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Architectural Guidelines for Multipath TCP Development
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Abstract

Hosts are often connected by multiple paths, but TCP restricts communications to a single path per transport connection. Resource usage within the network would be more efficient were these multiple paths able to be used concurrently. This should enhance user experience through improved resilience to network failure and higher throughput.

This document outlines architectural guidelines for the development of a Multipath Transport Protocol, with references to how these architectural components come together in the development of a Multipath TCP protocol. This document lists certain high level design decisions that provide foundations for the design of the MPTCP protocol, based upon these architectural requirements.

Status of this Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

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This Internet-Draft will expire on July 25, 2011.

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1. Introduction

As the Internet evolves, demands on Internet resources are ever-increasing, but often these resources (in particular, bandwidth) cannot be fully utilised due to protocol constraints both on the end-systems and within the network. If these resources could be used concurrently, end user experience could be greatly improved. Such enhancements would also reduce the necessary expenditure on network infrastructure that would otherwise be needed to create an equivalent improvement in user experience. By the application of resource pooling [3], these available resources can be 'pooled' such that they appear as a single logical resource to the user.

Multipath transport aims to realize some of the goals of resource pooling by simultaneously making use of multiple disjoint (or partially disjoint) paths across a network. The two key benefits of multipath transport are:

- To increase the resilience of the connectivity by providing multiple paths, protecting end hosts from the failure of one.
- To increase the efficiency of the resource usage, and thus increase the network capacity available to end hosts.

Multipath TCP is a modified version of TCP [1] that implements a multipath transport and achieves these goals by pooling multiple paths within a transport connection, transparently to the application. Multipath TCP is primarily concerned with utilising multiple paths end-to-end, where one or both end host is multi-homed. It may also have applications where multiple paths exist within the network and can be manipulated by an end host, such as using different port numbers with ECMP [4].

MPTCP, defined in [5], is a specific protocol that instantiates the Multipath TCP concept. This document looks both at general architectural principles for a Multipath TCP fulfilling the goals described in Section 2, as well as the key design decisions behind MPTCP, which are detailed in Section 5.

Although multihoming and multipath functions are not new to transport protocols (SCTP [6] being a notable example), MPTCP aims to gain wide-scale deployment by recognising the importance of application and network compatibility goals. These goals, discussed in detail in Section 2, relate to the appearance of MPTCP to the network (so non-MPTCP-aware entities see it as TCP) and to the application (through providing an service equivalent to TCP for non-MPTCP-aware applications).

This document has three key purposes: (i) it describes goals for a multipath transport - goals that MPTCP is designed to meet; (ii) it lays out an architectural basis for MPTCP's design - a discussion that applies to other multipath transports as well; and (iii) it discusses and documents high-level design decisions made in MPTCP's development, and considers their implications.

Companion documents to this architectural overview are those which provide details of the protocol extensions [5], congestion control algorithms [7], and application-level considerations [8]. Put together, these components specify a complete Multipath TCP design. We note that specific components are replaceable in accordance with the layer and functional decompositions discussed in this document.

1.1. Requirements Language

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119 [2].

1.2. Terminology

Regular/Single-Path TCP: The standard version of the TCP [1] protocol in use today, operating between a single pair of IP addresses.

Multipath TCP: A modified version of the TCP protocol that supports the simultaneous use of multiple paths between hosts.

Path: A sequence of links between a sender and a receiver, defined in this context by a source and destination address pair.

Host: An end host either initiating or terminating a Multipath TCP connection.

MPTCP: The proposed protocol extensions specified in [5] to provide a Multipath TCP implementation.

Subflow: A flow of TCP segments operating over an individual path, which forms part of a larger Multipath TCP connection.

(Multipath TCP) Connection: A set of one or more subflows combined to provide a single Multipath TCP service to an application at a host.

1.3. Reference Scenario

The diagram shown in Figure 1 illustrates a typical usage scenario for Multipath TCP. Two hosts, A and B, are communicating with each other. These hosts are multi-homed and multi-addressed, providing two disjoint connections to the Internet. The addresses on each host are referred to as A1, A2, B1 and B2. There are therefore up to four different paths between the two hosts: A1-B1, A1-B2, A2-B1, A2-B2.

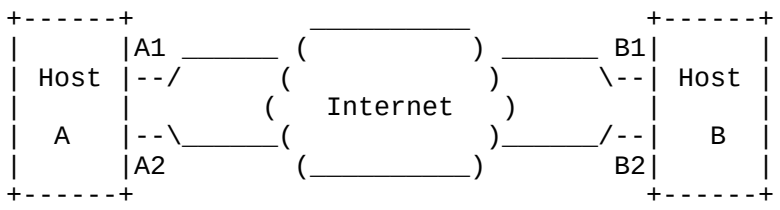

Figure 1: Simple Multipath TCP Usage Scenario

The scenario could have any number of addresses (1 or more) on each host, as long as the number of paths available between the two hosts is 2 or more (i.e. num_addr(A) * num_addr(B) > 1). The paths created by these address combinations through the Internet need not be entirely disjoint - potential fairness issues introduced by shared bottlenecks need to be handled by the Multipath TCP congestion controller. Furthermore, the paths through the Internet often do not provide a pure end-to-end service, and instead may be affected by middleboxes such as NATs and Firewalls.

2. Goals

This section outlines primary goals that Multipath TCP aims to meet. These are broadly broken down into: functional goals, which steer services and features that Multipath TCP must provide; and compatibility goals, which determine how Multipath TCP should appear to entities that interact with it.

2.1. Functional Goals

In supporting the use of multiple paths, Multipath TCP has the following two functional goals.

- Improve Throughput: Multipath TCP MUST support the concurrent use of multiple paths. To meet the minimum performance incentives for deployment, a Multipath TCP connection over multiple paths SHOULD achieve no lesser throughput than a single TCP connection over the best constituent path.

- o Improve Resilience: Multipath TCP MUST support the use of multiple paths interchangeably for resilience purposes, by permitting segments to be sent and re-sent on any available path. It follows that, in the worst case, the protocol MUST be no less resilient than regular single-path TCP.

As distribution of traffic among available paths and responses to congestion are done in accordance with resource pooling principles [3], a secondary effect of meeting these goals is that widespread use of Multipath TCP over the Internet should improve overall network utility by shifting load away from congested bottlenecks and by taking advantage of spare capacity wherever possible.

Furthermore, Multipath TCP SHOULD feature automatic negotiation of its use. A host supporting Multipath TCP that requires the other host to do so too must be able to detect reliably whether this host does in fact support the required extensions, using them if so, and otherwise automatically falling back to single-path TCP.

2.2. Compatibility Goals

In addition to the functional goals listed above, a Multipath TCP must meet a number of compatibility goals in order to support deployment in today's Internet. These goals fall into the following categories:

2.2.1. Application Compatibility

Application compatibility refers to the appearance of Multipath TCP to the application both in terms of the API that can be used and the expected service model that is provided.

Multipath TCP MUST follow the same service model as TCP [1]: in-order, reliable, and byte-oriented delivery. Furthermore, a Multipath TCP connection SHOULD provide the application with no worse throughput or resilience than it would expect from running a single TCP connection over any one of its available paths. A Multipath TCP may not, however, be able to provide the same level of consistency of throughput and latency as a single TCP connection. These, and other, application considerations are discussed in detail in [8].

A multipath-capable equivalent of TCP MUST retain some level of backward compatibility with existing TCP APIs, so that existing applications can use the newer transport merely by upgrading the operating systems of the end-hosts. This does not preclude the use of an advanced API to permit multipath-aware applications to specify preferences, nor for users to configure their systems in a different way from the default, for example switching on or off the automatic

use of multipath extensions.

It is possible for regular TCP sessions today to survive brief breaks in connectivity by retaining state at end hosts before a timeout occurs. It would be desirable to support similar session continuity in MPTCP, however the circumstances could be different. Whilst in regular TCP the IP addresses will remain constant across the break in connectivity, in MPTCP a different interface may appear. It is desirable (but not mandated) to support this kind of "break-before-make" session continuity. This places constraints on security mechanisms, however, as discussed in Section 5.8. Timeouts for this function would be locally configured.

2.2.2. Network Compatibility

In the traditional Internet architecture, network devices operate at the network layer and lower layers, with the layers above the network layer instantiated only at the end-hosts. While this architecture, shown in Figure 2, was initially largely adhered to, this layering no longer reflects the "ground truth" in the Internet with the proliferation of middleboxes [9]. Middleboxes routinely interpose on the transport layer; sometimes even completely terminating transport connections, thus leaving the application layer as the first real end-to-end layer, as shown in Figure 3.

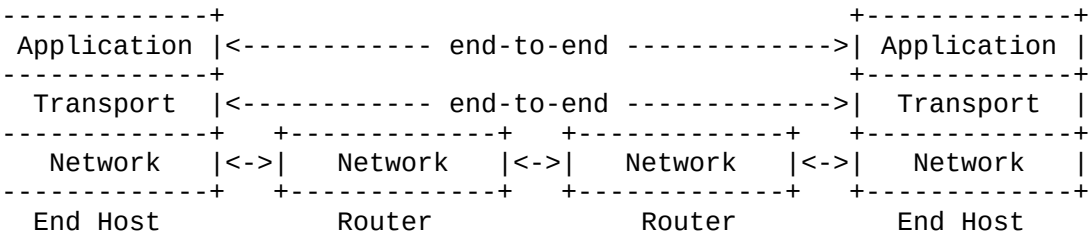

Figure 2: Traditional Internet Architecture

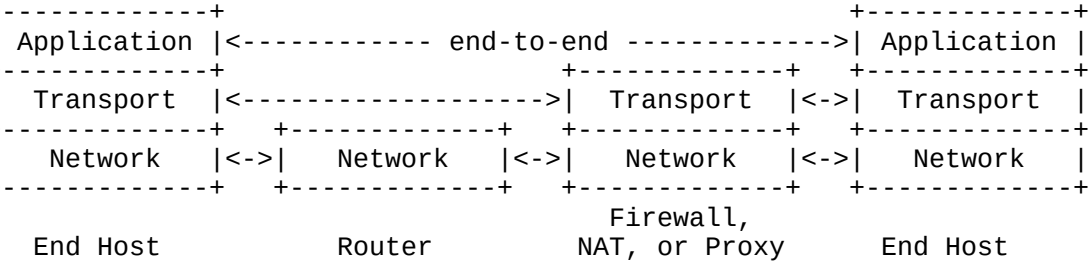

Figure 3: Internet Reality

Middleboxes that interpose on the transport layer result in loss of "fate-sharing" [10], that is, they often hold "hard" state that, when lost or corrupted, results in loss or corruption of the end-to-end transport connection.

The network compatibility goal requires that the multipath extension to TCP retains compatibility with the Internet as it exists today, including making reasonable efforts to be able to traverse predominant middleboxes such as firewalls, NATs, and performance enhancing proxies [9]. This requirement comes from recognizing middleboxes as a significant deployment bottleneck for any transport that is not TCP or UDP, and constrains Multipath TCP to appear as TCP does on the wire and to use established TCP extensions where necessary. To ensure end-to-endness of the transport, we further require Multipath TCP to preserve fate-sharing without making any assumptions about middlebox behavior.

A detailed analysis of middlebox behaviour and the impact on the Multipath TCP architecture is presented in Section 7. In addition, network compatibility must be retained to the extent that Multipath TCP MUST fall back to regular TCP if there are insurmountable incompatibilities for the multipath extension on a path.

Middleboxes may also cause some TCP features to be able to exist on one subflow but not another. Typically these will be at the subflow level (such as SACK [11]) and thus do not affect the connection-level behaviour. In the future, any proposed TCP connection-level extensions should consider how they can co-exist with MPTCP.

The modifications to support Multipath TCP remain at the transport layer, although some knowledge of the underlying network layer is required. Multipath TCP SHOULD work with IPv4 and IPv6 interchangeably, i.e. one connection may operate over both IPv4 and IPv6 networks.

2.2.3. Compatibility with other network users

As a corollary to both network and application compatibility, the architecture must enable new Multipath TCP flows to coexist gracefully with existing single-path TCP flows, competing for bandwidth neither unduly aggressively nor unduly timidly (unless low-precedence operation is specifically requested by the application, such as with LEDBAT). The use of multiple paths MUST NOT unduly harm users using single-path TCP at shared bottlenecks, beyond the impact that would occur from another single-path TCP flow. Multiple Multipath TCP flows on a shared bottleneck MUST share bandwidth between each other with similar fairness to that which occurs at a shared bottleneck with single-path TCP.

2.3. Security Goals

The extension of TCP with multipath capabilities will bring with it a number of new threats, analysed in detail in [12]. The security goal for Multipath TCP is to provide a service no less secure than regular, single-path TCP. This will be achieved through a combination of existing TCP security mechanisms (potentially modified to align with the Multipath TCP extensions) and of protection against the new multipath threats identified. The design decisions derived from this goal are presented in Section 5.8.

2.4. Related Protocols

There are several similarities between SCTP [6] and MPTCP, in that both can make use of multiple addresses at end hosts to give some multi-path capability. In SCTP, the primary use case is to support redundancy and mobility for multihomed hosts (i.e. a single path will change one of its end host addresses); the simultaneous use of multiple paths is not supported . Extensions are proposed to support simultaneous multipath transport [13], but these are yet to be standardised. By far the most widely used stream-based transport protocol is, however, TCP [1], and SCTP does not meet the network and application compatibility goals specified in Section 2.2. For network compatibility, there are issues with various middleboxes (especially NATs) that are unaware of SCTP and consequently end up blocking it. For application compatibility, applications need to actively choose to use SCTP, and with the deployment issues very few choose to do so. MPTCP’s compatibility goals are in part based on these observations of SCTP’s deployment issues.

3. An Architectural Basis For Multipath TCP

We now present one possible transport architecture that we believe can effectively support the goals for Multipath TCP. The new Internet model described here is based on ideas proposed earlier in Tng ("Transport next-generation") [14]. While by no means the only possible architecture supporting multipath transport, Tng incorporates many lessons learned from previous transport research and development practice, and offers a strong starting point from which to consider the extant Internet architecture and its bearing on the design of any new Internet transports or transport extensions.

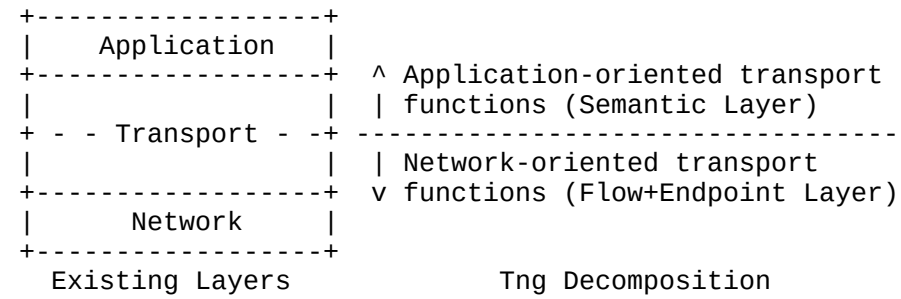

Figure 4: Decomposition of Transport Functions

Tng loosely splits the transport layer into "application-oriented" and "network-oriented" layers, as shown in Figure 4. The application-oriented "Semantic" layer implements functions driven primarily by concerns of supporting and protecting the application's end-to-end communication, while the network-oriented "Flow+Endpoint" layer implements functions such as endpoint identification (using port numbers) and congestion control. These network-oriented functions, while traditionally located in the ostensibly "end-to-end" Transport layer, have proven in practice to be of great concern to network operators and the middleboxes they deploy in the network to enforce network usage policies [15] [16] or optimize communication performance [17]. Figure 5 shows how middleboxes interact with different layers in this decomposed model of the transport layer: the application-oriented layer operates end-to-end, while the network-oriented layer operates "segment-by-segment" and can be interposed upon by middleboxes.

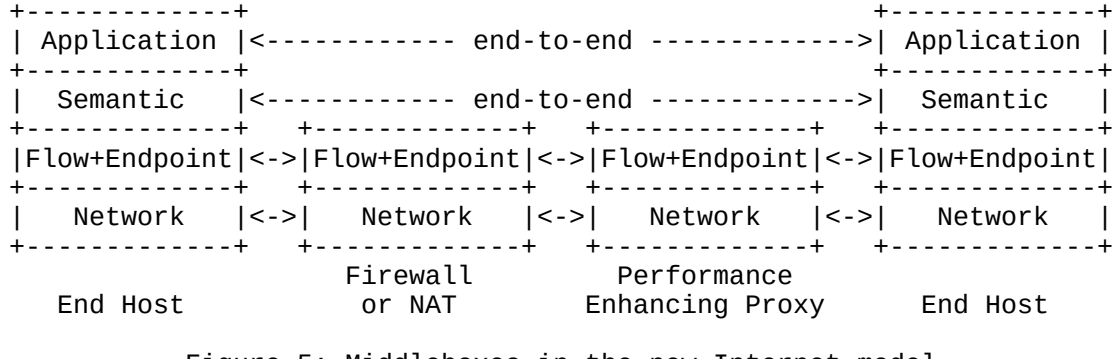

Figure 5: Middleboxes in the new Internet model

MPTCP's architectural design follows Tng's decomposition as shown in Figure 6. MPTCP, which provides application compatibility through the preservation of TCP-like semantics of global ordering of application data and reliability, is an instantiation of the

"application-oriented" Semantic layer; whereas the subflow TCP component, which provides network compatibility by appearing and behaving as a TCP flow in the network, is an instantiation of the "network-oriented" Flow+Endpoint layer.

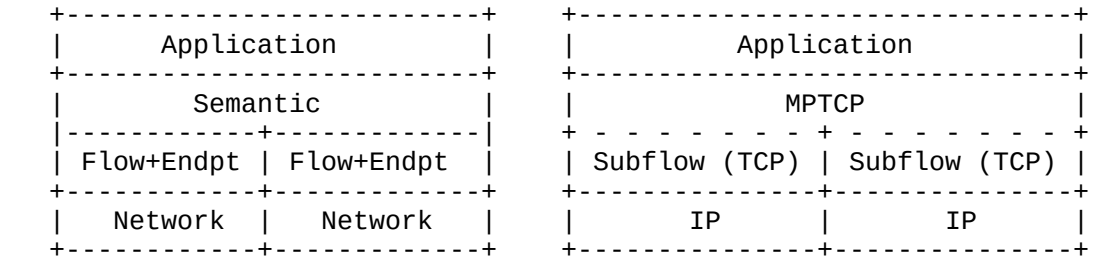

Figure 6: Relationship between Tng (left) and MPTCP (right)

As a protocol extension to TCP, MPTCP thus explicitly acknowledges middleboxes in its design, and specifies a protocol that operates at two scales: the MPTCP component operates end-to-end, while it allows the TCP component to operate segment-by-segment.

