB server.

6.1.2   Benchmarking Software

For measuring the link speed and making sure the tests are running over the correct network interface, iPerf[2] will be used. iPerf allows for the testing of the maximum bandwidth over a network, and will indicate any possible issues or bottlenecks in the test environment. The MsQuic repository contains a tool for testing the performance of MsQuic called SecNetPerf[3], which implements a QUIC performance protocol defined in [72]. The tool enables the testing of link throughput, which means that the tool is used to get a baseline performance of MsQuic over the link used for testing. This value can then be compared to the throughput of the SMB over QUIC solution, to determine the overhead introduced by the implementation of the QUIC transport layer together with the SMB protocol.

To ensure reproducibility and comparability of the results, filesystem benchmarks are run using the Flexible I/O (fio) tester [4], which enables running synthetic workloads on the shares. The fio tester supports many different scenarios, workloads as well as fine-grained control over parameters such as file and block size, sequential and random reads as well as time-based workloads.

6.2   Test scenarios

6.2.1   Interoperability tests

The first step in testing, before performance evaluation, is to make sure that the two implementations outlined are able to connect, authenticate and set up the SMB session. Using the Windows SMB over QUIC client, it is ensured that the client is able to connect to the server, receive and accept the server certificate, and additionally perform some basic file operations to ensure that the data transfer is working properly. Additionally, the connection should remain stable during the lifetime of the connection, and teardown should be handled gracefully.

6.2.2   Benchmarking scenarios

The exact commands used in running the benchmarks are available in Appendix A. The tests were chosen in such a way as to both benchmark the solutions under optimal conditions, as well as represent real world scenarios. The FIO benchmarks were all run for a time of 120 seconds to ensure that the results would represent an average throughput under sustained load and minimize the impact of external conditions.

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[2]https://iperf.fr/
[3]https://github.com/microsoft/msquic/tree/main/src/perf
[4]https://fio.readthedocs.io/en/latest/fio_doc.html

iPerf: iPerf is run using the standard setup, with no additional parameters. This benchmark tests the maximum possible network throughput during a time period of 10 seconds.

SecNetPerf: For running the SecNetPerf performance test to get an overview of the link speed, two tests are run, one for checking the link speed each way. The test is run with the maximum throughput execution profile, and runs for 60 seconds, first testing uplink speed, then downlink speed (from the perspective of the client). This gives a base value for what the optimal throughput is expected to be, as logically the addition of the overhead of QUIC transport layer, SMB protocol and filesystem IO should not improve performance. This test also should work as a check that the environment is correctly set up, as unexpected results may point to problems in the configuration.

FIO Scenario 1: Sequential read and write of single large file   The sequential read and write of one file is used to establish baseline numbers for the performance of the transport. The test is designed such that it puts little strain on the filesystem IO, instead of having the transport layer as the primary bottleneck for performance. The tests are run against a pre-allocated 5GB file, allowing for a maximum throughput test during continuous load. This test uses a large block size to maximize the performance.

FIO Scenario 2: Random read and writes of a single large file   In comparison to scenario 1, where the maximum throughput is tested, scenario 2 with random reads and writes instead tests the performance when there is a lot of small operations. The main performance metric of scenario 2 is the number of I/O operations per second (IOPS). To simulate this workload, it is run against a 1 GB file, with a small blocksize together with random writes and reads. This corresponds to workloads where many small operations are done consistently, for example in spreadsheets.

FIO Scenario 3: Read and write of many small files   The final FIO scenario is the reading and writing of many small files concurrently, testing workloads where there is a lot of concurrency. The test creates 16 files of size 100 kB and performs random reads and writes against the files. This allows the benchmarking of a scenario with many processes simultaneously performing workloads, stress testing different parts of the processing than scenario 1 and 2.

6.3   Results

This section will outline the results from the benchmarks outlined in Section 6. For each of the scenarios there are two sets of graphs, one outlining the IO performance in MB/s, as well as the IOPS. As the benchmarks used consistent block sizes, it is expected that these two metrics are heavily correlated. Each of the scenarios were run against SMB over QUIC, as well as against SMB using TCP. The QUIC benchmarks were ran against an unencrypted share, instead relying on the QUIC protocol's integrated encryption. The TCP benchmarks were run against an encrypted SMB share, so that

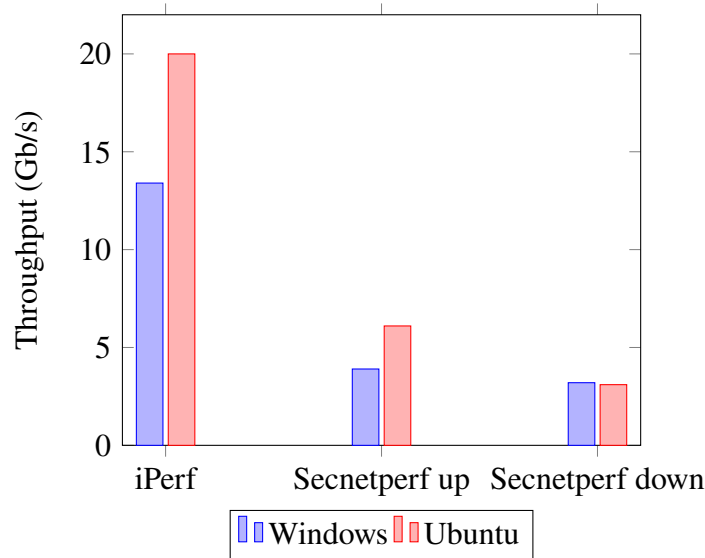the performance numbers would be comparable with the performance number from SMB over QUIC.