4. A Functional Decomposition of MPTCP

The previous two sections have discussed the goals for a Multipath TCP design, and provided a basis for decomposing the functions of a transport protocol in order to better understand the form a solution should take. This section builds upon this analysis by presenting the functional components that are used within the MPTCP design.

MPTCP makes use of (what appear to the network to be) standard TCP sessions, termed "subflows", to provide the underlying transport per path, and as such these retain the network compatibility desired. MPTCP-specific information is carried in a TCP-compatible manner, although this mechanism is separate from the actual information being transferred so could evolve in future revisions. Figure 7 illustrates the layered architecture.

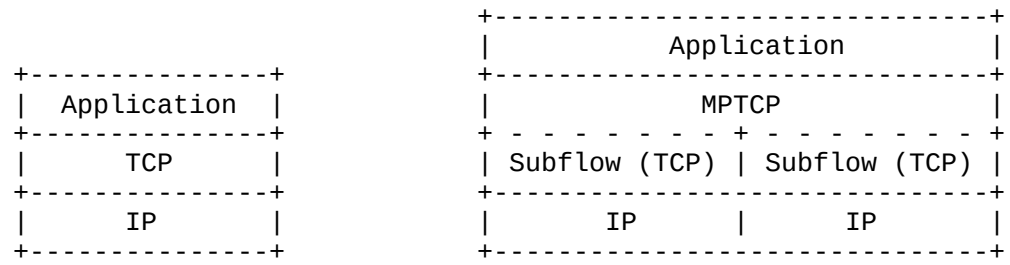

Figure 7: Comparison of Standard TCP and MPTCP Protocol Stacks

Situated below the application, the MPTCP extension in turn manages multiple TCP subflows below it. In order to do this, it must implement the following functions:

- Path Management: This is the function to detect and use multiple paths between two hosts. MPTCP uses the presence of multiple IP addresses at one or both of the hosts as an indicator of this. The path management features of the MPTCP protocol are the mechanisms to signal alternative addresses to hosts, and mechanisms to set up new subflows joined to an existing MPTCP connection.

- Packet Scheduling: This function breaks the bytestream received from the application into segments to be transmitted on one of the available subflows. The MPTCP design makes use of a data sequence mapping, associating segments sent on different subflows to a connection-level sequence numbering, thus allowing segments sent on different subflows to be correctly re-ordered at the receiver. The packet scheduler is dependent upon information about the availability of paths exposed by the path management component, and then makes use of the subflows to transmit queued segments. This function is also responsible for connection-level re-ordering on receipt of packets from the TCP subflows, according to the attached data sequence mappings.

- Subflow (single-path TCP) Interface: A subflow component takes segments from the packet-scheduling component and transmits them over the specified path, ensuring detectable delivery to the host. MPTCP uses TCP underneath for network compatibility; TCP ensures in-order, reliable delivery. TCP adds its own sequence numbers to the segments; these are used to detect and retransmit lost packets at the subflow layer. On receipt, the subflow passes its reassembled data to the packet scheduling component for connection-level reassembly; the data sequence mapping from the sender’s packet scheduling component allows re-ordering of the entire bytestream.

- o Congestion Control: This function coordinates congestion control across the subflows. As specified, this congestion control algorithm MUST ensure that a MPTCP connection does not unfairly take more bandwidth than a single path TCP flow would take at a shared bottleneck. An algorithm to support this is specified in [7].

These functions fit together as follows. The Path Management looks after the discovery (and if necessary, initialisation) of multiple paths between two hosts. The Packet Scheduler then receives a stream of data from the application destined for the network, and undertakes the necessary operations on it (such as segmenting the data into connection-level segments, and adding a connection-level sequence number) before sending it on to a subflow. The subflow then adds its own sequence number, ACKs, and passes them to network. The receiving subflow re-orders data (if necessary) and passes it to the packet scheduling component, which performs connection level re-ordering, and sends the data stream to the application. Finally, the congestion control component exists as part of the packet scheduling, in order to schedule which segments should be sent at what rate on which subflow.

5. High-Level Design Decisions

There is seemingly a wide range of choices when designing a multipath extension to TCP. However, the goals as discussed earlier in this document constrain the possible solutions, leaving relative little choice in many areas. Here, we outline high-level design choices that draw from the architectural basis discussed earlier in Section 3, which the design of MPTCP [5] takes into account.

5.1. Sequence Numbering

MPTCP uses two levels of sequence spaces: a connection level sequence number, and another sequence number for each subflow. This permits connection-level segmentation and reassembly, and retransmission of the same part of connection-level sequence space on different subflow-level sequence space.

The alternative approach would be to use a single connection level sequence number, which gets sent on multiple subflows. This has two problems: first, the individual subflows will appear to the network as TCP sessions with gaps in the sequence space; this in turn may upset certain middleboxes such as intrusion detection systems, or certain transparent proxies, and would thus go against the network compatibility goal. Second, the sender would not be able to attribute packet losses or receptions to the correct path when the

same segment is sent on multiple paths (i.e. in the case of retransmissions).

The sender must be able to tell the receiver how to reassemble the data, for delivery to the application. In order to achieve this, the receiver must determine how subflow-level data (carrying subflow sequence numbers) maps at the connection level. We refer to this as the Data Sequence Mapping. This mapping takes the form (data seq, subflow seq, length), i.e. for a given number of bytes (the length), the subflow sequence space beginning at the given sequence number maps to the connection-level sequence space (beginning at the given data seq number). This information could conceivably have various sources.

One option to signal the Data Sequence Mapping would be to use existing fields in the TCP segment (such as subflow seqno, length) and only add the data sequence number to each segment, for instance as a TCP option. This would be vulnerable, however, to middleboxes that resegment or assemble data, since there is no specified behaviour for coalescing TCP options. If one signalled (data seqno, length), this would still be vulnerable to middleboxes that coalesce segments and do not understand MPTCP signalling so do not correctly rewrite the options.

Because of these potential issues, the design decision taken in the MPTCP protocol is that whenever a mapping for subflow data needs to be conveyed to the other host, all three pieces of data (data seq, subflow seq, length) must be sent. To reduce the overhead, it would be permissible for the mapping to be sent periodically and cover more than a single segment. Further experimentation is required to determine what tradeoffs exist regarding the frequency at which mappings should be sent. It could also be excluded entirely in the case of a connection before more than one subflow is used, where the data-level and subflow-level sequence space is the same.

5.2. Reliability and Retransmissions

MPTCP features acknowledgements at connection-level as well as subflow-level acknowledgements, in order to provide a robust service to the application.

Under normal behaviour, MPTCP can use the data sequence mapping and subflow ACKs to decide when a connection-level segment was received. The transmission of TCP ACKs for a subflow are handled entirely at the subflow level, in order to maintain TCP semantics and trigger subflow-level retransmissions. This has certain implications on end-to-end semantics. It means that once a segment is ACKed at the subflow level it cannot be discarded in the re-order buffer at the

connection level. Secondly, unlike in standard TCP, a receiver cannot simply drop out-of-order segments if needed (for instance, due to memory pressure). Under certain circumstances, therefore, it may be desirable to drop segments after acknowledgement on the subflow but before delivery to the application, and this can be facilitated by a connection-level acknowledgement.

Furthermore, it is possible to conceive of some cases where connection-level acknowledgements could improve robustness. Consider a subflow traversing a transparent proxy: if the proxy ACKs a segment and then crashes, the sender will not retransmit the lost segment on another subflow, as it thinks the segment has been received. The connection grinds to a halt despite having other working subflows, and the sender would be unable to determine the cause of the problem. An example situation where this may occur would be mobility between wireless access points, each of which operates a transport-level proxy. Finally, as an optimisation, it may be feasible for a connection-level acknowledgement to be transmitted over the shortest Round-Trip Time (RTT) path, potentially reducing send buffer requirements (see Section 5.3).

Therefore, to provide a fully robust multipath TCP solution given the above constraints, MPTCP for use on the public Internet MUST feature explicit connection-level acknowledgements, in addition to subflow-level acknowledgements. A connection-level acknowledgement would only be required in order to signal when the receive window moves forward; the heuristics for using such a signal are discussed in more detail in the protocol specification [5].

Regarding retransmissions, it MUST be possible for a segments to be retransmitted on a different subflow to that on which it was originally sent. This is one of MPTCP's core goals, in order to maintain integrity during temporary or permanent subflow failure, and this is enabled by the dual sequence number space.

The scheduling of retransmissions will have significant impact on MPTCP user experience. The current MPTCP specification suggests that data outstanding on subflows that have timed out should be rescheduled for transmission on different subflows. This behaviour aims to minimize disruption when a path breaks, and uses the first timeout as indicators. More conservative versions would be to use second or third timeouts for the same segment.

Typically, fast retransmit on an individual subflow will not trigger retransmission on another subflow, although this may still be desirable in certain cases, for instance to reduce the receive buffer requirements. However, in all cases with retransmissions on different subflows, the lost segments SHOULD still be sent on the

path that lost them. This is currently believed to be necessary to maintain subflow integrity, as per the network compatibility goal. By doing this, some efficiency is lost, and it is unclear at this point what the optimal retransmit strategy is.

Large-scale experiments are therefore required in order to determine the most appropriate retransmission strategy, and recommendations will be refined once more information is available.

5.3. Buffers

To ensure in-order delivery, MPTCP must use a connection level receive buffer, where segments are placed until they are in order and can be read by the application.

In regular, single-path TCP, it is usually recommended to set the receive buffer to 2*BDP (Bandwidth-Delay Product, i.e. BDP = BW*RTT, where BW = Bandwidth and RTT = Round-Trip Time). One BDP allows supporting reordering of segments by the network. The other BDP allows the connection to continue during fast retransmit: when a segment is fast retransmitted, the receiver must be able to store incoming data during one more RTT.

For MPTCP, the story is a bit more complicated. The ultimate goal is that a subflow packet loss or subflow failure should not affect the throughput of other working subflows; the receiver should have enough buffering to store all data until the missing segment is re-transmitted and reaches the destination.

The worst case scenario would be when the subflow with the highest RTT/RTO (Round-Trip Time or Retransmission TimeOut) experiences a timeout; in that case the receiver has to buffer data from all subflows for the duration of the RTO. Thus, the smallest connection-level receive buffer that would be needed to avoid stalling with subflow failures is sum(BW_i)*RTO_max, where BW_i = Bandwidth for each subflow and RTO_max is the largest RTO across all subflows.

This is an order of magnitude more than the receive buffer required for a single connection, and is probably too expensive for practical purposes. A more sensible requirement is to avoid stalls in the absence of timeouts. Therefore, the RECOMMENDED receive buffer is 2*sum(BW_i)*RTT_max, where RTT_max is the largest RTT across all subflows. This buffer sizing ensures subflows do not stall when fast retransmit is triggered on any subflow.

The resulting buffer size should be small enough for practical use. However, there may be extreme cases where fast, high throughput paths (e.g. 100Mb/s, 10ms RTT) are used in conjunction with slow paths

(e.g. 1Mb/s, 1000ms RTT). In that case the required receive buffer would be 12.5MB, which is likely too big. In extreme cases such as this example, it may be prudent to only use some of the fastest available paths for the MPTCP connection, potentially using the slow path(s) for backup only.

Send Buffer: The RECOMMENDED send buffer is the same size as the recommended receive buffer i.e., 2*sum(BW_i)*RTT_max. This is because the sender must store locally the segments sent but unacknowledged by the connection level ACK. The send buffer size matters particularly for hosts that maintain a large number of ongoing connections. If the required send buffer is too large, a host can choose to only send data on the fast subflows, using the slow subflows only in cases of failure.

5.4. Signalling

Since MPTCP uses TCP as its subflow transport mechanism, a MPTCP connection will also begin as a single TCP connection. Nevertheless, it must signal to the peer that it supports MPTCP and wishes to use it on this connection. As such, a TCP Option will be used to transmit this information, since this is the established mechanism for indicating additional functionality on a TCP session.

In addition, further signalling is required during the operation of a MPTCP session, such as that for reassembly for multiple subflows, and for informing the other host about potential other available addresses.

The MPTCP protocol design will, however, use TCP Options for this additional signalling. This has been chosen as the mechanism most fitting in with the goals as specified in Section 2. With this mechanism, the signalling requires to operate MPTCP is transported separately from the data, allowing it to be created and processed separately from the data stream, and retaining architectural compatibility with network entities.

This decision is the consensus of the Working Group (following detailed discussions at IETF78), and the main reasons for this are as follows:

- TCP options are the traditional signalling method for TCP;
- A TCP option on a SYN is the most compatible way for an end host to signal it is MPTCP-capable;
- If connection-level ACKs are signalled in the payload then they may suffer from packet loss and may be congestion-controlled,

which may affect the data throughput in the forward direction and could lead to head-of-line blocking;

- Middleboxes, such as NAT traversal helpers, can easily parse TCP options, e. g., to rewrite addresses.

On the other hand, the main drawbacks of TCP options compared to TLV encoding in the payload are:

- There is limited space for signalling messages;
- A middlebox may, potentially, drop a packet with an unknown option;
- The transport of control information in options is not necessarily reliable.

The detailed design of MPTCP alleviates these issues as far as possible by carefully considering the size of MPTCP options, and seamlessly falling back to regular TCP on the loss of control data.

Both option and payload encoding may interfere with offloading of TCP processing to high speed network interface cards, such as segmentation, checksumming, and reassembly. For network cards supporting MPTCP, signalling in TCP options should simplify offloading due to the separate handling of MPTCP signalling and data.

5.5. Path Management

Currently, the network does not expose path diversity between pairs of IP addresses. In order to achieve path diversity from today's IP networks, in the typical case MPTCP uses multiple addresses at one or both hosts to infer different paths across the network. It is expected that these paths, whilst not necessarily entirely non-overlapping, will be sufficiently disjoint to allow multipath to achieve improved throughput and robustness. The use of multiple IP addresses is a simple mechanism that requires no additional features in the network.

Multiple different (source, destination) address pairs will thus be used as path selectors in most cases. Each path will be identified by a standard five-tuple (i.e. source address, destination address, source port, destination port, protocol), however, which can allow the extension of MPTCP to use ports as well as addresses as path selectors. This will allow hosts to use port-based load balancing with MPTCP, for example if the network routes different ports over different paths (which may be the case with technologies such as Equal Cost MultiPath (ECMP) routing [4]). It should be noted,

however, that ISPs often undertake traffic engineering in order to optimise resource utilisation within their networks, and care should be taken (by both ISPs and developers) that MPTCP using broadly similar paths does not adversely interfere with this.

For increased chance of successfully setting up additional subflows (such as when one end is behind a firewall, NAT, or other restrictive middlebox), either host SHOULD be able to add new subflows to a MPTCP connection. MPTCP MUST be able to handle paths that appear and disappear during the lifetime of a connection (for example, through the activation of an additional network interface).

The path management is a separate function from the packet scheduling, subflow interface, and congestion control functions of MPTCP, as documented in Section 4. As such it would be feasible to replace this IP-address-based design with an alternative path selection mechanism in the future, with no significant changes to the other functional components.

5.6. Connection Identification

Since a MPTCP connection may not be bound to a traditional 5-tuple (source address and port, destination address and port, protocol number) for the entirety of its existence, it is desirable to provide a new mechanism for connection identification. This will be useful for MPTCP-aware applications, and for the MPTCP implementation (and MPTCP-aware middleboxes) to have a unique identifier with which to associate the multiple subflows.

Therefore, each MPTCP connection requires a connection identifier at each host, which is locally unique within that host. In many ways, this is analogous to an ephemeral port number in regular TCP. The manifestation and purpose of such an identifier is out of the scope of this architecture document.

Legacy applications will not, however, have access to this identifier and in such cases a MPTCP connection will be identified by the 5-tuple of the first TCP subflow. It is out of the scope of this document, however, to define the behaviour of the MPTCP implementation if the first TCP subflow later fails. If there are MPTCP-unaware applications that make assumptions about continued existence of the initial address pair, their behaviour could be disrupted by carrying on regardless. It is expected that this is a very small, possibly negligible, set of applications, however. MPTCP MUST NOT be used for applications that request to bind to a specific address or interface, since such applications are making a deliberate choice of path in use.

Since the requirements of applications are not clear at this stage, however, it is as yet unconfirmed whether carrying on in the event of the loss of the initial address pair would be a damaging assumption to make. This behaviour will be an implementation-specific solution, and as such it is expected to be chosen by implementors once more research has been undertaken to determine its impact.

5.7. Congestion Control

As discussed in network-layer compatibility requirements Section 2.2.3, there are three goals for the congestion control algorithms used by a MPTCP implementation: improve throughput (at least as well as a single-path TCP connection would perform); do no harm to other network users (do not take up more capacity on any one path than if it was a single path flow using only that route - this is particularly relevant for shared bottlenecks); and balance congestion by moving traffic away from the most congested paths. To achieve these goals, the congestion control algorithms on each subflow must be coupled in some way. A proposal for a suitable congestion control algorithm is given in [7].

5.8. Security

A detailed threat analysis for Multipath TCP is presented in a separate document [12]. This focuses on flooding attacks and hijacking attacks that can be launched against a Multipath TCP connection.

The basic security goal of Multipath TCP, as introduced in Section 2.3, can be stated as: "provide a solution that is no worse than standard TCP".

From the threat analysis, and with this goal in mind, three key security requirements can be identified. A multi-addressed Multipath TCP SHOULD be able to:

- Provide a mechanism to confirm that the parties in a subflow handshake are the same as in the original connection setup (e.g. require use of a key exchanged in the initial handshake in the subflow handshake, to limit the scope for hijacking attacks).

- Provide verification that the peer can receive traffic at a new address before adding it (i.e. verify that the address belongs to the other host, to prevent flooding attacks).

- Provide replay protection, i.e. ensure that a request to add/ remove a subflow is ’fresh’.

Additional mechanisms have been deployed as part of standard TCP stacks to provide resistance to Denial-of-Service attacks. For example, there are various mechanisms to protect against TCP reset attacks [18], and Multipath TCP should continue to support similar protection. In addition, TCP SYN Cookies [19] were developed to allow a TCP server to defer the creation of session state in the SYN_RCVD state, and remain stateless until the ESTABLISHED state had been reached. Multipath TCP should, ideally, continue to provide such functionality and, at a minimum, avoid significant computational burden prior to reaching the ESTABLISHED state (of the Multipath TCP connection as a whole).

It should be noted that aspects of the Multipath TCP design space place constraints on the security solution:

- The use of TCP options significantly limits the amount of information that can be carried in the handshake.
- The need to work through middleboxes results in the need to handle mutability of packets.
- The desire to support a 'break-before-make' (as well as a 'make-before-break') approach to adding subflows (within a limited time period) implies that a host cannot rely on using a pre-existing subflow to support the addition of a new one.

The MPTCP protocol will be designed with these security requirements in mind, and the protocol specification [5] will document how these are met.

6. Software Interactions

6.1. Interactions with Applications

In the case of applications that have used an existing API call to bind to a specific address or interface, the MPTCP extension MUST NOT be used. This is because the applications are indicating a clear choice of path to use and thus will have expectations of behaviour that must be maintained, in order to adhere to the application compatibility goals.

Interactions with applications are presented in [8] - including, but not limited to, performances changes that may be expected, semantic changes, and new features that may be requested through an enhanced API.

TCP features the ability to send "Urgent" data, the delivery of which

to the application may or may not be out-of-band. The use of this feature is not recommended due to security implications and implementation differences [20]. MPTCP requires contiguous data to support its Data Sequence Mapping over multiple segments, and therefore the Urgent pointer cannot interrupt an existing mapping. An MPTCP implementation MAY choose to support sending Urgent data, and if it does, it SHOULD send the Urgent data on the soonest available unassigned subflow sequence space. Incoming Urgent data SHOULD be mapped to connection-level sequence space and delivered to the application analogous to Urgent data in regular TCP.

6.2. Interactions with Management Systems

To enable interactions between TCP and network management systems, the TCP [21] and TCP Extended Statistics (ESTATS) [22] MIBs have been defined. MPTCP should share the these MIBs for aspects that are designed to be transparent to the application.

It is anticipated that a MPTCP MIB will be defined in the future, once experience of experimental MPTCP deployments is gathered. This MIB would provide access to MPTCP-specific properties such as whether MPTCP is enabled, and the number and properties of the individual paths in use.

7. Interactions with Middleboxes

As discussed in Section 2.2, it is a goal of MPTCP to be deployable today and thus compatible with the majority of middleboxes. This section summarises the issues that may arise with NATs, firewalls, proxies, intrusion detection systems, and other middleboxes that, if not considered in the protocol design, may hinder its deployment.

This section is intended primarily as a description of options and considerations only. Protocol-specific solutions to these issues will be given in the companion documents.

Multipath TCP will be deployed in a network that no longer provides just basic datagram delivery. A myriad of middleboxes are deployed to optimize various perceived problems with the Internet protocols: NATs primarily address IP address space shortage [15], Performance Enhancing Proxies (PEPs) optimize TCP for different link characteristics [17], firewalls [16] and intrusion detection systems try to block malicious content from reaching a host, and traffic normalizers [23] ensure a consistent view of the traffic stream to Intrusion Detection Systems (IDS) and hosts.

All these middleboxes optimize current applications at the expense of

future applications. In effect, future applications will often need to behave in a similar fashion to existing ones, in order to increase the chances of successful deployment. Further, the precise behaviour of all these middleboxes is not clearly specified, and implementation errors make matters worse, raising the bar for the deployment of new technologies.

The following list of middlebox classes documents behaviour that could impact the use of MPTCP. This list is used in [5] to describe the features of the MPTCP protocol that are used to mitigate the impact of these middlebox behaviours.

- NATs: Network Address Translators decouple the host's local IP address (and, in the case of NAPTs, port) with that which is seen in the wider Internet when the packets are transmitted through a NAT. This adds complexity, and reduces the chances of success, when signalling IP addresses.

- PEPs: Performance Enhancing Proxies, which aim to improve the performance of protocols over low-performance (e.g. high latency or high error rate) links. As such, they may "split" a TCP connection and behaviour such as proactive ACKing may occur, and therefore it is no longer guaranteed that one host is communicating directly with another. PEPs, firewalls or other middleboxes may also change the declared receive window size.

- Traffic Normalizers: These aim to eliminate ambiguities and potential attacks at the network level, and amongst other things are unlikely to permit holes in TCP-level sequence space (which has impact on MPTCP's retransmission and subflow sequence numbering design choices).

- Firewalls: on top of preventing incoming connections, firewalls may also attempt additional protection such as sequence number randomization (so a sender cannot reliably know what TCP sequence number the receiver will see).

- Intrusion Detection Systems: IDSs may look for traffic patterns to protect a network, and may have false positives with MPTCP and drop the connections during normal operation. Future MPTCP-aware middleboxes will require the ability to correlate the various paths in use.

- Content-aware Firewalls: Some middleboxes may actively change data in packets, such as re-writing URIs in HTTP traffic.

In addition, all classes of middleboxes may affect TCP traffic in the following ways:

- o TCP Options: some middleboxes may drop packets with unknown TCP options, or strip those options from the packets.

- o Segmentation and Coalescing: middleboxes (or even something as close to the end host as TCP Segmentation Offloading (TSO) on a Network Interface Card (NIC)) may change the packet boundaries from those which the sender intended. It may do this by splitting packets, or coalescing them together. This leads to two major impacts: we cannot guarantee where a packet boundary will be, and we cannot say for sure what a middlebox will do with TCP options in these cases (they may be repeated, dropped, or sent only once).

8. Contributors

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10. IANA Considerations

None.

11. Security Considerations

This informational document provides an architectural overview for Multipath TCP and so does not, in itself, raise any security issues. A separate threat analysis [12] lists threats that can exist with a Multipath TCP. However, a protocol based on the architecture in this document will have a number of security requirements. The high level goals for such a protocol are identified in Section 2.3, whilst

Section 5.8 provides more detailed discussion of security requirements and design decisions which are applied in the MPTCP protocol design [5].

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Appendix A. Changelog

(For removal by the RFC Editor)

A.1. Changes since draft-ietf-mptcp-architecture-04

- Responded to IETF Last Call and IESG review comments.

A.2. Changes since draft-ietf-mptcp-architecture-03

- Responded to AD review comments.

A.3. Changes since draft-ietf-mptcp-architecture-02

- Responded to WG last call review comments. Included editorial fixes, adding Section 2.4, and improving Section 5.4 and Section 7.

A.4. Changes since draft-ietf-mptcp-architecture-01

- Responded to review comments.
- Added security sections.

A.5. Changes since draft-ietf-mptcp-architecture-00

- Added middlebox compatibility discussion (Section 7).
- Clarified path identification (TCP 4-tuple) in Section 5.5.
- Added brief scenario and diagram to Section 1.3.

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Coupled Congestion Control for Multipath Transport Protocols
draft-ietf-mptcp-congestion-07

Abstract

Often endpoints are connected by multiple paths, but communications are usually restricted to a single path per connection. Resource usage within the network would be more efficient were it possible for these multiple paths to be used concurrently. Multipath TCP is a proposal to achieve multipath transport in TCP.

New congestion control algorithms are needed for multipath transport protocols such as Multipath TCP, as single path algorithms have a series of issues in the multipath context. One of the prominent problems is that running existing algorithms such as standard TCP independently on each path would give the multipath flow more than its fair share at a bottleneck link traversed by more than one of its subflows. Further, it is desirable that a source with multiple paths available will transfer more traffic using the least congested of the paths, achieving a property called resource pooling where a bundle of links effectively behaves like one shared link with bigger-capacity. This would increase the overall efficiency of the network and also its robustness to failure.

This document presents a congestion control algorithm which couples the congestion control algorithms running on different subflows by linking their increase functions, and dynamically controls the overall aggressiveness of the multipath flow. The result is a practical algorithm that is fair to TCP at bottlenecks while moving traffic away from congested links.

Status of this Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

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This Internet-Draft will expire on January 30, 2012.

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Table of Contents

1. Requirements Language

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119 [RFC2119] .

2. Introduction

Multipath TCP (MPTCP, [I-D.ietf-mptcp-multiaddressed]) is a set of extensions to regular TCP [RFC0793] that allows one TCP connection to be spread across multiple paths. MPTCP distributes load through the creation of separate "subflows" across potentially disjoint paths.

How should congestion control be performed for multipath TCP? First, each subflow must have its own congestion control state (i.e. cwnd) so that capacity on that path is matched by offered load. The simplest way to achieve this goal is to simply run standard TCP congestion control on each subflow. However this solution is unsatisfactory as it gives the multipath flow an unfair share when the paths taken by its different subflows share a common bottleneck.

Bottleneck fairness is just one requirement multipath congestion control should meet. The following three goals capture the desirable properties of a practical multipath congestion control algorithm:

- Goal 1 (Improve Throughput) A multipath flow should perform at least as well as a single path flow would on the best of the paths available to it.

- Goal 2 (Do no harm) A multipath flow should not take up more capacity from any of the resources shared by its different paths, than if it was a single flow using only one of these paths. This guarantees it will not unduly harm other flows.

- Goal 3 (Balance congestion) A multipath flow should move as much traffic as possible off its most congested paths, subject to meeting the first two goals.

Goals 1 and 2 together ensure fairness at the bottleneck. Goal 3 captures the concept of resource pooling [WISCHIK]: if each multipath flow sends more data through its least congested path, the traffic in the network will move away from congested areas. This improves robustness and overall throughput, among other things. The way to achieve resource pooling is to effectively "couple" the congestion control loops for the different subflows.

We propose an algorithm that couples the additive increase function

of the subflows, and uses unmodified TCP behavior in case of a drop. The algorithm relies on the traditional TCP mechanisms to detect drops, to retransmit data, etc.

Detecting shared bottlenecks reliably is quite difficult, but is just one part of a bigger question. This bigger question is how much bandwidth a multipath user should use in total, even if there is no shared bottleneck.

The congestion controller aims to set the multipath flow's aggregate bandwidth to be the same as a regular TCP flow would get on the best path available to the multipath flow. To estimate the bandwidth of a regular TCP flow, the multipath flow estimates loss rates and round trip times and computes the target rate. Then it adjusts the overall aggresiveness (parameter alpha) to achieve the desired rate.

While the mechanism above applies always, its effect depends on whether the multipath TCP flow influences or does not influence the link loss rates (low vs. high statistical multiplexing). If MPTCP does not influence link loss rates, MPTCP will get the same throughput as TCP on the best path. In cases with low statistical multiplexing, where the multipath flow influences the loss rates on the path, the multipath throughput will be strictly higher than a single TCP would get on any of the paths. In particular, if using two idle paths, multipath throughput will be sum of the two paths' throughput.

This algorithm ensures bottleneck fairness and fairness in the broader, network sense. We acknowledge that current TCP fairness criteria are far from ideal, but a multipath TCP needs to be deployable in the current Internet. If needed, new fairness criteria can be implemented by the same algorithm we propose by appropriately scaling the overall aggressiveness.

It is intended that the algorithm presented here can be applied to other multipath transport protocols, such as alternative multipath extensions to TCP, or indeed any other congestion-aware transport protocols. However, for the purposes of example this document will, where appropriate, refer to the MPTCP protocol.

The design decisions and evaluation of the congestion control algorithm are published in [NSDI].

The algorithm presented here only extends standard TCP congestion control for multipath operation. It is foreseeable that other congestion controllers will be implemented for multipath transport to achieve the bandwidth-scaling properties of the newer congestion control algorithms for regular TCP (such as Compound TCP and Cubic).

3. Coupled Congestion Control Algorithm

The algorithm we present only applies to the increase phase of the congestion avoidance state specifying how the window inflates upon receiving an ack. The slow start, fast retransmit, and fast recovery algorithms, as well as the multiplicative decrease of the congestion avoidance state are the same as in standard TCP[RFC5681].

Let cwnd_i be the congestion window on the subflow i. Let tot_cwnd be the sum of the congestion windows of all subflows in the connection. Let p_i, rtt_i and mss_i be the loss rate, round trip time (i.e. smoothed round trip time estimate used by TCP) and maximum segment size on subflow i.

We assume throughout this document that the congestion window is maintained in bytes, unless otherwise specified. We briefly describe the algorithm for packet-based implementations of cwnd in section Section 4.2.

Our proposed "Linked Increases" algorithm MUST:

o For each ack received on subflow i, increase cwnd_i by

$$\min\left(\frac{alpha * bytes\_acked * mss\_i}{tot\_cwnd}, \frac{bytes\_acked * mss\_i}{cwnd\_i}\right) \quad (1)$$

The increase formula (1) takes the minimum between the computed increase for the multipath subflow (first argument to min), and the increase TCP would get in the same scenario (the second argument). In this way, we ensure that any multipath subflow cannot be more aggressive than a TCP flow in the same circumstances, hence achieving goal 2 (do no harm).

"alpha" is a parameter of the algorithm that describes the aggresiveness of the multipath flow. To meet Goal 1 (improve throughput), the value of alpha is chosen such that the aggregate throughput of the multipath flow is equal to the throughput a TCP flow would get if it ran on the best path.

To get an intuition of what the algorithm is trying to do, let’s take the case where all the subflows have the same round trip time and MSS. In this case the algorithm will grow the total window by approximately alpha*MSS per RTT. This increase is distributed to the individual flows according to their instantaneous window size. Subflow i will increase by alpha*cwnd_i/tot_cwnd segments per RTT.

Note that, as in standard TCP, when tot_cwnd is large the increase

may be 0. In this case the increase MUST be set to 1. We discuss how to implement this formula in practice in the next section.

We assume implementations use an approach similar to appropriate byte counting (ABC, [RFC3465]), where the bytes_acked variable records the number of bytes newly acknowledged. If this is not the case, bytes_acked SHOULD be set to mss_i.

To compute tot_cwnd, it is an easy mistake to sum up cwnd_i across all subflows: when a flow is in fast retransmit, its cwnd is typically inflated and no longer represents the real congestion window. The correct behavior is to use the ssthresh value for flows in fast retransmit when computing tot_cwnd. To cater for connections that are app limited, the computation should consider the minimum between flight_size_i and cwnd_i, and flight_size_i and ssthresh_i where appropriate.

The total throughput of a multipath flow depends on the value of alpha and the loss rates, maximum segment sizes and round trip times of its paths. Since we require that the total throughput is no worse than the throughput a single TCP would get on the best path, it is impossible to choose a-priori a single value of alpha that achieves the desired throughput in every occasion. Hence, alpha must be computed based on the observed properties of the paths.

The formula to compute alpha is:

$$alpha = tot\_cwnd * \frac{\max_i \frac{cwnd\_i}{rtt\_i^2}}{\left(\sum_i \frac{cwnd\_i}{rtt\_i}\right)^2} \quad (2)$$

The formula (2) is derived by equalizing the rate of the multipath flow with the rate of a TCP running on the best path, and solving for alpha.

4. Implementation Considerations

Equation (2) implies that alpha is a floating point value. This would require performing costly floating point operations whenever an ACK is received, Further, in many kernels floating point operations are disabled. There is an easy way to approximate the above

calculations using integer arithmetic.

4.1. Computing alpha in Practice

Let alpha_scale be an integer. When computing alpha, use alpha_scale * tot_cwnd instead of tot_cwnd, and do all the operations in integer arithmetic.

Then, scale down the increase per ack by alpha_scale. The resulting algorithm is a simple change from Equation (1):

o For each ack received on subflow i, increase cwnd_i by

$$\min\left(\frac{alpha * bytes\_acked * mss\_i}{alpha\_scale * tot\_cwnd}, \frac{bytes\_acked * mss\_i}{cwnd\_i}\right) \quad (3)$$

The alpha_scale parameter denotes the precision we want for computing alpha. Observe that the errors in computing the numerator or the denominator in the formula for alpha are quite small, as the cwnd in bytes is typically much larger than the RTT (measured in ms).

With these changes, all the operations can be done using integer arithmetic. We propose alpha_scale be a small power of two, to allow using faster shift operations instead of multiplication and division. Our experiments show that using alpha_scale=512 works well in a wide range of scenarios. Increasing alpha_scale increases precision, but also increases the risk of overflow when computing alpha. Using 64bit operations would solve this issue. Another option is to dynamically adjust alpha_scale when computing alpha; in this way we avoid overflow and obtain maximum precision.

It is possible to implement the algorithm by calculating tot_cwnd on each ack, however this would be costly especially when the number of subflows is large. To avoid this overhead the implementation MAY choose to maintain a new per connection state variable called tot_cwnd. If it does so, the implementation will update tot_cwnd value whenever the individual subflows' windows are updated. Updating only requires one more addition or subtraction operation compared to the regular, per subflow congestion control code, so its performance impact should be minimal.

Computing alpha per ack is also costly. We propose alpha to be a per connection variable, computed whenever there is a drop and once per RTT otherwise. More specifically, let cwnd_new be the new value of the congestion window after it is inflated or after a drop. Update alpha only if the quotient of cwnd_i/mss_i differs from the quotient of cwnd_new_i/mss_i.

In certain cases with small RTTs, computing alpha can still be expensive. We observe that if RTTs were constant, it is sufficient to compute alpha once per drop, as alpha does not change between drops (the insight here is that cwnd_i/cwnd_j = constant as long as both windows increase). Experimental results show that even if round trip times are not constant, using average round trip time per sawtooth instead of instantaneous round trip time (i.e. TCP's smoothed RTT estimator) gives good precision for computing alpha. Hence, it is possible to compute alpha only once per drop using a modified version of equation (2) where rtt_i is replaced with rtt_avg_i.

If using average round trip time, rtt_avg_i will be computed by sampling the rtt_i whenever the window can accommodate one more packet, i.e. when cwnd / mss < (cwnd+increase)/mss. The samples are averaged once per sawtooth into rtt_avg_i. This sampling ensures that there is no sampling bias for larger windows.

Given tot_cwnd and alpha, the congestion control algorithm is run for each subflow independently, with similar complexity to the standard TCP increase code [RFC5681].

4.2. Implementation Considerations when CWND is Expressed in Packets

When the congestion control algorithm maintains cwnd in packets rather than bytes, the algorithms above must change to take into account path mss.

To compute the increase when an ack is received, the implementation for multipath congestion control is a simple extension of the standard TCP code. In standard TCP cwnd_cnt is an additional state variable that tracks the number of segments acked since the last cwnd increment; cwnd is incremented only when cwnd_cnt > cwnd; then cwnd_cnt is set to 0.

In the multipath case, cwnd_cnt_i is maintained for each subflow as above, and cwnd_i is increased by 1 when cwnd_cnt_i > max(alpha_scale * tot_cwnd / alpha, cwnd_i).