Figure 17: Link speed throughput

As can be seen in Figure 17, the link speed was verified using both iPerf and SecNetPerf. The performance tests were run using the Windows 2025 client machine as the performance test client, against both the Windows and Ubuntu server hosts. Using the default settings of iPerf, the throughput reached 20 Gb/s against the Ubuntu host, and 13.4 Gb/s against Windows. Similarly, using SecNetPerf, speeds of at least 3 Gb/s where reached both ways, for both servers. This indicates that under optimal conditions, a single QUIC connection and stream using MsQuic can at most reach speeds of 3Gb/s on the test hardware, indicating the point where the throughput is limited by the MsQuic library.

The interoperability of the SMB over QUIC implementation, as outlined in Chapter 5, was manually tested using the Windows 2025 SMB over QUIC client. This was accomplished by using PowerShell to mount the share. By specifying */transport:quic* when mounting the share, it is possible to force the client to exclusively try to connect via QUIC, completely forgoing TCP. A self-signed certificate was used in the server for testing purposes, and the client was configured to accept the certificate by specifying */skipcertcheck* when mounting the share. Via manual testing it was shown that the client could mount the share, create a text file, write some data to the file, delete the file and finally disconnect from the server, without any issues. This showed that the compatibility aspect of the implementation had been reached.

6.3.1 Scenario 1: Large sequential I/O

The results of benchmark scenario 1 are outlined in Figure 18. Looking at the write workloads it can be seen from the graphs that there is one significant outlier, the throughput of the write workload from the Windows Client to the Windows server using
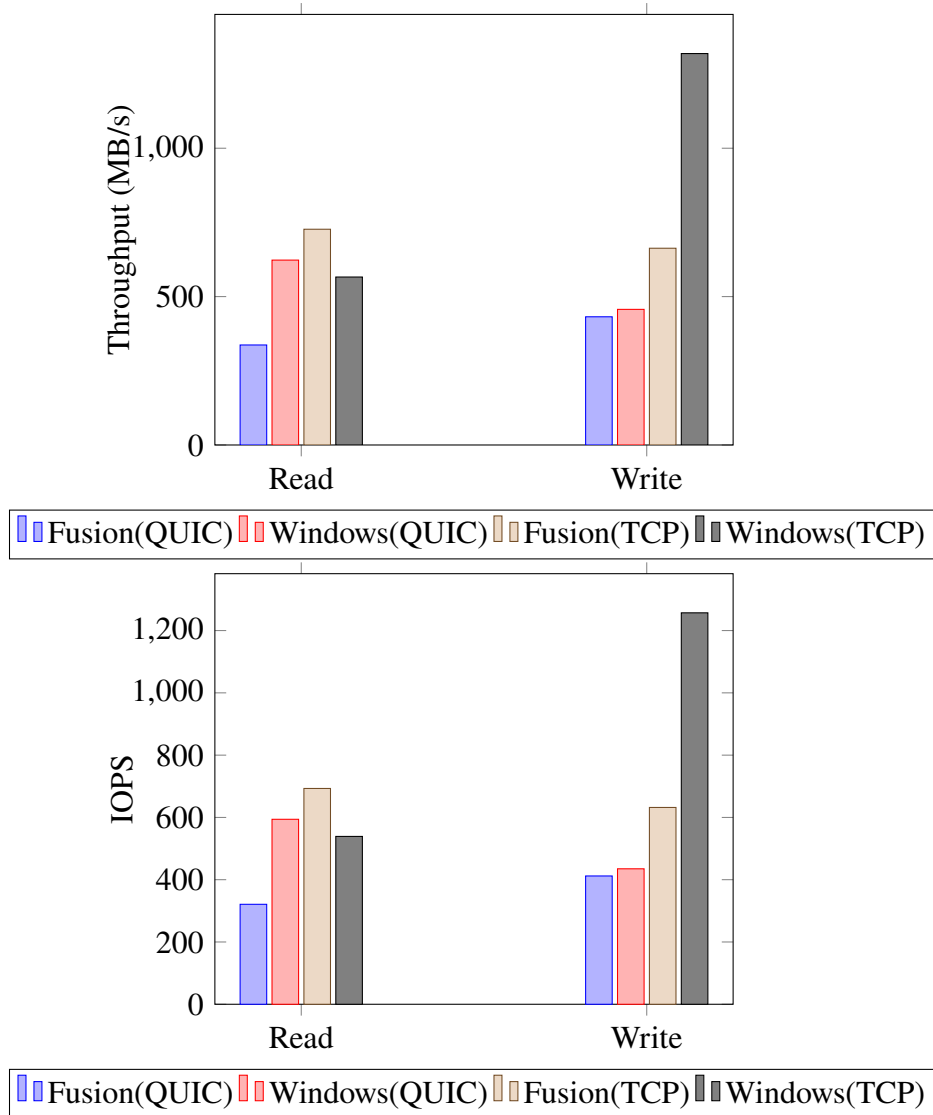
Figure 18: Scenario 1 results: Large sequential I/O

encrypted TCP. The reason for this outlier is not evident, the gain could potentially be from offloading of cryptographic processing to a separate thread or some other optimization in the threading model of the implementation. Due to this outlier the Windows SMB over QUIC implementation suffers a 65% performance reduction when compared to encrypted TCP. In comparison, the Fusion QUIC transport prototype sees a 35% reduction in throughput for the write workload when compared to encrypted TCP. For the read workload the Windows QUIC implementation actually performed slightly better than using encrypted TCP, while the Fusion QUIC implementation sees a 50% reduction in throughput as compared to encrypted TCP, suggesting that there are optimizations that could be made to the transmitting data path in the QUIC prototype, to bring the comparative performance in line with the read throughput. This is also evident when comparing Windows QUIC performance with Fusion

QUIC performance. The write workload, which is mainly bounded by the client's performance, shows very similar performance between both QUIC implementations. In comparison, the read workload, which is more dependent on the server, shows that the Fusion QUIC implementation has a 45% lower throughput than the Windows QUIC implementation.