When computing alpha for packet-based stacks, the errors in computing the terms in the denominator are larger (this is because cwnd is much smaller and rtt may be comparatively large). Let max be the index of the subflow used in the numerator. To reduce errors, it is easiest to move rtt_max (once calculated) from the numerator to the denominator, changing equation (2) to obtain the equivalent formula below.

$$alpha = alpha\_scale * tot\_cwnd * \frac{cwnd\_max}{\left(\sum_i \frac{rtt\_max * cwnd\_i}{rtt\_i} \right)^2} \quad (4)$$

Note that the calculation of alpha does not take into account path mss, and is the same for stacks that keep cwnd in bytes or packets. With this formula, the algorithm for computing alpha will match the rate of TCP on the best path in B/s for byte-oriented stacks, and in packets/s in packet-based stacks. In practice, mss rarely changes between paths so this shouldn't be a problem.

However, it is simple to derive formulae allowing packet-based stacks to achieve byte rate fairness (and viceversa) if needed. In particular, for packet-based stacks wanting byte-rate fairness, equation (4) above changes as follows: cwnd_max is replaced by cwnd_max * mss_max * mss_max, while cwnd_i is replaced with cwnd_i * mss_i.

5. Discussion

The algorithm we've presented fully achieves Goals 1 and 2, but does not achieve full resource pooling (Goal 3). Resource pooling requires that no traffic should be transferred on links with higher loss rates. To achieve perfect resource pooling, one must couple both increase and decrease of congestion windows across subflows, as in [KELLY].

There are a few problems with such a fully-coupled controller. First, it will probe insufficiently paths with high loss rates, and will fail to detect free capacity when it becomes available. Second, such controllers tend to exhibit "flappiness": when the paths have similar levels of congestion, the congestion controller will tend to allocate all the window to one random subflow, and allocate zero window to the other subflows. The controller will perform random flips between these stable points. This doesn't seem desirable in general, and is particularly bad when the achieved rates depend on the RTT (as in the current Internet): in such a case, the resulting rate with fluctuate unpredictably depending on which state the controller is in, hence violating Goal 1.

By only coupling increases our proposal probes high-loss paths, detecting free capacity quicker. Our proposal does not suffer from flappiness but also achieves less resource pooling. The algorithm will allocate window to the subflows such that p_i * cwnd_i =

constant, for all i. Thus, when the loss rates of the subflows are equal, each subflow will get an equal window, removing flappiness. When the loss rates differ, progressively more window will be allocated to the flow with the lower loss rate. In contrast, perfect resource pooling requires that all the window should be allocated on the path with the lowest loss rate. Further details can be found in [NSDI].

6. Security Considerations

One security concern relates to what we call the traffic-shifting attack: on-path attackers can drop packets belonging to a multipath subflow, which in turn makes the path seem congested and will force the sender's congestion controller to avoid that path and push more data over alternate subflows.

The attacker's goal is to create congestion on the corresponding alternative paths. This behaviour is entirely feasible, but will only have minor effects: by design, the coupled congestion controller is less (or similarly) aggressive on any of its paths than a single TCP flow. Thus, the biggest effect this attack can have is to make a multipath subflow be as aggressive as a single TCP flow.

Another effect of the traffic-shifting attack is that the new path can monitor all the traffic, whereas before it could only see a subset of traffic. We believe that if privacy is needed, splitting traffic across multiple paths with MPTCP is not the right solution in the first place; end-to-end encryption should be used instead.

Besides the traffic-shifting attack mentioned above, the coupled congestion control algorithm defined in this draft adds no other security considerations to those found in [I-D.ietf-mptcp-multiaddressed] and [RFC6181]. Detailed security analysis for the Multipath TCP protocol itself is included in [I-D.ietf-mptcp-multiaddressed] and [RFC6181].

7. Acknowledgements

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8. IANA Considerations

This document does not require any action from IANA.

9. References

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TCP Extensions for Multipath Operation with Multiple Addresses
draft-ietf-mptcp-multiaddressed-12

Abstract

TCP/IP communication is currently restricted to a single path per connection, yet multiple paths often exist between peers. The simultaneous use of these multiple paths for a TCP/IP session would improve resource usage within the network, and thus improve user experience through higher throughput and improved resilience to network failure.

Multipath TCP provides the ability to simultaneously use multiple paths between peers. This document presents a set of extensions to traditional TCP to support multipath operation. The protocol offers the same type of service to applications as TCP (i.e. reliable bytestream), and provides the components necessary to establish and use multiple TCP flows across potentially disjoint paths.

Table of Contents

1. Introduction

MPTCP is a set of extensions to regular TCP [1] to provide a Multipath TCP [2] service, which enables a transport connection to operate across multiple paths simultaneously. This document presents the protocol changes required to add multipath capability to TCP; specifically, those for signaling and setting up multiple paths ("subflows"), managing these subflows, reassembly of data, and termination of sessions. This is not the only information required to create a Multipath TCP implementation, however. This document is complemented by three others:

- Architecture [2], which explains the motivations behind Multipath TCP, contains a discussion of high-level design decisions on which this design is based, and an explanation of a functional separation through which an extensible MPTCP implementation can be developed.

- Congestion Control [5], presenting a safe congestion control algorithm for coupling the behaviour of the multiple paths in order to "do no harm" to other network users.

- Application Considerations [6], discussing what impact MPTCP will have on applications, what applications will want to do with MPTCP, and as a consequence of these factors, what API extensions an MPTCP implementation should present.

1.1. Design Assumptions

In order to limit the potentially huge design space, the working group imposed two key constraints on the multipath TCP design presented in this document:

- It must be backwards-compatible with current, regular TCP, to increase its chances of deployment

- It can be assumed that one or both hosts are multihomed and multiaddressed

To simplify the design we assume that the presence of multiple addresses at a host is sufficient to indicate the existence of multiple paths. These paths need not be entirely disjoint: they may share one or many routers between them. Even in such a situation making use of multiple paths is beneficial, improving resource utilisation and resilience to a subset of node failures. The congestion control algorithms defined in [5] ensure this does not act detrimentally. Furthermore, there may be some scenarios where different TCP ports on a single host can provide disjoint paths (such

as through certain ECMP implementations [7]), and so the MPTCP design also supports the use of ports in path identifiers.

There are three aspects to the backwards-compatibility listed above (discussed in more detail in [2]):

External Constraints: The protocol must function through the vast majority of existing middleboxes such as NATs, firewalls and proxies, and as such must resemble existing TCP as far as possible on the wire. Furthermore, the protocol must not assume the segments it sends on the wire arrive unmodified at the destination: they may be split or coalesced; TCP options may be removed or duplicated.

Application Constraints: The protocol must be usable with no change to existing applications that use the common TCP API (although it is reasonable that not all features would be available to such legacy applications). Furthermore, the protocol must provide the same service model as regular TCP to the application.

Fall-back: The protocol should be able to fall back to standard TCP with no interference from the user, to be able to communicate with legacy hosts.

The complementary application considerations document [6] discusses the necessary features of an API to provide backwards-compatibility, as well as API extensions to convey the behaviour of MPTCP at a level of control and information equivalent to that available with regular, single-path TCP.

Further discussion of the design constraints and associated design decisions are given in the MPTCP Architecture document [2].

1.2. Multipath TCP in the Networking Stack

MPTCP operates at the transport layer and aims to be transparent to both higher and lower layers. It is a set of additional features on top of standard TCP; Figure 1 illustrates this layering. MPTCP is designed to be usable by legacy applications with no changes; detailed discussion of its interactions with applications is given in [6].

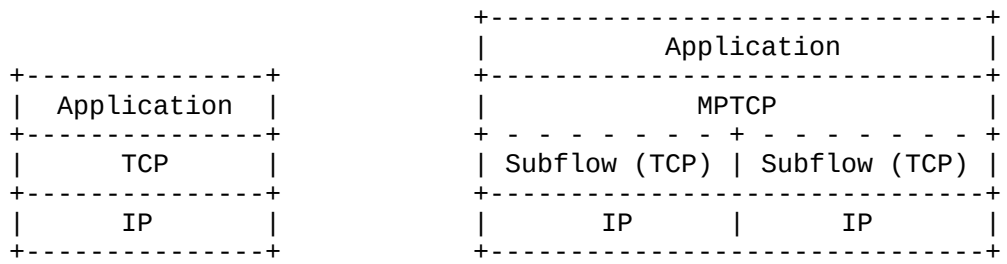


Figure 1: Comparison of Standard TCP and MPTCP Protocol Stacks

1.3. Terminology

This document makes use of a number of terms which are either MPTCP-specific, or have defined meaning in the context of MPTCP, as follows:

Path: A sequence of links between a sender and a receiver, defined in this context by a 4-tuple of source and destination address/port pairs.

Subflow: A flow of TCP segments operating over an individual path, which forms part of a larger MPTCP connection. A subflow is started and terminated similarly to a regular TCP connection.

(MPTCP) Connection: A set of one or more subflows, over which an application can communicate between two hosts. There is a one-to-one mapping between a connection and an application socket.

Data-level: The payload data is nominally transferred over a connection, which in turn is transported over subflows. Thus the term "data-level" is synonymous with "connection level", in contrast to "subflow-level" which refers to properties of an individual subflow.

Token: A locally unique identifier given to a multipath connection by a host. May also be referred to as a "Connection ID".

Host: A end host operating an MPTCP implementation, and either initiating or accepting an MPTCP connection.

In addition to these terms, note that MPTCP's interpretation of, and effect on, regular single-path TCP semantics are discussed in Section 4.

1.4. MPTCP Concept

This section provides a high-level summary of normal operation of MPTCP, and is illustrated by the scenario shown in Figure 2. A detailed description of operation is given in Section 3.

- To a non-MPTCP-aware application, MPTCP will behave the same as normal TCP. Extended APIs could provide additional control to MPTCP-aware applications [6]. An application begins by opening a TCP socket in the normal way. MPTCP signaling and operation is handled by the MPTCP implementation.

- An MPTCP connection begins similarly to a regular TCP connection. This is illustrated in Figure 2 where an MPTCP connection is established between addresses A1 and B1 on Hosts A and B respectively.

- If extra paths are available, additional TCP sessions (termed MPTCP "subflows") are created on these paths, and are combined with the existing session, which continues to appear as a single connection to the applications at both ends. The creation of the additional TCP session is illustrated between Address A2 on Host A and Address B1 on Host B.

- MPTCP identifies multiple paths by the presence of multiple addresses at hosts. Combinations of these multiple addresses equate to the additional paths. In the example, other potential paths that could be set up are A1<->B2 and A2<->B2. Although this additional session is shown as being initiated from A2, it could equally have been initiated from B1.

- The discovery and setup of additional subflows will be achieved through a path management method; this document describes a mechanism by which a host can initiate new subflows by using its own additional addresses, or by signaling its available addresses to the other host.

- MPTCP adds connection-level sequence numbers to allow the reassembly of segments arriving on multiple subflows with differing network delays.

- Subflows are terminated as regular TCP connections, with a four way FIN handshake. The MPTCP connection is terminated by a connection-level FIN.

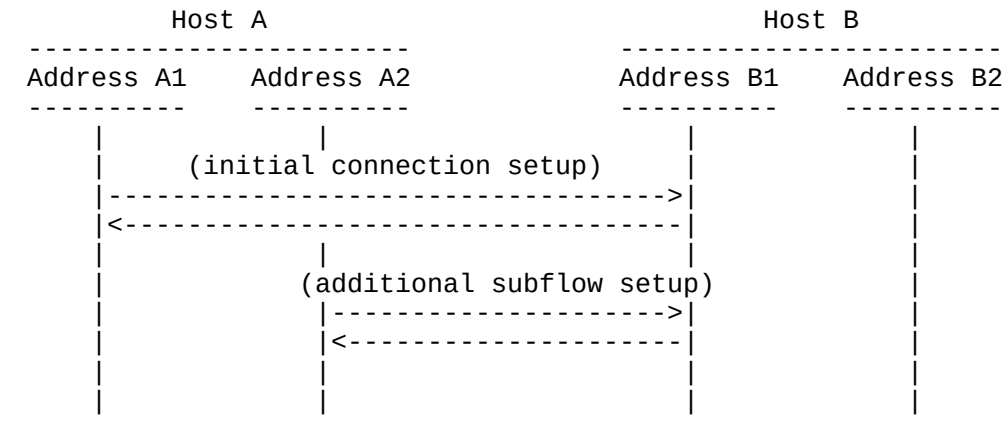


Figure 2: Example MPTCP Usage Scenario

1.5. Requirements Language

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119 [3].

2. Operation Overview

This section presents a single description of common MPTCP operation, with reference to the protocol operation. This is a high-level overview of the key functions; the full specification follows in Section 3. Extensibility and negotiated features are not discussed here. Considerable reference is made to symbolic names of MPTCP options throughout this section - these are subtypes of the IANA-assigned MPTCP option (see Section 8), and their formats are defined in the detailed protocol specification which follows in Section 3.

A Multipath TCP connection provides a bidirectionnal bytestream between two hosts communicating like normal TCP and thus does not require any change to the applications. However, Multipath TCP enables the hosts to use different paths with different IP addresses to exchange packets belonging to the MPTCP connection. A Multipath TCP connection appears like a normal TCP connection to an application. However, to the network layer each MPTCP subflows looks like a regular TCP flow whose segments carry a new TCP option type. Multipath TCP manages the creation, removal and utilization of these subflows to send data. The number of subflows that are managed within a Multipath TCP connection is not fixed and it can fluctuate during the lifetime of the Multipath TCP connection.

All MPTCP operations are signaled with a TCP option - a single numerical type for MPTCP, with "sub-types" for each MPTCP message. What follows is a summary of the purpose and rationale of these messages.

2.1. Initiating an MPTCP connection

This is the same signaling as for initiating a normal TCP connection, but the SYN, SYN/ACK and ACK packets also carry the MP_CAPABLE option. This is variable-length and serves multiple purposes. Firstly, it verifies whether the remote host supports Multipath TCP; and secondly, this option allows the hosts to exchange some information to authenticate the establishment of additional subflows. Further details are given in Section 3.1.

```
Host-A                                  Host-B
------                                  ------
MP_CAPABLE              ->
[A's key, flags]
                        <-              MP_CAPABLE
                                        [B's key, flags]
ACK + MP_CAPABLE        ->
[A's key, B's key, flags]
```

2.2. Associating a new subflow with an existing MPTCP connection

The exchange of keys in the MP_CAPABLE handshake provides material that can be used to authenticate the endpoints when new subflows will be setup. Additional subflows begin in the same way as initiating a normal TCP connection, but the SYN, SYN/ACK and ACK packets also carry the MP_JOIN option.

Host-A initiates a new subflow between one of its addresses and one of Host-B's addresses. The token - generated from the key - is used to identify which MPTCP connection it is joining, and the HMAC is used for authentication. The HMAC uses the keys exchanged in the MP_CAPABLE handshake, and the random numbers (nonces) exchanged in these MP_JOIN options. MP_JOIN also contains flags and an Address ID that can be used to refer to the source address without the sender needing to know if it has been changed by a NAT. Further details in Section 3.2.

```
   Host-A                                  Host-B
   ------                                  ------
   MP_JOIN                 ->
   [B's token, A's nonce,
    A's Address ID, flags]
                           <-              MP_JOIN
                                           [B's HMAC, B's nonce,
                                            B's Address ID, flags]
   ACK + MP_JOIN           ->
   [A's HMAC]

                           <-              ACK
```

2.3. Informing the other Host about another potential address

The set of IP addresses associated to a multihomed host may change during the lifetime of an MPTCP connection. MPTCP supports the addition and removal of addresses on a host both implicitly and explicitly. If Host-A has established a subflow starting at address IP#-A1 and wants to open a second subflow starting at address IP#-A2, it simply initiates the establishment of the subflow as explained above. The remote host will then be implicitly informed about the new address.

In some circumstances, a host may want to advertise to the remote host the availability of an address without establishing a new subflow, for example when a NAT prevents setup in one direction. In the example below, Host-A informs Host-B about its alternative IP address (IP#-A2). Host-B may later send an MP_JOIN to this new address. Due to the presence of middleboxes that may translate IP addresses, this option uses an address identifier to unambiguously identify an address on a host. Further details in Section 3.4.1.

```
   Host-A                                  Host-B
   ------                                  ------
   ADD_ADDR                ->
   [IP#-A2,
    IP#-A2's Address ID]
```

There is a corresponding signal for address removal, making use of the Address ID that is signalled in the add address handshake. Further details in Section 3.4.2.

```
   Host-A                                  Host-B
   ------                                  ------
   REMOVE_ADDR             ->
   [IP#-A2's Address ID]
```

2.4. Data transfer using MPTCP

To ensure reliable, in-order delivery of data over subflows that may appear and disappear at any time, MPTCP uses a 64-bit Data Sequence Number (DSN) to number all data sent over the MPTCP connection. Each subflow has its own 32 bits sequence number space and an MPTCP option maps the subflow sequence space to the data sequence space. In this way, data can be retransmitted on different subflows (mapped to the same DSN) in the event of failure.

The "Data Sequence Signal" carries the "Data Sequence Mapping". The Data Sequence Mapping consists of the subflow sequence number, data sequence number, and length for which this mapping is valid. This option can also carry a connection-level acknowledgement (the "Data ACK") for the received DSN.

With MPTCP, all subflows share the same receive buffer and advertise the same receive window. There are two levels of acknowledgement in MPTCP. Regular TCP acknowledgments are used on each subflow to acknowledge the reception of the segments sent over the subflow independently of their DSN. In addition, there are connection-level acknowledgments for the data sequence space. These acknowledgments track the advancement of the bytestream and slide the receiving window.

Further details are in Section 3.3.

```
Host-A                                  Host-B
------                                  ------
DATA_SEQUENCE_SIGNAL        ->
[Data Sequence Mapping]
[Data ACK]
[Checksum]
```

2.5. Requesting a change in a path’s priority

Hosts can indicate at initial subflow setup whether they wish the subflow to be used as a regular or backup path - a backup path being only used if there are no regular paths available. During a connection, Host-A can request a change in the priority of a subflow through the MP_PRIO signal to Host-B. Further details in Section 3.3.8.

```
Host-A                                  Host-B
------                                  ------
MP_PRIO                     ->
```

2.6. Closing an MPTCP connection

When Host-A wants to inform Host-B that it has no more data to send, it signals this "Data FIN" as part of the Data Sequence Signal (see above). It has the same semantics and behaviour as a regular TCP FIN, but at the connection level. Once all the data on the MPTCP connection has been successfully received, then this message is acknowledged at the connection level with a DATA_ACK. Further details in Section 3.3.3.

```
Host-A                                  Host-B
------                                  ------
DATA_SEQUENCE_SIGNAL      ->
[Data FIN]

                          <-            (MPTCP DATA_ACK)
```

2.7. Notable features

It is worth highlighting that MPTCP's signaling has been designed with several key requirements in mind:

- To cope with NATs on the path, addresses are referred to by Address IDs, in case the IP packet's source address gets changed by a NAT. Setting up a new TCP flow is not possible if the passive opener is behind a NAT; to allow subflows to be created when either end is behind a NAT, MPTCP uses the ADD_ADDR message.