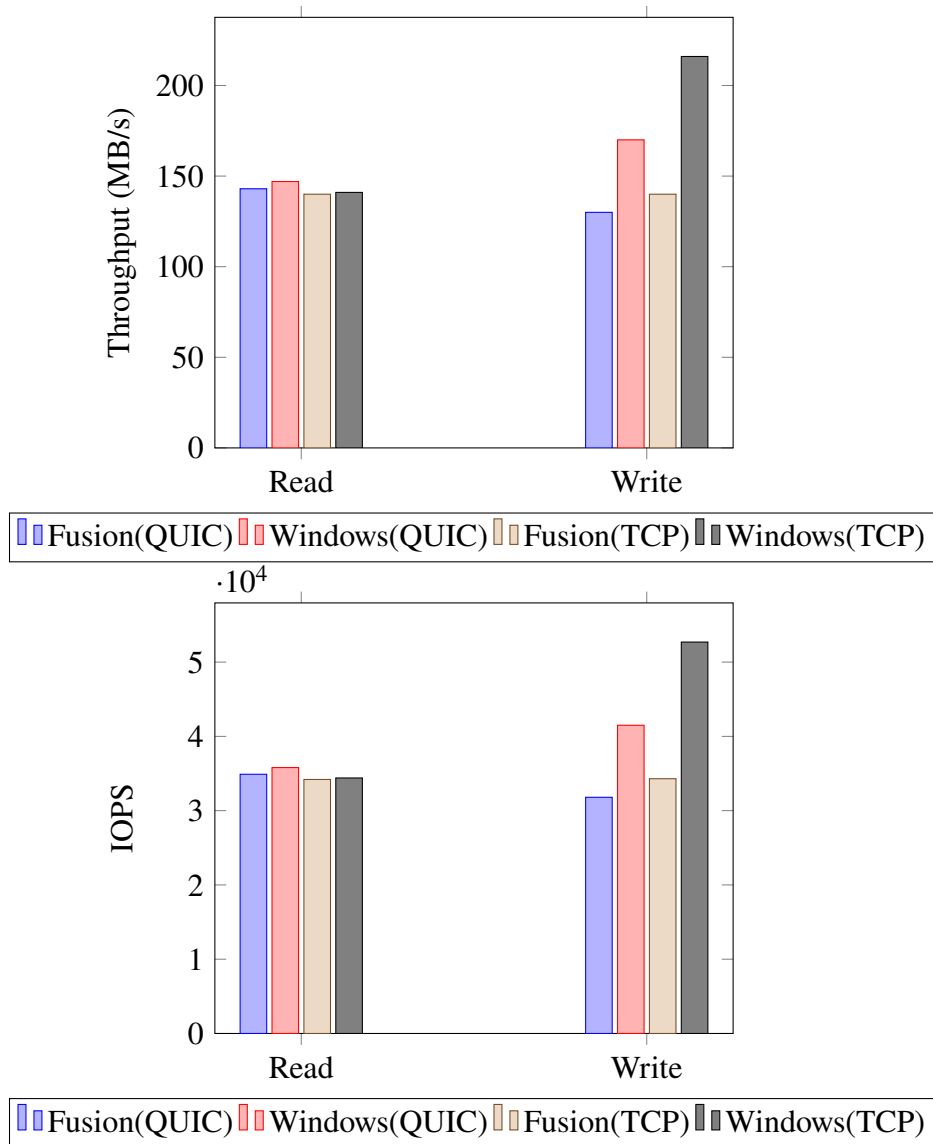### 6.3.2 Scenario 2: Single file random I/O

Figure 19: Scenario 2 results: single file random I/O

Scenario 2, in comparison to scenario 1, shows much less impact from the different transports, which can be seen in Figure 19. For the read workload it is clear that the difference in results is insignificant, with all transports and hosts showing very similar throughput, for both QUIC and encrypted TCP. In the case of the write workload it

can be seen that as in scenario 1, the Windows server shows a higher throughput, and the difference between QUIC and encrypted TCP is significant for the Windows implementation. Again, the Windows QUIC implementation has slightly higher throughput than the Fusion QUIC implementation, but the difference is less severe than in Scenario 1, with the Fusion QUIC implementation suffering a 30% decrease in throughput. From Scenario 2, it can be seen that the bottleneck is not only the network transport layer, resulting in similar performance regardless if QUIC or encrypted TCP is used.

6.3.3 Scenario 3: multiple files random I/O
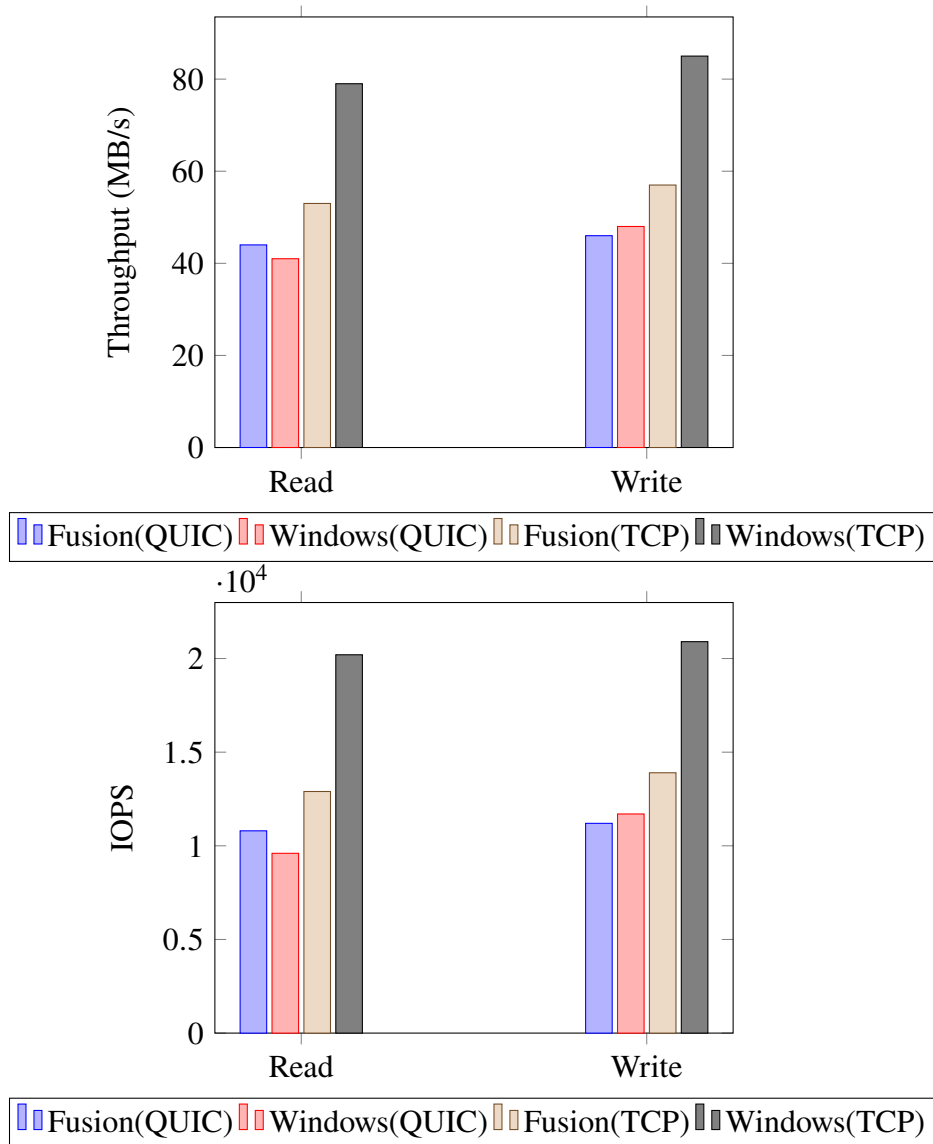
Figure 20: Scenario 3 results: multiple files random I/O

Scenario 3, illustrated in Figure 20, shows that Windows using encrypted TCP has

a significant advantage in this scenario, when compared to the other implementations. This advantage holds for both for read and write workloads. However, when comparing the performance of SMB over QUIC, the throughput of Windows and Fusion is very close, with Fusion being slightly favored in the read workload, and Windows having a slight advantage in the write workload. The throughput of Fusion's implementation of QUIC and encrypted TCP is close, just as in scenario 2. In comparison, the performance impact of using QUIC instead of encrypted TCP is more significant for the Windows implementation, which can be seen in the results of all three scenarios.

7 Conclusions

This thesis has explored the feasibility of implementing SMB on top of QUIC, and a prototype SMB over QUIC implementation has been created. Due to the fact that the Windows SMB over QUIC client and server implementation only uses one stream per connection, the semantics of SMB over QUIC is very similar to that of the traditional TCP-based implementation, making it very feasible to implement SMB on top of QUIC. The biggest hurdle in implementing SMB over QUIC is the lack of native QUIC support in Linux-based systems, necessitating the use of external libraries to provide support for the QUIC protocol. The external library chosen for the implementation, MsQuic, differs in processing model and interface from standard BSD-style sockets, leading to the design and work needed to integrate the QUIC layer being non-trivial.

The biggest benefits of using a QUIC-based SMB implementation as compared to the TCP-based implementation is the improved cryptography of TLS 1.3 and the move from TCP port 445, which is commonly blocked by ISPs, to UDP port 443, which is also used by HTTP/3, enabling the SMB traffic to traverse the internet without getting blocked. The biggest disadvantage is the negative performance impact of using QUIC as compared to TCP in high-throughput scenarios.

During the process of this thesis, two SMB over QUIC implementations have been benchmarked during optimal network conditions, and the results have been presented. Additionally, the same benchmarks were run against SMB over encrypted TCP, and the performance of SMB using TCP and QUIC was compared. The benchmarks show that during optimal network conditions the QUIC-based implementations' performance suffers when compared to TCP-based implementations, but the performance numbers from the QUIC-based implementations are promising and the performance is already sufficient for general-purpose tasks. The rest of this section of the thesis will summarize the key points and findings of the thesis and add some discussion about the possible expansions of SMB over QUIC.

7.1 Discussion

This work has been successful in designing and implementing a prototype QUIC transport layer for the Fusion SMB server to provide SMB over QUIC functionality. While the prototype's performance somewhat suffers in scenarios that are heavily dependent on the network transport throughput, the Fusion SMB over QUIC implementation still managed to nearly match the performance of encrypted TCP for workloads that are not only dependent on the network transport performance. The prototype is a promising first step in implementing an SMB over QUIC implementation for Linux based systems.

While there is room for improvements in the performance of the QUIC transport layer, the implemented prototype still shows relatively good performance. While under ideal conditions the QUIC transport layer suffers when compared to encrypted TCP, it must be highlighted that one of the main advantages of SMB over QUIC is the ability to access SMB shares from external networks, as compared to legacy SMB access which is most commonly done inside internal networks or over Virtual Private Networks

(VPN). This is since that the standard SMB TCP port 445 is commonly blocked by ISPs, affecting the ability of SMB traffic to traverse the internet. Considering that most ISPs only offer internet bandwidths of up to 1 Gb/s (125 MB/s), the prototype is already able to sufficiently support access over the internet.