- MPTCP falls back to ordinary TCP if MPTCP operation is not possible. For example if one host is not MPTCP capable, or if a middlebox alters the payload.

- To meet the threats identified in [8], the following steps are taken: keys are sent in the clear in the MP_CAPABLE messages; MP_JOIN messages are secured with HMAC-SHA1 ([9], [4]) using those keys; and standard TCP validity checks are made on the other messages (ensuring sequence numbers are in-window).

3. MPTCP Protocol

This section describes the operation of the MPTCP protocol, and is subdivided into sections for each key part of the protocol operation.

All MPTCP operations are signalled using optional TCP header fields. A single TCP option number ("Kind") will be assigned by IANA for MPTCP (see Section 8), and then individual messages will be determined by a "sub-type", the values of which will also be stored

in an IANA registry (and are also listed in Section 8).

Throughout this document, when reference is made to an MPTCP option by symbolic name, such as "MP_CAPABLE", this refers to a TCP option with the single MPTCP option type, and with the sub-type value of the symbolic name as defined in Section 8. This sub-type is a four-bit field - the first four bits of the option payload, as shown in Figure 3. The MPTCP messages are defined in the following sections.

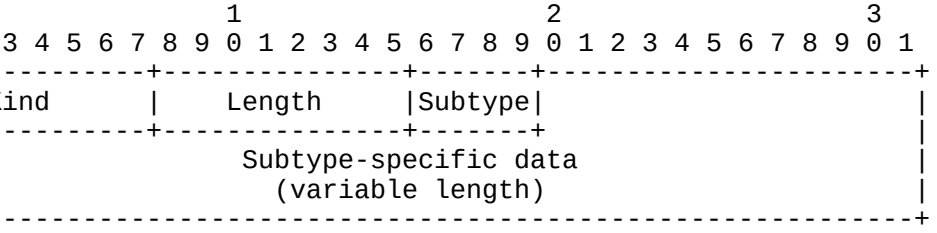


Figure 3: MPTCP option format

Those MPTCP options associated with subflow initiation are used on packets with the SYN flag set. Additionally, there is one MPTCP option for signaling metadata to ensure segmented data can be recombined for delivery to the application.

The remaining options, however, are signals that do not need to be on a specific packet, such as those for signaling additional addresses. Whilst an implementation may desire to send MPTCP options as soon as possible, it may not be possible to combine all desired options (both those for MPTCP and for regular TCP, such as SACK [10]) on a single packet. Therefore, an implementation may choose to send duplicate ACKs containing the additional signaling information. This changes the semantics of a duplicate ACK, these are usually only sent as a signal of a lost segment [11] in regular TCP. Therefore, an MPTCP implementation receiving a duplicate ACK which contains an MPTCP option MUST NOT treat it as a signal of congestion. Additionally, an MPTCP implementation SHOULD NOT send more than two duplicate ACKs in a row for the purposes of sending MPTCP options alone, in order to ensure no middleboxes misinterpret this as a sign of congestion.

Furthermore, standard TCP validity checks (such as ensuring the Sequence Number and Acknowledgement Number are within window) MUST be undertaken before processing any MPTCP signals, as described in [12].

3.1. Connection Initiation

Connection Initiation begins with a SYN, SYN/ACK, ACK exchange on a single path. Each packet contains the Multipath Capable (MP_CAPABLE)

TCP option (Figure 4). This option declares its sender is capable of performing multipath TCP and wishes to do so on this particular connection.

This option is used to declare the 64 bit key which the sender has generated for this MPTCP connection. This key is used to authenticate the addition of future subflows to this connection. This is the only time the key will be sent in clear on the wire (unless "fast close", Section 3.5, is used); all future subflows will identify the connection using a 32 bit "token". This token is a cryptographic hash of this key. The algorithm for this process is dependent on the authentication algorithm selected; the method of selection is defined later in this section.

This key is generated by its sender, and its method of generation is implementation-specific. The key MUST be hard to guess, and it MUST be unique for the sending host at any one time. Recommendations for generating random numbers for use in keys are given in [13]. Connections will be indexed at each host by the token (a one-way hash of the key). Therefore, an implementation will require a mapping from each token to the corresponding connection, and in turn to the keys for the connection.

There is a risk that two different keys will hash to the same token. The risk of hash collisions is usually small, unless the host is handling many tens of thousands of connections. Therefore, an implementation SHOULD check its list of connection tokens to ensure there is not a collision before sending its key in the SYN/ACK. This would, however, be costly for a server with thousands of connections. The subflow handshake mechanism (Section 3.2) will ensure that new subflows only join the correct connection, however, through the cryptographic handshake, as well as checking the connection tokens in both directions, and ensuring sequence numbers are in-window, so in the worst case if there was a token collision, the new subflow would not succeed, but the MPTCP connection would continue to provide a regular TCP service.

The MP_CAPABLE option is carried on the SYN, SYN/ACK, and ACK packets that start the first subflow of an MPTCP connection. The data carried by each packet is as follows, where A = initiator and B = listener.

- SYN (A->B): A's Key for this connection.
- SYN/ACK (B->A): B's Key for this connection.
- ACK (A->B): A's Key followed by B's Key.

The contents of the option is determined by the SYN and ACK flags of the packet, verified by the option's length field. For the diagram shown in Figure 4, "sender" and "receiver" refer to the sender or receiver of the TCP packet (which can be either host). If the SYN flag is set, a single key is included; if only an ACK flag is set, both keys are present.

B's Key is echoed in the ACK in order to allow the listener (host B) to act statelessly until the TCP connection reaches the ESTABLISHED state. If the listener acts in this way, however, it MUST generate its key in a way that would allow it to verify that it generated the key when it is echoed in the ACK.

This exchange allows the safe passage of MPTCP options on SYN packets to be determined. If any of these options are dropped, MPTCP will gracefully fall back to regular single-path TCP, as documented in Section 3.6. Note that new subflows MUST NOT be established (using the process documented in Section 3.2) until a DSS option has been successfully received across the path (as documented in Section 3.3).

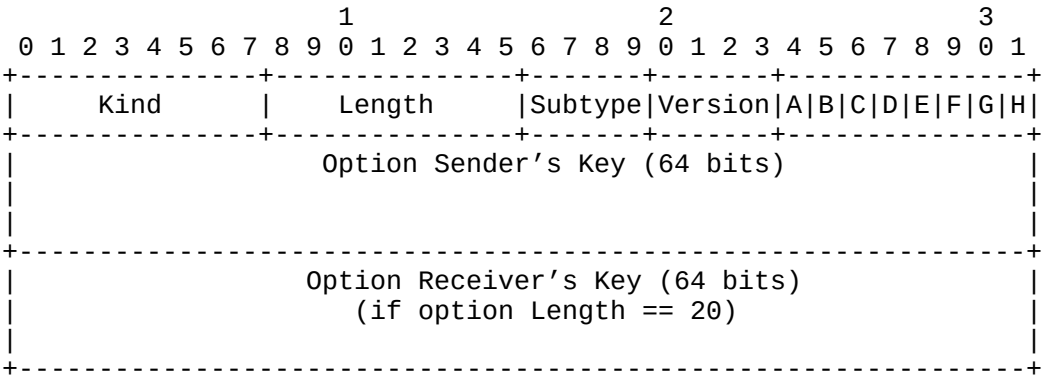


Figure 4: Multipath Capable (MP_CAPABLE) option

The first four bits of the first octet in the MP_CAPABLE option (Figure 4) define the MPTCP option subtype (see Section 8; for MP_CAPABLE, this is 0), and the remaining four bits of this octet specifies the MPTCP version in use (for this specification, this is 0).

The second octet is reserved for flags, allocated as follows:

A: The leftmost bit, labelled "A", SHOULD be set to 1 to indicate "Checksum Required", unless the system administrator has decided that checksums are not required (for example, if the environment is controlled and no middleboxes exist that might adjust the payload).

B: The second bit, labelled "B", is an extensibility flag, and MUST be set to 0 for current implementations. This will be used for an extensibility mechanism in a future specification, and the impact of this flag will be defined at a later date. If receiving a message with the "B" flag set to 1, and this is not understood, then this SYN MUST be silently ignored; the sender is expected to retry with a format compatible with this legacy specification. Note that the length of the MP_CAPABLE option, and the meanings of bits "C" through "H", may be altered by setting B=1.

C through H: The remaining bits, labelled "C" through "H", are used for crypto algorithm negotiation. Currently only the rightmost bit, labelled "H", is assigned. Bit "H" indicates the use of HMAC-SHA1 (as defined in Section 3.2). An implementation that only supports this method MUST set bit "H" to 1, and bits "C" through "G" to 0.

A crypto algorithm MUST be specified. If flag bits C through H are all 0, the MP_CAPABLE option MUST be treated as invalid and ignored (that is, it must be treated as a regular TCP handshake).

The selection of the authentication algorithm also impacts the algorithm used to generate the token and the Initial Data Sequence Number. In this specification, with only the SHA-1 algorithm (bit "H") specified and selected, the token MUST be a truncated (most significant 32 bits) SHA-1 hash ([4], [14]) of the key. A different, 64 bit truncation (the least significant 64 bits) of the SHA-1 hash of the key MUST be used as the Initial Data Sequence Number. Note that the key MUST be hashed in network byte order. Also note that the "least significant" bits MUST be the rightmost bits of the SHA-1 digest, as per [4]. Future specifications of the use of the crypto bits may choose to specify different algorithms for token and IDSN generation.

Both the crypto and checksum bits negotiate capabilities in similar ways. For the Checksum Required bit (labelled "A"), if either host requires the use of checksums, checksums MUST be used. In other words, the only way for checksums not to be used is if both hosts in their SYNs set A=0. This decision is confirmed by the setting of the "A" bit in the third packet (the ACK) of the handshake. For example, if the initiator sets A=0 in the SYN, but the responder sets A=1 in the SYN/ACK, checksums MUST be used in both directions, and the

initiator will set A=1 in the ACK. The decision whether to use checksums will be stored by an implementation in a per-connection binary state variable.

For crypto negotiation, the responder has the choice. The initiator creates a proposal setting a bit for each algorithm it supports to 1 (in this version of the specification, there is only one proposal, so bit "H" will be always set to 1). The responder responds with only one bit set - this is the chosen algorithm. The rationale for this behaviour is that the responder will typically be a server with potentially many thousands of connections, so it may wish to choose an algorithm with minimal computational complexity, depending on the load. If a responder does not support (or does not want to support) any of the initiator's proposals, it can respond without an MP_CAPABLE option, thus forcing a fall-back to regular TCP.

The MP_CAPABLE option is only used in the first subflow of a connection, in order to identify the connection; all following subflows will use the "Join" option (see Section 3.2) to join the existing connection.

If a SYN contains an MP_CAPABLE option but the SYN/ACK does not, it is assumed that the passive opener is not multipath capable and thus the MPTCP session MUST operate as a regular, single-path TCP. If a SYN does not contain a MP_CAPABLE option, the SYN/ACK MUST NOT contain one in response. If the third packet (the ACK) does not contain the MP_CAPABLE option, then the session MUST fall back to operating as a regular, single-path TCP. This is to maintain compatibility with middleboxes on the path that drop some or all TCP options. Note that an implementation MAY choose to attempt sending MPTCP options more than one time before making this decision to operate as regular TCP (see Section 3.8).

If the SYN packets are unacknowledged, it is up to local policy to decide how to respond. It is expected that a sender will eventually fall back to single-path TCP (i.e. without the MP_CAPABLE Option) in order to work around middleboxes that may drop packets with unknown options; however, the number of multipath-capable attempts that are made first will be up to local policy. It is possible that MPTCP and non-MPTCP SYNs could get re-ordered in the network. Therefore, the final state is inferred from the presence or absence of the MP_CAPABLE option in the third packet of the TCP handshake. If this option is not present, the connection SHOULD fall back to regular TCP, as documented in Section 3.6.

The initial Data Sequence Number (IDSN) on a MPTCP connection is generated from the Key. The algorithm for IDSN generation is also determined from the negotiated authentication algorithm. In this

specification, with only the SHA-1 algorithm specified and selected, the IDSN of a host MUST be the least significant 64 bits of the SHA-1 hash of its key, i.e. IDSN-A = Hash(Key-A) and IDSN-B = Hash(Key-B). This deterministic generation of the IDSN allows a receiver to ensure that there are no gaps in sequence space at the start of the connection. The SYN with MP_CAPABLE occupies the first octet of Data Sequence Space, although this does not need to be acknowledged at the connection level until the first data is sent (see Section 3.3).

3.2. Starting a New Subflow

Once an MPTCP connection has begun with the MP_CAPABLE exchange, further subflows can be added to the connection. Hosts have knowledge of their own address(es), and can become aware of the other host's addresses through signaling exchanges as described in Section 3.4. Using this knowledge, a host can initiate a new subflow over a currently unused pair of addresses. It is permitted for either host in a connection to initiate the creation of a new subflow, but it is expected that this will normally be the original connection initiator (see Section 3.8 for heuristics).

A new subflow is started as a normal TCP SYN/ACK exchange. The Join Connection (MP_JOIN) TCP option is used to identify the connection to be joined by the new subflow. It uses keying material that was exchanged in the initial MP_CAPABLE handshake (Section 3.1), and that handshake also negotiates the crypto algorithm in use for the MP_JOIN handshake.

This section specifies the behaviour of MP_JOIN using the HMAC-SHA1 algorithm. An MP_JOIN option is present in the SYN, SYN/ACK and ACK of the three-way handshake, although in each case with a different format.

In the first MP_JOIN on the SYN packet, illustrated in Figure 5, the initiator sends a token, random number, and address ID.

The token is used to identify the MPTCP connection and is a cryptographic hash of the receiver's key, as exchanged in the initial MP_CAPABLE handshake (Section 3.1). In this specification, the tokens presented in this option are generated by the SHA-1 ([4], [14]) algorithm, truncated to the most significant 32 bits. The token included in the MP_JOIN option is the token that the receiver of the packet uses to identify this connection, i.e. Host A will send Token-B (which is generated from Key-B). Note that the hash generation algorithm can be overridden by the choice of cryptographic handshake algorithm, as defined in Section 3.1.

The MP_JOIN SYN not only sends the token (which is static for a

connection) but also Random Numbers (nonces) that are used to prevent replay attacks on the authentication method. Recommendations for the generation of random numbers for this purpose are given in [13].

The MP_JOIN option includes an "Address ID". This is an identifier that only has significance within a single connection, where it identifies the source address of this packet, even if the IP header has been changed in transit by a middlebox. The Address ID allows address removal (Section 3.4.2) without needing to know what the source address at the receiver is, thus allowing address removal through NATs. The Address ID also allows correlation between new subflow setup attempts and address signaling (Section 3.4.1), to prevent setting up duplicate subflows on the same path, if a MP_JOIN and ADD_ADDR are sent at the same time.

The Address IDs of the subflow used in the initial SYN exchange of the first subflow in the connection are implicit, and have the value zero. A host MUST store the mappings between Address IDs and addresses both for itself and the remote host. An implementation will also need to know which local and remote Address IDs are associated with which established subflows, for when addresses are removed from a local or remote host.

The MP_JOIN option on packets with the SYN flag set also includes 4 bits of flags, 3 of which are currently reserved and MUST be set to zero by the sender. The final bit, labelled 'B', indicates whether the sender of this option wishes this subflow to be used as a backup path (B=1) in the event of failure of other paths, or whether it wants it to be used as part of the connection immediately. By setting B=1, the sender of the option is requesting the other host to only send data on this subflow if there are no available subflows where B=0. Subflow policy is discussed in more detail in Section 3.3.8.

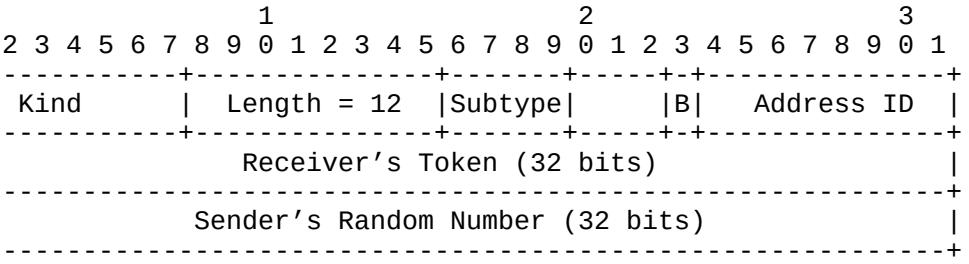


Figure 5: Join Connection (MP_JOIN) option (for initial SYN)

When receiving a SYN with an MP_JOIN option that contains a valid token for an existing MPTCP connection, the recipient SHOULD respond

with a SYN/ACK also containing an MP_JOIN option containing a random number and a truncated (leftmost 64 bits) Hash-based Message Authentication Code (HMAC). This version of the option is shown in Figure 6. If the token is unknown, or the host wants to refuse subflow establishment (for example, due to a limit on the number of subflows it will permit), the receiver will send back an RST, analogous to an unknown port in TCP. Although calculating an HMAC requires cryptographic operations, it is believed that the 32 bit token in the MP_JOIN SYN gives sufficient protection against blind state exhaustion attacks and therefore there is no need to provide mechanisms to allow a responder to operate statelessly at the MP_JOIN stage.

An HMAC is sent by both hosts - by the initiator (Host A) in the third packet (the ACK) and by the responder (Host B) in the second packet (the SYN/ACK). Doing the HMAC exchange at this stage allows both hosts to have first exchanged random data (in the first two SYN packets) that is used as the "message". This specification defines that HMAC as defined in [9] is used, along with the SHA-1 hash algorithm [4] (potentially implemented as in [14]), thus generating a 160-bit / 20 octet HMAC. Due to option space limitations, the HMAC included in the SYN/ACK is truncated to the leftmost 64 bits, but this is acceptable since random numbers are used, and thus an attacker only has one chance to guess the HMAC correctly (if the HMAC is incorrect, the TCP connection is closed, so a new MP_JOIN negotiation with a new random number is required).