During the process of working on this thesis it became clear that the extensions to the SMB protocol made by Microsoft to support SMB over QUIC were done in such a way that they do not take advantage of many of the improvements offered by QUIC when compared to TCP. There are only two QUIC specific features utilized by SMB. First, transport level encryption with the SMB protocol, moving the encryption from an SMB specific per-share basis to instead use QUIC's native encryption. Second, SMB over QUIC implements mutual authentication, allowing authentication of clients via certificates. However, there is at least one significant QUIC feature missing, from the [MS-SMB2] documentation *"By default, Windows SMB2 clients and servers always use a single stream per QUIC connection. Sending or receiving more than one stream per connection will be blocked by the underlying transport QUIC."* [3]. This, in turn, results in the SMB protocol entirely neglecting to use one of the biggest improvements brought on by QUIC, the ability to use multiple data streams over one connection. The following are some of the author's suggestions on how the SMB protocol could properly implement support for QUIC and its modern features.

As QUIC streams are by design lightweight, the overhead of opening new streams, when compared to for example the overhead introduced by filesystem I/O, is negligible. The main drawback to implementing a system of multiple streams per connection is the increased complexity, as there are no guarantees of in-order delivery between the streams. A server or client might transmit two streams, one after the other, and they may arrive in the reverse order. However, the SMB protocol already has added support for SMB Multichannel, which enables the server and client to transmit and receive over multiple Network Interface Controllers (NIC), or utilize multiple paths over one NIC via Receive Side Scaling (RSS), simultaneously. Thus, the problem of data transfer over multiple data streams has already been solved via earlier extensions to the protocol.

The main issue is then working out how multiple requests should be split up between different streams, and what the granularity of these splits could be. The naive approach is to map every request or compounded request to its own stream. One stream would then consist of a request and a response, or a set of compounded requests and compounded responses. This is similar to the design of HTTP/3 or DNS over QUIC. The advantage of this approach is that it takes full advantage of the multiplexed streams implemented by QUIC, going a long way towards negating HOL blocking. Another approach is to use streams on a per-open basis, setting up a stream for each file or directory that is opened, and sending all operations concerning that file or directory over its specific stream. The advantage here is lesser complexity than having all requests on separate streams, while still managing to take advantage of QUIC's advanced features.

There are some other advantages that may be implemented regardless of how the requests are split between streams. An advantage could be gained by creating a dedicated control stream, in addition to the normal data streams. The data streams

would handle the main data transfer requests, such as SMB2 OPEN, READ, WRITE and so on. By using QUIC it is possible to prioritize the streams, potentially allowing fast operations, such as directory listing and file or directory opens, to be completed quickly without being blocked by for example large reads and writes. In addition to the data streams, the control stream would be responsible for control information, including but not limited to NEGOTIATE, SESSION_SETUP, CHANGE_NOTIFY, CANCEL and ECHO requests, enabling the control channel to handle control messages separately from the data transfers. An advantage of a dedicated control stream is that the SMB credit system could be moved there, enabling the server and client to handle credit requests and responses separately from other SMB commands.

7.2 Future work

While a prototype for the QUIC transport layer has been achieved, there is still room for further development and investigation. Currently the solution works as a middle layer between MsQUIC and the Fusion SMB server, creating an interface that both sides can use for communication. While this has allowed for creating the prototype in the allocated timeframe, and shows promising performance when compared to encrypted TCP, there are still possible gains to be had. By continuing development, it could be possible to natively support the MsQuic library, or more precisely support the event driven processing model presented by MsQuic. At the same time this could improve the overhead from memory management, by minimizing the number of allocations, deallocations and copies that is made on the data path between the MsQuic library and the Fusion SMB server, improving both performance and long-term stability.

The focus of the prototype was ensuring stability, preventing memory leaks or errors. Due to the time constraint this resulted in inefficient memory management to reduce implementation complexity. Future work could incrementally improve memory management, for example by implementing pools of pre-allocated objects to store information about received buffers.

An aspect that was not explored by this thesis is the existence of a multitude of different QUIC libraries. One of the advantages presented by QUIC is the fact that most implementations are located in the user space. By prototyping any number of libraries, it is possible to compare performance and ease-of-use, enabling a more informed decision on which library is desirable for use in the final implementation, or possibly supporting multiple libraries on a case-by-case basis. Considering that the development is done for Linux based machines, one of the most promising options is the QUIC kernel module being developed, as outlined in Section 5.

One of the advantages of QUIC as compared to TCP is the improved latency of connection establishment. By benchmarking the set up of many SMB over QUIC connections, against setting up SMB over encrypted TCP connections, it could be possible to benchmark this aspect of the implementation. However, as Microsoft's SMB over QUIC client is at the time of writing to the author's knowledge the only publicly available SMB over QUIC client, the process of setting up the number of clients required for the test is far from straightforward. Once an alternative client is available, it may enable running connection set up benchmarking.

As discussed, the Microsoft SMB over QUIC implementation does not take advantage of most of QUIC's advanced features, instead opting to use QUIC in a very similar manner to TCP. Further work could focus on improving the design of SMB over QUIC, as well as prototyping an implementation that takes advantage of QUIC's improvements over TCP.

References

[1] J. Postel. *Transmission Control Protocol*. RFC 793. Sept. 1981. DOI: 10. 17487/RFC0793. URL: https://www.rfc-editor.org/info/rfc793.

[2] J. Postel. *User Datagram Protocol*. RFC 768. Aug. 1980. DOI: 10.17487/ RFC0768. URL: https://www.rfc-editor.org/info/rfc768.

[3] Microsoft. *[MS-SMB2]: Server Message Block (SMB) Protocol Versions 2 and 3*. URL: https://learn.microsoft.com/en-us/openspecs/windows_ protocols/ms-smb2/5606ad47-5ee0-437a-817e-70c366052962 (visited on Aug. 26, 2025).

[4] A. Langley et al. "The QUIC Transport Protocol: Design and Internet-Scale Deployment". In: *Proceedings of the Conference of the ACM Special Interest Group on Data Communication*. SIGCOMM '17. Los Angeles, CA, USA: Association for Computing Machinery, 2017, pp. 183–196. ISBN: 9781450346535. DOI: 10.1145/3098822.3098842. URL: https://doi. org/10.1145/3098822.3098842.

[5] J. Iyengar and M. Thomson. *QUIC: A UDP-Based Multiplexed and Secure Transport*. RFC 9000. May 2021. DOI: 10.17487/RFC9000. URL: https: //www.rfc-editor.org/info/rfc9000.

[6] M. Thomson and S. Turner. *Using TLS to Secure QUIC*. RFC 9001. May 2021. DOI: 10.17487/RFC9001. URL: https://www.rfc-editor.org/info/ rfc9001.

[7] M. Bishop. *HTTP/3*. RFC 9114. June 2022. DOI: 10.17487/RFC9114. URL: https://www.rfc-editor.org/info/rfc9114.

[8] S. Cook et al. "QUIC: Better for what and for whom?" In: *2017 IEEE International Conference on Communications (ICC)*. 2017, pp. 1–6. DOI: 10.1109/ICC.2017.7997281.

[9] T. Shreedhar et al. "Evaluating QUIC Performance Over Web, Cloud Storage, and Video Workloads". In: *IEEE Transactions on Network and Service Management* 19.2 (2022), pp. 1366–1381. DOI: 10.1109/TNSM.2021. 3134562.

[10] K. Nepomuceno et al. "QUIC and TCP: A Performance Evaluation". In: *2018 IEEE Symposium on Computers and Communications (ISCC)*. 2018, pp. 00045–00051. DOI: 10.1109/ISCC.2018.8538687.

[11] Broadband Internet Technical Advisory Group (BITAG). *Port Blocking*. Tech. rep. Technical report documenting ISP port-blocking practices including TCP/139 and TCP/445. BITAG, Feb. 2015. URL: https://www.bitag. org/documents/Port-Blocking.pdf.

[12] Microsoft. *MsQuic*. URL: https://github.com/microsoft/msquic (visited on Aug. 26, 2025).

[13] Tuxera. *Tuxera Fusion SMB*. URL: https://www.tuxera.com/products/tuxera-fusion-smb/ (visited on Aug. 26, 2025).

[14] J. Postel. *Internet Protocol*. RFC 791. Sept. 1981. DOI: 10.17487/RFC0791. URL: https://www.rfc-editor.org/info/rfc791.

[15] A. Ford et al. *TCP Extensions for Multipath Operation with Multiple Addresses*. RFC 8684. Mar. 2020. DOI: 10.17487/RFC8684. URL: https://www.rfc-editor.org/info/rfc8684.

[16] J. Kurose and K. Ross. *Computer networking: A top-down approach, global edition*. en. 8th ed. London, England: Pearson Education, June 2021.

[17] Margaret Rouse. *Data Stream*. URL: https://www.techopedia.com/definition/6757/data-stream (visited on Aug. 28, 2025).

[18] B. Ford. "Structured streams: a new transport abstraction". In: *Proceedings of the 2007 Conference on Applications, Technologies, Architectures, and Protocols for Computer Communications*. SIGCOMM '07. Kyoto, Japan: Association for Computing Machinery, 2007, pp. 361–372. ISBN: 9781595937131. DOI: 10.1145/1282380.1282421. URL: https://doi.org/10.1145/1282380.1282421.

[19] R. R. Stewart. *Stream Control Transmission Protocol*. RFC 4960. Sept. 2007. DOI: 10.17487/RFC4960. URL: https://www.rfc-editor.org/info/rfc4960.

[20] R. R. Stewart, M. Tüxen, and karen Nielsen. *Stream Control Transmission Protocol*. RFC 9260. June 2022. DOI: 10.17487/RFC9260. URL: https://www.rfc-editor.org/info/rfc9260.

[21] H. Nielsen, R. T. Fielding, and T. Berners-Lee. *Hypertext Transfer Protocol – HTTP/1.0*. RFC 1945. May 1996. DOI: 10.17487/RFC1945. URL: https://www.rfc-editor.org/info/rfc1945.

[22] H. Nielsen et al. *Hypertext Transfer Protocol – HTTP/1.1*. RFC 2616. June 1999. DOI: 10.17487/RFC2616. URL: https://www.rfc-editor.org/info/rfc2616.

[23] E. Rescorla. *HTTP Over TLS*. RFC 2818. May 2000. DOI: 10.17487/RFC2818. URL: https://www.rfc-editor.org/info/rfc2818.

[24] Google. *SPDY: An experimental protocol for a faster web*. URL: https://www.chromium.org/spdy/spdy-whitepaper/ (visited on Aug. 29, 2025).

[25] M. Thomson and C. Benfield. *HTTP/2*. RFC 9113. June 2022. DOI: 10.17487/RFC9113. URL: https://www.rfc-editor.org/info/rfc9113.

[26] M. Belshe, R. Peon, and M. Thomson. *Hypertext Transfer Protocol Version 2 (HTTP/2)*. RFC 7540. May 2015. DOI: 10.17487/RFC7540. URL: https://www.rfc-editor.org/info/rfc7540.

[27] R. Peon and H. Ruellan. *HPACK: Header Compression for HTTP/2*. RFC 7541. May 2015. DOI: 10.17487/RFC7541. URL: https://www.rfc-editor.org/info/rfc7541.