The initiator’s authentication information is sent in its first ACK (the third packet of the handshake), as shown in Figure 7. This data needs to be sent reliably, since it is the only time this HMAC is sent and therefore receipt of this packet MUST trigger a regular TCP ACK in response, and the packet MUST be retransmitted if this ACK is not received. In other words, sending the ACK/MP_JOIN packet places the subflow in the PRE_ESTABLISHED state, and it moves to the ESTABLISHED state only on receipt of an ACK from the receiver. It is not permitted to send data while in the PRE_ESTABLISHED state. The reserved bits in this option MUST be set to zero by the sender.

The key for the HMAC algorithm, in the case of the message transmitted by Host A, will be Key-A followed by Key-B, and in the case of Host B, Key-B followed by Key-A. These are the keys that were exchanged in the original MP_CAPABLE handshake. The "message" for the HMAC algorithm in each case is the concatenations of Random Number for each host (denoted by R): for Host A, R-A followed by R-B; and for Host B, R-B followed by R-A.

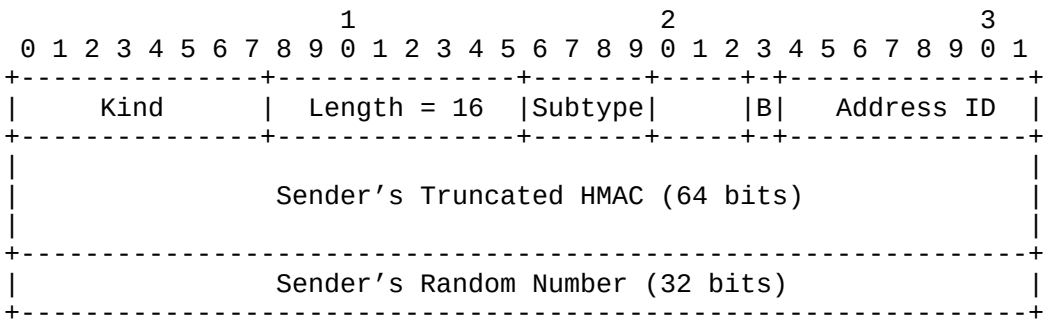


Figure 6: Join Connection (MP_JOIN) option (for responding SYN/ACK)

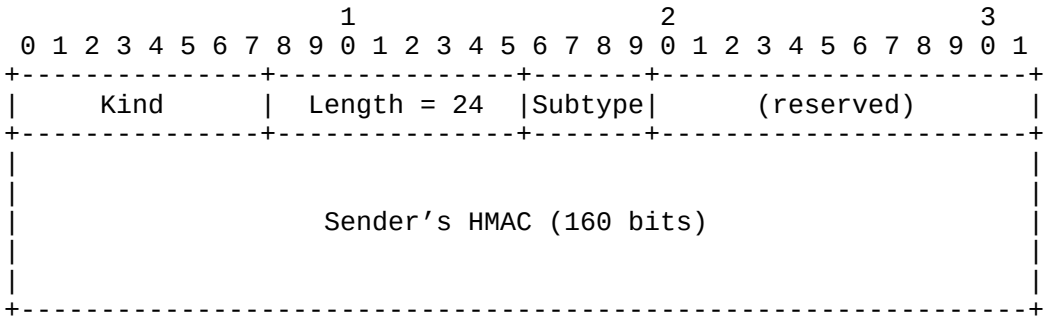
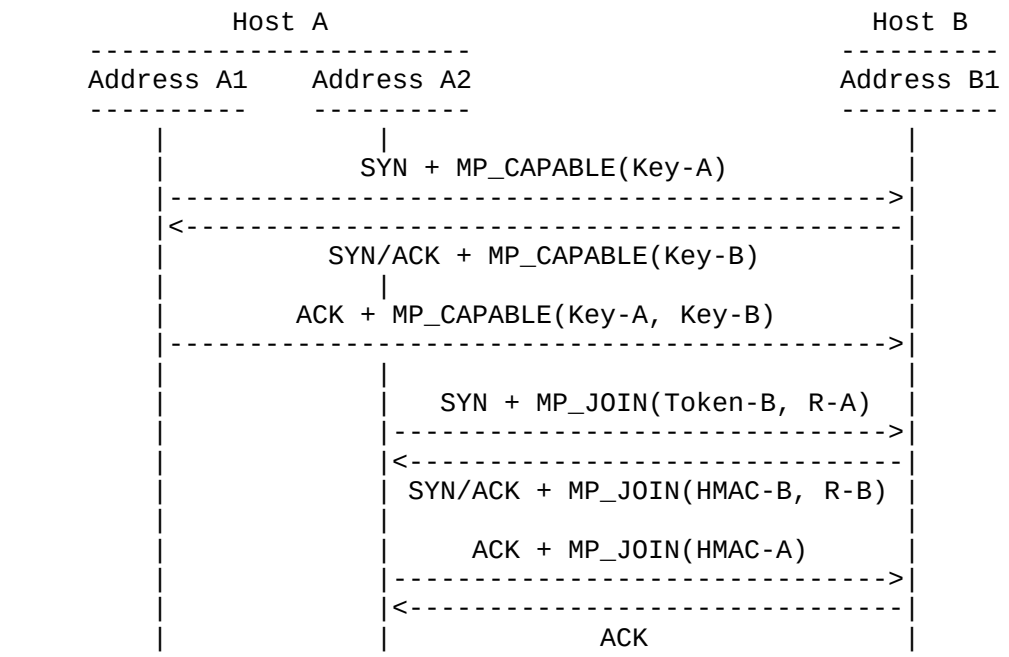


Figure 7: Join Connection (MP_JOIN) option (for third ACK)

These various TCP options fit together to enable authenticated subflow setup as illustrated in Figure 8.



HMAC-A = HMAC(Key=(Key-A+Key-B), Msg=(R-A+R-B))
HMAC-B = HMAC(Key=(Key-B+Key-A), Msg=(R-B+R-A))

Figure 8: Example use of MPTCP Authentication

If the token received at Host B is unknown or local policy prohibits the acceptance of the new subflow, the recipient MUST respond with a TCP RST for the subflow.

If the token is accepted at Host B, but the HMAC returned to Host A does not match the one expected, Host A MUST close the subflow with a TCP RST.

If Host B does not receive the expected HMAC, or the MP_JOIN option is missing from the ACK, it MUST close the subflow with a TCP RST.

If the HMACs are verified as correct, then both hosts have authenticated each other as being the same peers as existed at the start of the connection, and they have agreed of which connection this subflow will become a part.

If the SYN/ACK as received at Host A does not have an MP_JOIN option, Host A MUST close the subflow with a RST.

This covers all cases of the loss of an MP_JOIN. In more detail, if

MP_JOIN is stripped from the SYN on the path from A to B, and Host B does not have a passive opener on the relevant port, it will respond with an RST in the normal way. If in response to a SYN with an MP_JOIN option, a SYN/ACK is received without the MP_JOIN option (either since it was stripped on the return path, or it was stripped on the outgoing path but the passive opener on Host B responded as if it were a new regular TCP session), then the subflow is unusable and Host A MUST close it with a RST.

Note that additional subflows can be created between any pair of ports (but see Section 3.8 for heuristics); no explicit application-level accept calls or bind calls are required to open additional subflows. To associate a new subflow with an existing connection, the token supplied in the subflow's SYN exchange is used for demultiplexing. This then binds the 5-tuple of the TCP subflow to the local token of the connection. A consequence is that it is possible to allow any port pairs to be used for a connection.

Demultiplexing subflow SYNs MUST be done using the token; this is unlike traditional TCP, where the destination port is used for demultiplexing SYN packets. Once a subflow is setup, demultiplexing packets is done using the five-tuple, as in traditional TCP. The five-tuples will be mapped to the local connection identifier (token). Note that Host A will know its local token for the subflow even though it is not sent on the wire - only the responder's token is sent.

3.3. General MPTCP Operation

This section discusses operation of MPTCP for data transfer. At a high level, an MPTCP implementation will take one input data stream from an application, and split it into one or more subflows, with sufficient control information to allow it to be reassembled and delivered reliably and in-order to the recipient application. The following subsections define this behaviour in detail.

The Data Sequence Mapping and the Data ACK are signalled in the Data Sequence Signal (DSS) option. Either or both can be signalled in one DSS, dependent on the flags set. The Data Sequence Mapping defines how the sequence space on the subflow maps to the connection level, and the Data ACK acknowledges receipt of data at the connection level. These functions are described in more detail in the following two subsections.

Either or both the Data Sequence Mapping and the Data ACK can be signalled in the DSS option, dependent on the flags set.

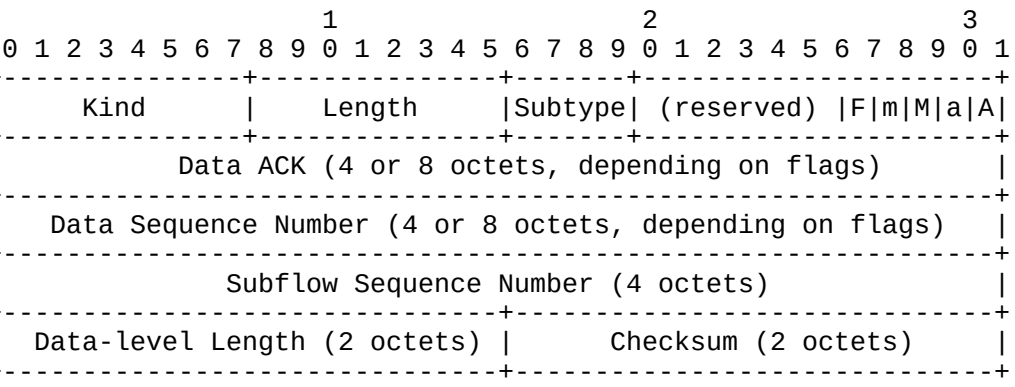


Figure 9: Data Sequence Signal (DSS) option

The flags when set define the contents of this option, as follows:

- A = Data ACK present
- a = Data ACK is 8 octets (if not set, Data ACK is 4 octets)
- M = Data Sequence Number, Subflow Sequence Number, Data-level Length, and Checksum present
- m = Data Sequence Number is 8 octets (if not set, DSN is 4 octets)

The flags ’a’ and ’m’ only have meaning if the corresponding ’A’ or ’M’ flags are set, otherwise they will be ignored. The maximum length of this option, with all flags set, is 28 octets.

The ’F’ flag indicates "DATA_FIN". If present, this means that this mapping covers the final data from the sender. This is the connection-level equivalent to the FIN flag in single-path TCP. A connection is not closed unless there has been a DATA_FIN exchange, or a timeout. The purpose of the DATA_FIN, along with the interactions between this flag, the subflow-level FIN flag, and the data sequence mapping are described in Section 3.3.3. The remaining reserved bits MUST be set to zero by an implementation of this specification.

Note that the Checksum is only present in this option if the use of MPTCP checksumming has been negotiated at the MP_CAPABLE handshake (see Section 3.1). The presence of the checksum can be inferred from the length of the option. If a checksum is present, but its use had not been negotiated in the MP_CAPABLE handshake, the checksum field MUST be ignored. If a checksum is not present when its use has been negotiated, the receiver MUST close the subflow with a RST as it is

considered broken.

3.3.1. Data Sequence Mapping

The data stream as a whole can be reassembled through the use of the Data Sequence Mapping components of the DSS option (Figure 9), which define the mapping from the subflow sequence number to the data sequence number. This is used by the receiver to ensure in-order delivery to the application layer. Meanwhile, the subflow-level sequence numbers (i.e. the regular sequence numbers in the TCP header) have subflow-only relevance. It is expected (but not mandated) that SACK [10] is used at the subflow level to improve efficiency.

The Data Sequence Mapping specifies a mapping from subflow sequence space to data sequence space. This is expressed in terms of starting sequence numbers for the subflow and the data level, and a length of bytes for which this mapping is valid. This explicit mapping for a range of data was chosen rather than per-packet signaling to assist with compatibility with situations where TCP/IP segmentation or coalescing is undertaken separately from the stack that is generating the data flow (e.g. through the use of TCP segmentation offloading on network interface cards, or by middleboxes such as performance enhancing proxies). It also allows a single mapping to cover many packets, which may be useful in bulk transfer situations.

A mapping is fixed, in that the subflow sequence number is bound to the data sequence number after the mapping has been processed. A sender MUST NOT change this mapping after it has been declared; however, the same data sequence number can be mapped to by different subflows for retransmission purposes (see Section 3.3.6). This would also permit the same data to be sent simultaneously on multiple subflows for resilience or efficiency purposes, especially in the case of lossy links. Although the detailed specification of such operation is outside the scope of this document, an implementation SHOULD treat the first data that is received at a subflow for the data sequence space as that which should be delivered to the application, and any later data for that sequence space ignored.

The data sequence number is specified as an absolute value, whereas the subflow sequence numbering is relative (the SYN at the start of the subflow has relative subflow sequence number 0). This is to allow middleboxes to change the Initial Sequence Number of a subflow, such as firewalls that undertake ISN randomization.

The data sequence mapping also contains a checksum of the data that this mapping covers, if use of checksums has been negotiated at the MP_CAPABLE exchange. Checksums are used to detect if the payload has

been adjusted in any way by a non-MPTCP-aware middlebox. If this checksum fails, it will trigger a failure of the subflow, or a fallback to regular TCP, as documented in Section 3.6, since MPTCP can no longer reliably know the subflow sequence space at the receiver to build data sequence mappings.

The checksum algorithm used is the standard TCP checksum [1], operating over the data covered by this mapping, along with a pseudo-header as shown in Figure 10.

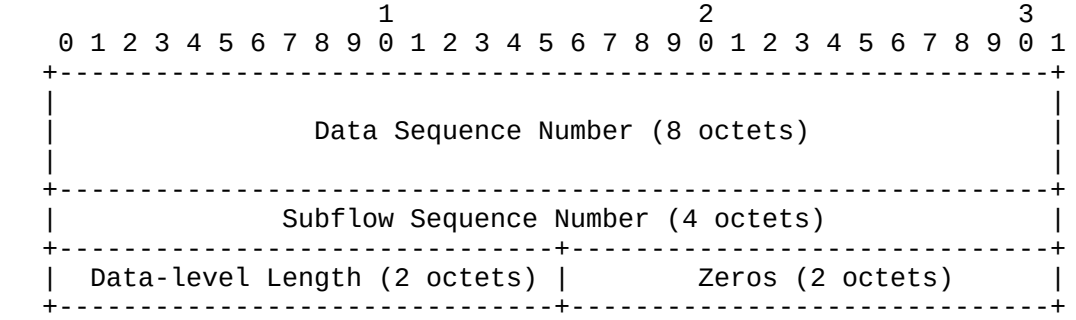


Figure 10: Pseudo-Header for DSS Checksum

Note that the Data Sequence Number used in the pseudo-header is always the 64 bit value, irrespective of what length is used in the DSS option itself. The standard TCP checksum algorithm has been chosen since it will be calculated anyway for the TCP subflow, and if calculated first over the data before adding the pseudo-headers, it only needs to be calculated once. Furthermore, since the TCP checksum is additive, the checksum for a DSN_MAP can be constructed by simply adding together the checksums for the data of each constituent TCP segment, and adding the checksum for the DSS pseudo-header.

Note that checksumming relies on the TCP subflow containing contiguous data, and therefore a TCP subflow MUST NOT use the Urgent Pointer to interrupt an existing mapping. Further note, however, that if Urgent data is received on a subflow, it SHOULD be mapped to the data sequence space and delivered to the application analogous to Urgent data in regular TCP.

To avoid possible deadlock scenarios, subflow-level processing should be undertaken separately from that at connection-level. Therefore, even if a mapping does not exist from the subflow space to the data-level space, the data SHOULD still be ACKed at the subflow (if it is in-window). This data cannot, however, be acknowledged at the data level (Section 3.3.2) because its data sequence numbers are unknown.

Implementations MAY hold onto such unmapped data for a short while in the expectation that a mapping will arrive shortly. Such unmapped data cannot be counted as being within the connection-level receive window because this is relative to the data sequence numbers, so if the receiver runs out of memory to hold this data, it will have to be discarded. If a mapping for that subflow-level sequence space does not arrive within a receive window of data, that subflow SHOULD be treated as broken, closed with an RST, and any unmapped data silently discarded.

Data sequence numbers are always 64 bit quantities, and MUST be maintained as such in implementations. If a connection is progressing at a slow rate, so protection against wrapped sequence numbers is not required, then it is permissible to include just the lower 32 bits of the data sequence number in the Data Sequence Mapping and/or Data ACK as an optimization, and an implementation can make this choice independently for each packet.

An implementation MUST send the full 64 bit Data Sequence Number if it is transmitting at a sufficiently high rate that the 32 bit value could wrap within the Maximum Segment Lifetime (MSL) [15]. The lengths of the DSNs used in these values (which may be different) are declared with flags in the DSS option. Implementations MUST accept a 32 bit DSN and implicitly promote it to a 64 bit quantity by incrementing the upper 32 bits of sequence number each time the lower 32 bits wrap. A sanity check MUST be implemented to ensure that a wrap occurs at an expected time (e.g. the sequence number jumps from a very high number to a very low number) and is not triggered by out-of-order packets.

As with the standard TCP sequence number, the data sequence number should not start at zero, but at a random value to make blind session hijacking harder. This specification requires setting the initial data sequence number (IDSN) of each host to the least significant 64 bits of the SHA-1 hash of the host’s key, as described in Section 3.1.

A Data Sequence Mapping does not need to be included in every MPTCP packet, as long as the subflow sequence space in that packet is covered by a mapping known at the receiver. This can be used to reduce overhead in cases where the mapping is known in advance; one such case is when there is a single subflow between the hosts, another is when segments of data are scheduled in larger than packet-sized chunks.

An "infinite" mapping can be used to fallback to regular TCP by mapping the subflow-level data to the connection-level data for the remainder of the connection (see Section 3.6). This is achieved by

setting the Data-level Length field of the DSS option to the reserved value of 0. The checksum, in such a case, will also be set to zero.

3.3.2. Data Acknowledgments

To provide full end-to-end resilience, MPTCP provides a connection-level acknowledgement, to act as a cumulative ACK for the connection as a whole. This is the "Data ACK" field of the DSS option (Figure 9). The Data ACK is analogous to the behaviour of the standard TCP cumulative ACK - indicating how much data has been successfully received (with no holes). This is in comparison to the subflow-level ACK, which acts analogous to TCP SACK, given that there may still be holes in the data stream at the connection level. The Data ACK specifies the next Data Sequence Number it expects to receive.

The Data ACK, as for the DSN, can be sent as the full 64 bit value, or as the lower 32 bits. If data is received with a 64 bit DSN, it MUST be acknowledged with a 64 bit Data ACK. If the DSN received is 32 bits, it is valid for the implementation to choose whether to send a 32 bit or 64 bit Data ACK.