[28] C. '. Krasic, M. Bishop, and A. Frindell. *QPACK: Field Compression for HTTP/3*. RFC 9204. June 2022. DOI: 10.17487/RFC9204. URL: https://www.rfc-editor.org/info/rfc9204.

[29] A. O. Freier, P. Karlton, and P. C. Kocher. *The Secure Sockets Layer (SSL) Protocol Version 3.0*. RFC 6101. Aug. 2011. DOI: 10.17487/RFC6101. URL: https://www.rfc-editor.org/info/rfc6101.

[30] E. Rescorla. *The Transport Layer Security (TLS) Protocol Version 1.3*. RFC 8446. Aug. 2018. DOI: 10.17487/RFC8446. URL: https://www.rfc-editor.org/info/rfc8446.

[31] C. Allen and T. Dierks. *The TLS Protocol Version 1.0*. RFC 2246. Jan. 1999. DOI: 10.17487/RFC2246. URL: https://www.rfc-editor.org/info/rfc2246.

[32] T. Dierks and E. Rescorla. *The Transport Layer Security (TLS) Protocol Version 1.1*. RFC 4346. Apr. 2006. DOI: 10.17487/RFC4346. URL: https://www.rfc-editor.org/info/rfc4346.

[33] K. Moriarty and S. Farrell. *Deprecating TLS 1.0 and TLS 1.1*. RFC 8996. Mar. 2021. DOI: 10.17487/RFC8996. URL: https://www.rfc-editor.org/info/rfc8996.

[34] E. Rescorla and T. Dierks. *The Transport Layer Security (TLS) Protocol Version 1.2*. RFC 5246. Aug. 2008. DOI: 10.17487/RFC5246. URL: https://www.rfc-editor.org/info/rfc5246.

[35] T. Polk and S. Turner. *Prohibiting Secure Sockets Layer (SSL) Version 2.0*. RFC 6176. Mar. 2011. DOI: 10.17487/RFC6176. URL: https://www.rfc-editor.org/info/rfc6176.

[36] E. Rescorla, H. Tschofenig, and N. Modadugu. *The Datagram Transport Layer Security (DTLS) Protocol Version 1.3*. RFC 9147. Apr. 2022. DOI: 10.17487/RFC9147. URL: https://www.rfc-editor.org/info/rfc9147.

[37] A. Silberschatz, P. B. Galvin, and G. Gagne. *Operating System Concepts*. 10th ed. John Wiley & Sons, 2021, pp. 757–769.

[38] T. Haynes and D. Noveck. *Network File System (NFS) Version 4 Protocol*. RFC 7530. Mar. 2015. DOI: 10.17487/RFC7530. URL: https://www.rfc-editor.org/info/rfc7530.

[39] Andrew Tridgell. *Myths About Samba*. URL: https://www.samba.org/samba/docs/myths_about_samba.html?ref=cybersecuritycrumbs.cryptopathe.me (visited on Sept. 2, 2025).

[40] R. Sandberg et al. "Design and implementation of the Sun network filesystem". In: *Proceedings of the summer 1985 USENIX conference*. 1985, pp. 119–130.

[41] T. Haynes. *Network File System (NFS) Version 4 Minor Version 2 Protocol.* RFC 7862. Nov. 2016. DOI: 10.17487/RFC7862. URL: https://www.rfc-editor.org/info/rfc7862.

[42] D. Noveck and C. Lever. *Network File System (NFS) Version 4 Minor Version 1 Protocol.* RFC 8881. Aug. 2020. DOI: 10.17487/RFC8881. URL: https://www.rfc-editor.org/info/rfc8881.

[43] H. de Saxcé, I. Oprescu, and Y. Chen. "Is HTTP/2 really faster than HTTP/1.1?" In: *2015 IEEE Conference on Computer Communications Workshops (INFOCOM WKSHPS)*. 2015, pp. 293–299. DOI: 10.1109/INFCOMW.2015.7179400.

[44] Y. Cheng et al. *TCP Fast Open.* RFC 7413. Dec. 2014. DOI: 10.17487/RFC7413. URL: https://www.rfc-editor.org/info/rfc7413.

[45] J. Corbet. *LPC: Making the net go faster.* URL: https://lwn.net/Articles/458610/ (visited on Sept. 17, 2025).

[46] K. Edeline and B. Donnet. "A Bottom-Up Investigation of the Transport-Layer Ossification". In: *2019 Network Traffic Measurement and Analysis Conference (TMA)*. 2019, pp. 169–176. DOI: 10.23919/TMA.2019.8784690.

[47] D. J. D. Touch and W. Eddy. *TCP Extended Data Offset Option.* Internet-Draft draft-ietf-tcpm-tcp-edo-15. Work in Progress. Internet Engineering Task Force, May 2025. 23 pp. URL: https://datatracker.ietf.org/doc/draft-ietf-tcpm-tcp-edo/15/.

[48] J. Roskind. *QUIC: Design Document and Specification Rationale — docs.google.com.* https://docs.google.com/document/d/1RNHkx_VvKWyWg6Lr8SZ-saqsQx7rFV-ev2jRFUoVD34/edit?usp=sharing. [Accessed 14-08-2025].

[49] Cloudflare. *Cloudflare Radar.* URL: https://radar.cloudflare.com/ (visited on Sept. 17, 2025).

[50] J. Iyengar and I. Swett. *QUIC Loss Detection and Congestion Control.* RFC 9002. May 2021. DOI: 10.17487/RFC9002. URL: https://www.rfc-editor.org/info/rfc9002.

[51] M. Thomson. *Version-Independent Properties of QUIC.* RFC 8999. May 2021. DOI: 10.17487/RFC8999. URL: https://www.rfc-editor.org/info/rfc8999.

[52] *QUIC, a multiplexed transport over UDP — chromium.org.* https://www.chromium.org/quic/. [Accessed 14-08-2025].

[53] *How to enable HTTP/3 support in Firefox.* https://www.ghacks.net/2020/07/01/how-to-enable-http-3-support-in-firefox/. [Accessed 14-08-2025].

[54] *Examining HTTP/3 usage one year on — blog.cloudflare.com.* https://blog.cloudflare.com/http3-usage-one-year-on/. [Accessed 14-08-2025].

[55] C. Huitema, S. Dickinson, and A. Mankin. *DNS over Dedicated QUIC Connections*. RFC 9250. May 2022. DOI: 10.17487/RFC9250. URL: https://www.rfc-editor.org/info/rfc9250.

[56] A. Gurtov et al. *The NewReno Modification to TCP's Fast Recovery Algorithm*. RFC 6582. Apr. 2012. DOI: 10.17487/RFC6582. URL: https://www.rfc-editor.org/info/rfc6582.

[57] L. Eggert, G. Fairhurst, and G. Shepherd. *UDP Usage Guidelines*. RFC 8085. Mar. 2017. DOI: 10.17487/RFC8085. URL: https://www.rfc-editor.org/info/rfc8085.

[58] *Server Message Block (SMB) Protocol Versions 2 and 3*. Tech. rep. Release: July 28, 2025. Version v83.0. Microsoft Corporation, July 2025. URL: https://winprotocoldoc.z19.web.core.windows.net/MS-SMB2/%5bMS-SMB2%5d.pdf.

[59] P. J. Leach et al. *The Simple and Protected Generic Security Service Application Program Interface (GSS-API) Negotiation Mechanism*. RFC 4178. Oct. 2005. DOI: 10.17487/RFC4178. URL: https://www.rfc-editor.org/info/rfc4178.

[60] D. C. Neuman et al. *The Kerberos Network Authentication Service (V5)*. RFC 4120. July 2005. DOI: 10.17487/RFC4120. URL: https://www.rfc-editor.org/info/rfc4120.

[61] *NT LAN Manager (NTLM) Authentication Protocol*. Tech. rep. Release: April 23, 2024. Version v36.0. Microsoft Corporation, Apr. 2024. URL: https://winprotocoldoc.z19.web.core.windows.net/MS-NLMP/%5bMS-NLMP%5d.pdf.

[62] R. J. Recio et al. *A Remote Direct Memory Access Protocol Specification*. RFC 5040. Oct. 2007. DOI: 10.17487/RFC5040. URL: https://www.rfc-editor.org/info/rfc5040.

[63] Microsoft. *SMB Direct*. URL: https://learn.microsoft.com/en-us/windows-server/storage/file-server/smb-direct (visited on Sept. 5, 2025).

[64] Microsoft. *SMB over QUIC*. URL: https://learn.microsoft.com/en-us/windows-server/storage/file-server/smb-over-quic (visited on Sept. 5, 2025).

[65] Microsoft. *SMB over QUIC*. URL: https://learn.microsoft.com/en-us/windows-server/storage/file-server/configure-smb-over-quic-client-access-control (visited on Sept. 5, 2025).

[66] QUIC Working Group. *Implementations and tools*. URL: https://github.com/quicwg/quicwg.github.io/blob/main/implementations.md (visited on Sept. 8, 2025).

[67] Xin Long. *QUIC in Linux Kernel*. URL: https://github.com/lxin/quic (visited on Sept. 8, 2025).

[68] M. Kempf et al. "QUIC on the Fast Lane: Extending Performance Evaluations on High-rate Links". In: *Computer Communications* 223 (2024), pp. 90–100. ISSN: 0140-3664. DOI: https://doi.org/10.1016/j.comcom.2024.04.038. URL: https://www.sciencedirect.com/science/article/pii/S014036642400166X.

[69] Microsoft. *MsQuic*. URL: https://microsoft.github.io/msquic/msquicdocs/index.html (visited on Sept. 8, 2025).

[70] S. Friedl et al. *Transport Layer Security (TLS) Application-Layer Protocol Negotiation Extension*. RFC 7301. July 2014. DOI: 10.17487/RFC7301. URL: https://www.rfc-editor.org/info/rfc7301.

[71] A. Cox. *Network Buffers And Memory Management*. URL: https://tldp.org/LDP/khg/HyperNews/get/net/net-intro.html (visited on Sept. 29, 2025).

[72] N. Banks. *QUIC Performance*. Internet-Draft draft-banks-quic-performance-00. Work in Progress. Internet Engineering Task Force, Dec. 2020. 8 pp. URL: https://datatracker.ietf.org/doc/draft-banks-quic-performance/00/.

A Benchmark commands

```
SecNetPerf:
./secnetperf -target:<server hostname> -exec:maxtput
-<down/up>:60s -ptput:1


FIO scenario 1 (single sequential)
fio --name=single-seq -rw=<write/read> --filesize=5g -bs=1m
--nrfiles=1 --ioengine=windowsaio --iodepth=4 --direct=1
--time_based --ramp_time=15 --runtime=120 --fallocate=native


FIO scenario 2 (single random)
fio --name=single-rand -rw=rand<write/read> --size=1g -bs=4k
--nrfiles=1 --ioengine=windowsaio --iodepth=16 --direct=1
--time_based --ramp_time=15 --runtime=120 --fallocate=native
--numjobs=1


FIO scenario 3 (multiple random)
fio --name=multiple-rand -rw=rand<write/read> --filesize=100k
-bs=4k --nrfiles=16 --ioengine=windowsaio --iodepth=4
--direct=1 --time_based --ramp_time=15 --runtime=120
--fallocate=native --create_serialize=0
```