The Data ACK proves that the data, and all required MPTCP signaling, has been received and accepted by the remote end. One key use of the Data ACK signal is that it is used to indicate the left edge of the advertised receive window. As explained in Section 3.3.4, the receive window is shared by all subflows and is relative to the Data ACK. Because of this, an implementation MUST NOT use the RCV.WND field of a TCP segment at connection-level if it does not also carry a DSS option with a Data ACK field. Furthermore, separating the connection-level acknowledgments from the subflow-level allows processing to be done separately, and a receiver has the freedom to drop segments after acknowledgement at the subflow level, for example due to memory constraints when many segments arrive out-of-order.

An MPTCP sender MUST NOT free data from the send buffer until it has been acknowledged by both a Data ACK received on any subflow and at the subflow level by all subflows the data was sent on. The former condition ensures liveness of the connection and the latter condition ensures liveness and self-consistence of a subflow when data needs to be retransmitted. Note, however, that if some data needs to be retransmitted multiple times over a subflow, there is a risk of blocking the sending window. In this case, the MPTCP sender can decide to terminate the subflow that is behaving badly by sending a RST.

The Data ACK MAY be included in all segments, however optimisations SHOULD be considered in more advanced implementations, where the Data

ACK is present in segments only when the Data ACK value advances, and this behaviour MUST be treated as valid. This behaviour ensures the sender buffer is freed, while reducing overhead when the data transfer is unidirectional.

3.3.3. Closing a Connection

In regular TCP a FIN announces the receiver that the sender has no more data to send. In order to allow subflows to operate independently and to keep the appearance of TCP over the wire, a FIN in MPTCP only affects the subflow on which it is sent. This allows nodes to exercise considerable freedom over which paths are in use at any one time. The semantics of a FIN remain as for regular TCP, i.e. it is not until both sides have ACKed each other's FINs that the subflow is fully closed.

When an application calls close() on a socket, this indicates that it has no more data to send, and for regular TCP this would result in a FIN on the connection. For MPTCP, an equivalent mechanism is needed, and this is referred to as the DATA_FIN.

A DATA_FIN is an indication that the sender has no more data to send, and as such can be used to verify that all data has been successfully received. A DATA_FIN, as with the FIN on a regular TCP connection, is a unidirectional signal.

The DATA_FIN is signalled by setting the 'F' flag in the Data Sequence Signal option (Figure 9) to 1. A DATA_FIN occupies one octet (the final octet) of the connection-level sequence space. Note that the DATA_FIN is included in the Data-Level Length, but not at the subflow level: for example, a segment with DSN 80, and Data-Level Length 11, with DATA_FIN set, would map 10 octets from the subflow into data sequnce space 80-89, the DATA_FIN is DSN 90, and therefore this segment including DATA_FIN would be acknowledged with a DATA_ACK of 91.

Note that when the DATA_FIN is not attached to a TCP segment containing data, the Data Sequence Signal MUST have Subflow Sequence Number of 0, a Data-Level Length of 1, and the Data Sequence Number that corresponds with the DATA_FIN itself. The checksum in this case will only cover the pseudo-header.

A DATA_FIN has the semantics and behaviour as a regular TCP FIN, but at the connection level. Notably, it is only DATA_ACKed once all data has been successfully received at the connection level. Note therefore that a DATA_FIN is decoupled from a subflow FIN. It is only permissible to combine these signals on one subflow if there is no data outstanding on other subflows. Otherwise, it may be

necessary to retransmit data on different subflows. Essentially, a host MUST NOT close all functioning subflows unless it is safe to do so, i.e. until all outstanding data has been DATA_ACKed, or that the segment with the DATA_FIN flag set is the only outstanding segment.

Once a DATA_FIN has been acknowledged, all remaining subflows MUST be closed with standard FIN exchanges. Both hosts SHOULD send FINs on all subflows, as a courtesy to allow middleboxes to clean up state even if an individual subflow has failed. It is also encouraged to reduce the timeouts (Maximum Segment Life) on subflows at end hosts. In particular, any subflows where there is still outstanding data queued (which has been retransmitted on other subflows in order to get the DATA_FIN acknowledged) MAY be closed with an RST.

A connection is considered closed once both hosts' DATA_FINs have been acknowledged by DATA_ACKs.

As specified above, a standard TCP FIN on an individual subflow only shuts down the subflow on which it was sent. If all subflows have been closed with a FIN exchange, but no DATA_FIN has been received and acknowledged, the MPTCP connection is treated as closed only after a timeout. This implies that an implementation will have TIME_WAIT states at both the subflow and connection levels (see Appendix C). This permits "break-before-make" scenarios where connectivity is lost on all subflows before a new one can be re-established.

3.3.4. Receiver Considerations

Regular TCP advertises a receive window in each packet, telling the sender how much data the receiver is willing to accept past the cumulative ack. The receive window is used to implement flow control, throttling down fast senders when receivers cannot keep up.

MPTCP also uses a unique receive window, shared between the subflows. The idea is to allow any subflow to send data as long as the receiver is willing to accept it; the alternative, maintaining per subflow receive windows, could end-up stalling some subflows while others would not use up their window.

The receive window is relative to the DATA_ACK. As in TCP, a receiver MUST NOT shrink the right edge of the receive window (i.e. DATA_ACK + receive window). The receiver will use the Data Sequence Number to tell if a packet should be accepted at connection level.

When deciding to accept packets at subflow level, regular TCP checks the sequence number in the packet against the allowed receive window. With multipath, such a check is done using only the connection level

window. A sanity check SHOULD be performed at subflow level to ensure that the subflow and mapped sequence numbers meet the following test: SSN - SUBFLOW_ACK <= DSN - DATA_ACK, where SSN is the subflow sequence number of the received packet and SUBFLOW_ACK is the RCV.NXT (next expected sequence number) of the subflow (with the equivalent connection-level definitions for DSN and DATA_ACK).

In regular TCP, once a segment is deemed in-window, it is either put in the in-order receive queue or in the out-of-order queue. In multipath TCP, the same happens but at connection-level: a segment is placed in the connection level in-order or out-of-order queue if it is in-window at both connection and subflow level. The stack still has to remember, for each subflow, which segments were received successfully so that it can ACK them at subflow level appropriately. Typically, this will be implemented by keeping per subflow out-of-order queues (containing only message headers, not the payloads) and remembering the value of the cumulative ACK.

It is important for implementers to understand how large a receiver buffer is appropriate. The lower bound for full network utilization is the maximum bandwidth-delay product of any one of the paths. However this might be insufficient when a packet is lost on a slower subflow and needs to be retransmitted (see Section 3.3.6). A tight upper bound would be the maximum RTT of any path multiplied by the total bandwidth available across all paths. This permits all subflows to continue at full speed while a packet is fast-retransmitted on the maximum RTT path. Even this might be insufficient to maintain full performance in the event of a retransmit timeout on the maximum RTT path. It is for future study to determine the relationship between retransmission strategies and receive buffer sizing.

3.3.5. Sender Considerations

The sender remembers receiver window advertisements from the receiver. It should only update its local receive window values when the largest sequence number allowed (i.e. DATA_ACK + receive window) increases, on the receipt of a DATA_ACK. This is important to allow using paths with different RTTs, and thus different feedback loops.

MPTCP uses a single receive window across all subflows, and if the receive window was guaranteed to be unchanged end-to-end, a host could always read the most recent receive window value. However, some classes of middleboxes may alter the TCP-level receive window. Typically these will shrink the offered window, although for short periods of time it may be possible for the window to be larger (however note that this would not continue for long periods since ultimately the middlebox must keep up with delivering data to the

receiver). Therefore, if receive window sizes differ on multiple subflows, when sending data MPTCP SHOULD take the largest of the most recent window sizes as the one to use in calculations. This rule is implicit in the requirement not to reduce the right edge of the window.

The sender MUST also remember the receive windows advertised by each subflow. The allowed window for subflow i is (ack_i, ack_i + rcv_wnd_i), where ack_i is the subflow-level cumulative ack of subflow i. This ensures data will not be sent to a middlebox unless there is enough buffering for the data.

Putting the two rules together, we get the following: a sender is allowed to send data segments with data-level sequence numbers between (DATA_ACK, DATA_ACK + receive_window). Each of these segments will be mapped onto subflows, as long as subflow sequence numbers are in the the allowed windows for those subflows. Note that subflow sequence numbers do not generally affect flow control if the same receive window is advertised across all subflows. They will perform flow control for those subflows with a smaller advertised receive window.

The send buffer MUST, at a minimum, be as big as the receive buffer, to enable the sender to reach maximum throughput.

3.3.6. Reliability and Retransmissions

The data sequence mapping allows senders to re-send data with the same data sequence number on a different subflow. When doing this, a host MUST still retransmit the original data on the original subflow, in order to preserve the subflow integrity (middleboxes could replay old data, and/or could reject holes in subflows), and a receiver will ignore these retransmissions. While this is clearly suboptimal, for compatibility reasons this is sensible behaviour. Optimisations could be negotiated in future versions of this protocol.

This protocol specification does not mandate any mechanisms for handling retransmissions, and much will be dependent upon local policy (as discussed in Section 3.3.8). One can imagine aggressive connection level retransmissions policies where every packet lost at subflow level is retransmitted on a different subflow (hence wasting bandwidth but possibly reducing application-to-application delays), or conservative retransmission policies where connection-level retransmits are only used after a few subflow level retransmission timeouts occur.

It is envisaged that a standard connection-level retransmission mechanism would be implemented around a connection-level data queue:

all segments that haven't been DATA_ACKed are stored. A timer is set when the head of the connection-level is ACKed at subflow level but its corresponding data is not ACKed at data level. This timer will guard against failures in re-transmission by middleboxes that pro-actively ACK data.

The sender MUST keep data in its send buffer as long as the data has not been acknowledged at both connection level and on all subflows it has been sent on. In this way, the sender can always retransmit the data if needed, on the same subflow or on a different one. A special case is when a subflow fails: the sender will typically resend the data on other working subflows after a timeout, and will keep trying to retransmit the data on the failed subflow too. The sender will declare the subflow failed after a predefined upper bound on retransmissions is reached (which MAY be lower than the usual TCP limits of the Maximum Segment Life), or on the receipt of an ICMP error, and only then delete the outstanding data segments.

Multiple retransmissions are triggers that will indicate that a subflow performs badly and could lead to a host resetting the subflow with an RST. However, additional research is required to understand the heuristics of how and when to reset underperforming subflows. For example, a highly asymmetric path may be mis-diagnosed as underperforming.

3.3.7. Congestion Control Considerations

Different subflows in an MPTCP connection have different congestion windows. To achieve fairness at bottlenecks and resource pooling, it is necessary to couple the congestion windows in use on each subflow, in order to push most traffic to uncongested links. One algorithm for achieving this is presented in [5]; the algorithm does not achieve perfect resource pooling but is "safe" in that it is readily deployable in the current Internet. By this, we mean that it does not take up more capacity on any one path than if it was a single path flow using only that route, so this ensures fair coexistence with single-path TCP at shared bottlenecks.

It is foreseeable that different congestion controllers will be implemented for MPTCP, each aiming to achieve different properties in the resource pooling/fairness/stability design space, as well as those for achieving different properties in quality of service, reliability and resilience.

Regardless of the algorithm used, the design of the MPTCP protocol aims to provide the congestion control implementations sufficient information to take the right decisions; this information includes, for each subflow, which packets were lost and when.

3.3.8. Subflow Policy

Within a local MPTCP implementation, a host may use any local policy it wishes to decide how to share the traffic to be sent over the available paths.

In the typical use case, where the goal is to maximise throughput, all available paths will be used simultaneously for data transfer, using coupled congestion control as described in [5]. It is expected, however, that other use cases will appear.

For instance, a possibility is an 'all-or-nothing' approach, i.e. have a second path ready for use in the event of failure of the first path, but alternatives could include entirely saturating one path before using an additional path (the 'overflow' case). Such choices would be most likely based on the monetary cost of links, but may also be based on properties such as the delay or jitter of links, where stability (of delay or bandwidth) is more important than throughput. Application requirements such as these are discussed in detail in [6].

The ability to make effective choices at the sender requires full knowledge of the path "cost", which is unlikely to be the case. It would be desirable for a receiver to be able to signal their own preferences for paths, since they will often be the multihomed party, and may have to pay for metered incoming bandwidth.

Whilst fine-grained control may be the most powerful solution, that would require some mechanism such as overloading the ECN signal [16], which is undesirable, and it is felt that there would not be sufficient benefit to justify an entirely new signal. Therefore the MP_JOIN option (see Section 3.2) contains the 'B' bit, which allows a host to indicate to its peer that this path should be treated as a backup path to use only in the event of failure of other working subflows (i.e. a subflow where the receiver has indicated B=1 SHOULD NOT be used to send data unless there are no usable subflows where B=0).

In the event that the available set of paths changes, a host may wish to signal a change in priority of subflows to the peer (e.g. a subflow that was previously set as backup should now take priority over all remaining subflows). Therefore, the MP_PRIO option, shown in Figure 11, can be used to change the 'B' flag of the subflow on which it is sent.

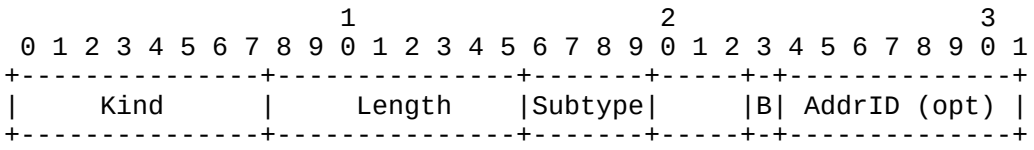


Figure 11: MP_PRIO option

It should be noted that the backup flag is a request from a data receiver to a data sender only, and the data sender SHOULD adhere to these requests. A host cannot assume that the data sender will do so, however, since local policies - or technical difficulties - may override MP_PRIO requests. Note also that this signal applies to a single direction, and so the sender of this option could choose to continue using the subflow to send data even if it has signalled B=1 to the other host.

This option can also be applied to other subflows than the one on which it is sent, by setting the optional Address ID field. This applies the given setting of B to all subflows in this connection that use the address identified by the given Address ID. The presence of this field is determined by the option length; if Length==4 then it is present, if Length==3 then it applies to the current subflow only. The use case of this is that a host can signal to its peer that an address is temporarily unavailable (for example, if it has radio coverage issues) and the peer should therefore drop to backup state on all subflows using that Address ID.

3.4. Address Knowledge Exchange (Path Management)

We use the term "path management" to refer to the exchange of information about additional paths between hosts, which in this design is managed by multiple addresses at hosts. For more detail of the architectural thinking behind this design, see the separate architecture document [2].

This design makes use of two methods of sharing such information, and both can be used on a connection. The first is the direct setup of new subflows, already described in Section 3.2, where the initiator has an additional address. The second method, described in the following subsections, signals addresses explicitly to the other host to allow it to initiate new subflows. The two mechanisms are complementary: the first is implicit and simple, while the explicit is more complex but is more robust. Together, the mechanisms allow addresses to change in flight (and thus support operation through NATs, since the source address need not be known), and also allow the signaling of previously unknown addresses, and of addresses belonging to other address families (e.g. both IPv4 and IPv6).

Here is an example of typical operation of the protocol:

- An MPTCP connection is initially set up between address/port A1 of host A and address/port B1 of host B. If host A is multihomed and multi-addressed, it can start an additional subflow from its address A2 to B1, by sending a SYN with a Join option from A2 to B1, using B's previously declared token for this connection. Alternatively, if B is multihomed, it can try to set up a new subflow from B2 to A1, using A's previously declared token. In either case, the SYN will be sent to the port already in use for the original subflow on the receiving host.

- Simultaneously (or after a timeout), an ADD_ADDR option (Section 3.4.1) is sent on an existing subflow, informing the receiver of the sender's alternative address(es). The recipient can use this information to open a new subflow to the sender's additional address. In our example, A will send ADD_ADDR option informing B of address/port A2. The mix of using the SYN-based option and the ADD_ADDR option, including timeouts, is implementation-specific and can be tailored to agree with local policy.

- If subflow A2-B1 is succesfully setup, host B can use the Address ID in the Join option to correlate this with the ADD_ADDR option that will also arrive on an existing subflow; now B knows not to open A2-B1, ignoring the ADD_ADDR. Otherwise, if B has not received the A2-B1 MP_JOIN SYN but received the ADD_ADDR, it can try to initiate a new subflow from one or more of its addresses to address A2. This permits new sessions to be opened if one host is behind a NAT.

Other ways of using the two signaling mechanisms are possible; for instance, signaling addresses in other address families can only be done explicitly using the Add Address option.

3.4.1. Address Advertisement

The Add Address (ADD_ADDR) TCP Option announces additional addresses (and optionally, ports) on which a host can be reached (Figure 12). Multiple instances of this TCP option can be added in a single message if there is sufficient TCP option space, otherwise multiple TCP messages containing this option will be sent. This option can be used at any time during a connection, depending on when the sender wishes to enable multiple paths and/or when paths become available. As with all MPTCP signals, the receiver MUST undertake standard TCP validity checks before acting upon it.

Every address has an Address ID which can be used for uniquely

identifying the address within a connection, for address removal. This is also used to identify MP_JOIN options (see Section 3.2) relating to the same address, even when address translators are in use. The Address ID MUST uniquely identify the address to the sender (within the scope of the connection), but the mechanism for allocating such IDs is implementation-specific.

All address IDs learnt via either MP_JOIN or ADD_ADDR SHOULD be stored by the receiver in a data structure that gathers all the Address ID to address mappings for a connection (identified by a token pair). In this way there is a stored mapping between Address ID, observed source address and token pair for future processing of control information for a connection. Note that an implementation MAY discard incoming address advertisements at will, for example for avoiding the required mapping state, or because advertised addresses are of no use to it (for example, IPv6 addresses when it has IPv4 only). Therefore, a host MUST treat address advertisements as soft state, and MAY choose to refresh advertisements periodically.

This option is shown in Figure 12. The illustration is sized for IPv4 addresses (IPVer = 4). For IPv6, the IPVer field will read 6, and the length of the address will be 16 octets (instead of 4).

The presence of the final two octets, specifying the TCP port number to use, are optional and can be inferred from the length of the option. Although it is expected that the majority of use cases will use the same port pairs as used for the initial subflow (e.g. port 80 remains port 80 on all subflows, as does the ephemeral port at the client), there may be cases (such as port-based load balancing) where the explicit specification of a different port is required. If no port is specified, MPTCP SHOULD attempt to connect to the specified address on the same port as is already in use by the subflow on which the ADD_ADDR signal was sent; this is discussed in more detail in Section 3.8.

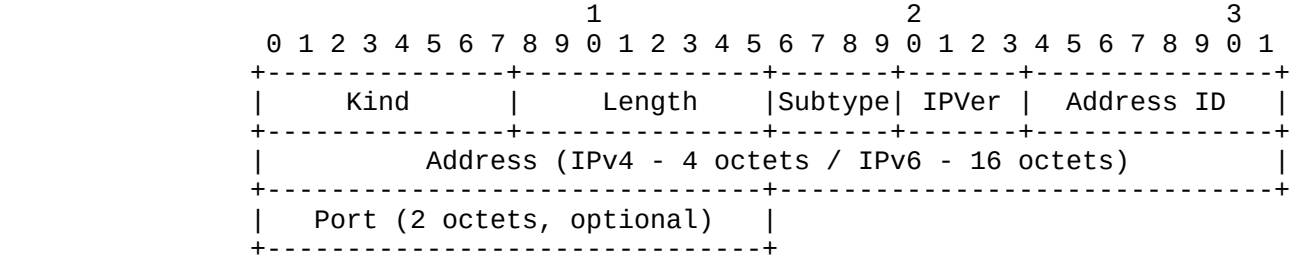


Figure 12: Add Address (ADD_ADDR) option

Due to the proliferation of NATs, it is reasonably likely that one

host may attempt to advertise private addresses [17]. It is not desirable to prohibit this, since there may be cases where both hosts have additional interfaces on the same private network, and a host MAY want to advertise such addresses. The MP_JOIN handshake to create a new subflow (Section 3.2) provides mechanisms to minimise security risks. The MP_JOIN message contains a 32 bit token that uniquely identifies the connection to the receiving host. If the token is unknown, the host will return with a RST. In the unlikely event that the token is known, subflow setup will continue, but the HMAC exchange must occur for authentication. This will fail, and will provide sufficient protection against two unconnected hosts accidentally setting up a new subflow upon the signal of a private address. Further security considerations around the issue of ADD_ADDR messages that accidentally mis-direct, or maliciously direct, new MP_JOIN attempts are discussed in Section 5.

Ideally, ADD_ADDR and REMOVE_ADDR options would be sent reliably, and in order, to the other end. This would ensure that this address management does not unnecessarily cause an outage in the connection when remove/add addresses are processed in reverse order, and also to ensure that all possible paths are used. Note, however, that losing reliability and ordering will not break the multipath connections, it will just reduce the opportunity to open multipath paths and to survive different patterns of path failures.

Therefore, implementing reliability signals for these TCP options is not necessary. In order to minimise the impact of the loss of these options, however, it is RECOMMENDED that a sender should send these options on all available subflows. If these options need to be received in-order, an implementation SHOULD only send one ADD_ADDR/ REMOVE_ADDR option per RTT, to minimise the risk of misordering.

A host can send an ADD_ADDR message with an already assigned Address ID, but the Address MUST be the same as previously assigned to this Address ID, and the Port MUST be different to one already in use for this Address ID. If these conditions are not met, the receiver SHOULD silently ignore the ADD_ADDR. A host wishing to replace an existing Address ID MUST first remove the existing one (Section 3.4.2).

A host that receives an ADD_ADDR but finds a connection setup to that IP address and port number is unsuccessful SHOULD NOT perform further connection attempts to this address/port combination for this connection. A sender that wants to trigger a new incoming connection attempt on a previously advertised address/port combination can therefore refresh ADD_ADDR information by sending the option again.

During normal MPTCP operation, it is unlikely that there will be

sufficient TCP option space for ADD_ADDR to be included along with those for data sequence numbering (Section 3.3.1). Therefore, it is expected that an MPTCP implementation will send the ADD_ADDR option on separate ACKs. As discussed earlier, however, an MPTCP implementation MUST NOT treat duplicate ACKs with any MPTCP option, with the exception of the DSS option, as indications of congestion [11], and an MPTCP implementation SHOULD NOT send more than two duplicate ACKs in a row for signaling purposes.

3.4.2. Remove Address

If, during the lifetime of an MPTCP connection, a previously-announced address becomes invalid (e.g. if the interface disappears), the affected host SHOULD announce this so that the peer can remove subflows related to this address.

This is achieved through the Remove Address (REMOVE_ADDR) option (Figure 13), which will remove a previously-added address (or list of addresses) from a connection and terminate any subflows currently using that address.

For security purposes, if a host receives a REMOVE_ADDR option, it must ensure the affected path(s) are no longer in use before it instigates closure. The receipt of REMOVE_ADDR SHOULD first trigger the sending of a TCP Keepalive [18] on the path, and if a response is received the path SHOULD NOT be removed. Typical TCP validity tests on the subflow (e.g. ensuring sequence and ack numbers are correct) MUST also be undertaken. An implementation can use indications of these test failures as part of intrusion detection or error logging.

The sending and receipt (if no keepalive response was received) of this message SHOULD trigger the sending of RSTs by both hosts on the affected subflow(s) (if possible), as a courtesy to cleaning up middlebox state, before cleaning up any local state.

Address removal is undertaken by ID, so as to permit the use of NATs and other middleboxes that rewrite source addresses. If there is no address at the requested ID, the receiver will silently ignore the request.

A subflow that is still functioning MUST be closed with a FIN exchange as in regular TCP, rather than using this option. For more information, see Section 3.3.3.

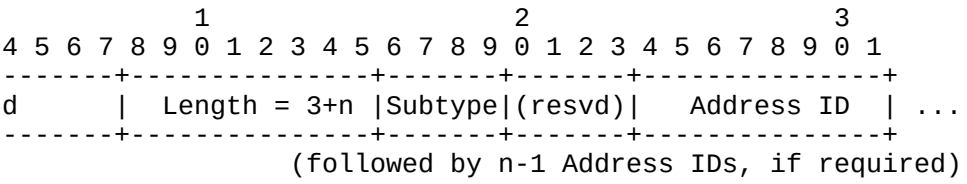


Figure 13: Remove Address (REMOVE_ADDR) option

3.5. Fast Close

Regular TCP has the means of sending a reset signal (RST) to abruptly close a connection. With MPTCP, the RST only has the scope of the subflow and will only close the concerned subflow but not affect the remaining subflows. MPTCP's connection will stay alive at the data-level, in order to permit break-before-make handover between subflows. It is therefore necessary to provide an MPTCP-level "reset" to allow the abrupt closure of the whole MPTCP connection, and this is the MP_FASTCLOSE option.

MP_FASTCLOSE is used to indicate to the peer that the connection will be abruptly closed and no data will be accepted any more. The reasons for triggering an MP_FASTCLOSE are implementation-specific. Regular TCP does not allow sending a RST while the connection is in a synchronized state [1]. Nevertheless, implementations allow the sending of a RST in this state, if for example the operating system is running out of resources. In these cases, MPTCP should send the MP_FASTCLOSE. This option is illustrated in Figure 14.

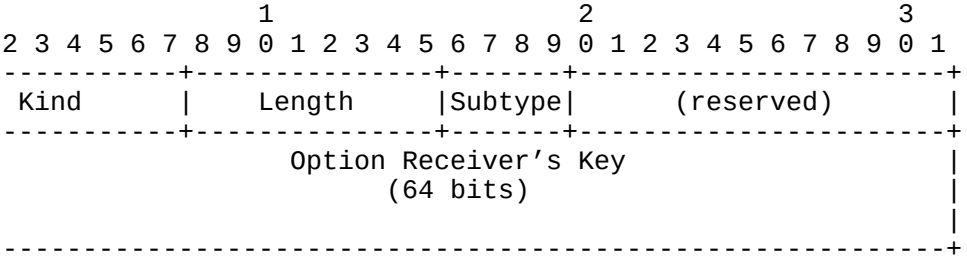


Figure 14: Fast Close (MP_FASTCLOSE) option

If Host A wants to force the closure of an MPTCP connection, the MPTCP Fast Close procedure is as follows:

- Host A sends an ACK containing the MP_FASTCLOSE option on one subflow, containing the key of Host B as declared in the initial connection handshake. On all the other subflows, Host A sends a regular TCP RST to close these subflows, and tears them down.

Host A now enters FASTCLOSE_WAIT state.

- Upon receipt of an MP_FASTCLOSE, containing the valid key, host B answers on the same subflow with a TCP RST and tears down all subflows. Host B can now close the whole MPTCP connection (it transitions directly to CLOSED state).

- As soon as Host A has received the TCP RST on the remaining subflow, it can close this subflow and tear down the whole connection (transition from FASTCLOSE_WAIT to CLOSED states). If Host A receives an MP_FASTCLOSE instead of a TCP RST, both hosts attempted fast closure simultaneously. Hose A should reply with a TCP RST and tear down the connection.

- If host A does not receive a TCP RST in reply to its MP_FASTCLOSE after one RTO (the RTO of the subflow where the MPTCP_RST has been sent), it SHOULD retransmit the MP_FASTCLOSE. The number of retransmissions SHOULD be limited to avoid this connection from being retained for a long time, but this limit is implementation-specific. A RECOMMENDED number is 3.

3.6. Fallback

Sometimes, middleboxes will exist on a path that could prevent the operation of MPTCP. MPTCP has been designed in order to cope with many middlebox modifications (see Section 6), but there are still some cases where a subflow could fail to operate within the MPTCP requirements. These cases are notably: the loss of TCP options on a path; and the modification of payload data. If such an event occurs, it is necessary to "fall back" to the previous, safe operation. This may either be falling back to regular TCP, or removing a problematic subflow.

At the start of an MPTCP connection (i.e. the first subflow), it is important to ensure that the path is fully MPTCP-capable and the necessary TCP options can reach each host. The handshake as described in Section 3.1 SHOULD fall back to regular TCP if either of the SYN messages do not have the MPTCP options: this is the same, and desired, behaviour in the case where a host is not MPTCP capable, or the path does not support the MPTCP options. When attempting to join an existing MPTCP connection (Section 3.2), if a path is not MPTCP capable and the TCP options do not get through on the SYNs, the subflow will be closed according to the MP_JOIN logic.

There is, however, another corner case which should be addressed. That is one of MPTCP options getting through on the SYN, but not on regular packets. This can be resolved if the subflow is the first subflow, and thus all data in flight is contiguous, using the

following rules.

A sender MUST include a DSS option with Data Sequence Mapping in every segment until one of the sent segments has been acknowledged with a DSS option containing a Data ACK. Upon reception of the acknowledgement, the sender has the confirmation that the DSS option passes in both directions and may choose to send fewer DSS options than once per segment.

If, however, an ACK is received for data (not just for the SYN) without a DSS option containing a Data ACK, the sender determines the path is not MPTCP capable. In the case of this occurring on an additional subflow (i.e. one started with MP_JOIN), the host MUST close the subflow with an RST. In the case of the first subflow (i.e. that started with MP_CAPABLE), it MUST drop out of an MPTCP mode back to regular TCP. The sender will send one final Data Sequence Mapping, with the Data-Level Length value of 0 indicating an infinite mapping (in case the path drops options in one direction only), and then revert to sending data on the single subflow without any MPTCP options.

Note that this rule essentially prohibits the sending of data on the third packet of an MP_CAPABLE or MP_JOIN handshake, since both that option and a DSS cannot fit in TCP option space. If the initiator is to send first, another segment must be sent that contains the data and DSS. Note also that an additional subflow cannot be used until the initial path has been verified as MPTCP-capable.

These rules should cover all cases where such a failure could happen: whether it’s on the forward or reverse path, and whether the server or the client first sends data. If lost options on data packets occur on any other subflow apart from the the initial subflow, it should be treated as a standard path failure. The data would not be DATA_ACKed (since there is no mapping for the data), and the subflow can be closed with an RST.

The case described above is a specialised case of fallback, for when the lack of MPTCP support is detected before any data is acknowledged at the connection level on a subflow. More generally, fallback (either closing a subflow, or to regular TCP) can become necessary at any point during a connection if a non-MPTCP-aware middlebox changes the data stream.

As described in Section 3.3, each portion of data for which there is a mapping is protected by a checksum. This mechanism is used to detect if middleboxes have made any adjustments to the payload (added, removed, or changed data). A checksum will fail if the data has been changed in any way. This will also detect if the length of

data on the subflow is increased or decreased, and this means the Data Sequence Mapping is no longer valid. The sender no longer knows what subflow-level sequence number the receiver is genuinely operating at (the middlebox will be faking ACKs in return), and cannot signal any further mappings. Furthermore, in addition to the possibility of payload modifications that are valid at the application layer, there is the possibility that false-positives could be hit across MPTCP segment boundaries, corrupting the data. Therefore, all data from the start of the segment that failed the checksum onwards is not trustworthy.

When multiple subflows are in use, the data in-flight on a subflow will likely involve data that is not contiguously part of the connection-level stream, since segments will be spread across the multiple subflows. Due to the problems identified above, it is not possible to determine what the adjustment has done to the data (notably, any changes to the subflow sequence numbering). Therefore, it is not possible to recover the subflow, and the affected subflow must be immediately closed with an RST, featuring an MP_FAIL option (Figure 15), which defines the Data Sequence Number at the start of the segment (defined by the Data Sequence Mapping) which had the checksum failure. Note that the MP_FAIL option requires the use of the full 64-bit sequence number, even if 32-bit sequence numbers are normally in use in the DSS signals on the path.

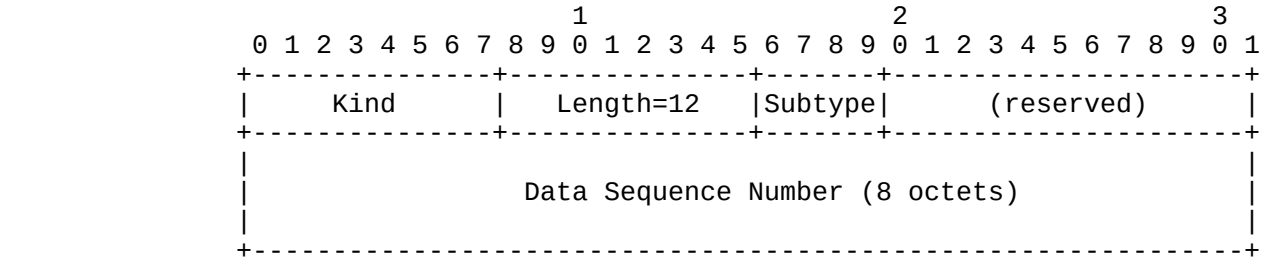


Figure 15: Fallback (MP_FAIL) option

The receiver MUST discard all data following the data sequence number specified. Failed data MUST NOT be DATA_ACKed and so will be re-transmitted on other subflows (Section 3.3.6).

A special case is when there is a single subflow and it fails with a checksum error. If it is known that all unacknowledged data in flight is contiguous (which will usually be the case with a single subflow), an infinite mapping can be applied to the subflow without the need to close it first, and essentially turn off all further MPTCP signaling. In this case, if a receiver identifies a checksum

failure when there is only one path, it will send back an MP_FAIL option on the subflow-level ACK, refering to the data-level sequence number of the start of the segment on which the checksum error was detected. The sender will receive this, and if all unacknowledged data in flight is contiguous, will signal an infinite mapping. This infinite mapping will be a DSS option (Section 3.3) on the first new packet, containing a Data Sequence Mapping that acts retroactively, referring to the start of the subflow sequence number of the last segment that was known to be delivered intact. From that point onwards data can be altered by a middlebox without affecting MPTCP, as the data stream is equivalent to a regular, legacy TCP session.

In the rare case that the data is not contiguous (which could happen when there is only one subflow but it is retransmitting data from a subflow that has recently been uncleanly closed), the receiver MUST close the subflow with an RST with MP_FAIL. The receiver MUST discard all data that follows the data sequence number specified. The sender MAY attempt to create a new subflow belonging to the same connection, and if it chooses to do so, SHOULD place the single subflow immediately in single-path mode by setting an infinite data sequence mapping. This mapping will begin from the data-level sequence number that was declared in the MP_FAIL.

After a sender signals an infinite mapping it MUST only use subflow ACKs to clear its send buffer. This is because Data ACKs may become misaligned with the subflow ACKs when middleboxes insert or delete data. The receive SHOULD stop generating Data ACKs after it receives an infinite mapping.

When a connection has fallen back, only one subflow can send data, otherwise the receiver would not know how to reorder the data. In practice, this means that all MPTCP subflows will have to be terminated except one. Once MPTCP falls back to regular TCP, it MUST NOT revert to MPTCP later in the connection.

It should be emphasised that we are not attempting to prevent the use of middleboxes that want to adjust the payload. An MPTCP-aware middlebox could provide such functionality by also rewriting checksums.

3.7. Error Handling

In addition to the fallback mechanism as described above, the standard classes of TCP errors may need to be handled in an MPTCP-specific way. Note that changing semantics - such as the relevance of an RST - are covered in Section 4. Where possible, we do not want to deviate from regular TCP behaviour.

The following list covers possible errors and the appropriate MPTCP behaviour:

- Unknown token in MP_JOIN (or HMAC failure in MP_JOIN ACK, or missing MP_JOIN in SYN/ACK response): send RST (analogous to TCP's behaviour on an unknown port)

- DSN out of Window (during normal operation): drop the data, do not send Data ACKs.

- Remove request for unknown address ID: silently ignore

3.8. Heuristics

There are a number of heuristics that are needed for performance or deployment but which are not required for protocol correctness. In this section we detail such heuristics. Note that discussion of buffering and certain sender and receiver window behaviours are presented in Section 3.3.4 and Section 3.3.5, as well as retransmission in Section 3.3.6.

3.8.1. Port Usage

Under typical operation an MPTCP implementation SHOULD use the same ports as already in use. In other words, the destination port of a SYN containing an MP_JOIN option SHOULD be the same as the remote port of the first subflow in the connection. The local port for such SYNs SHOULD also be the same as for the first subflow (and as such, an implementation SHOULD reserve ephemeral ports across all local IP addresses), although there may be cases where this is infeasible. This strategy is intended to maximize the probability of the SYN being permitted by a firewall or NAT at the recipient and to avoid confusing any network monitoring software.

There may also be cases, however, where the passive opener wishes to signal to the other host that a specific port should be used, and this facility is provided in the Add Address option as documented in Section 3.4.1. It is therefore feasible to allow multiple subflows between the same two addresses but using different port pairs, and such a facility could be used to allow load balancing within the network based on 5-tuples (e.g. some ECMP implementations [7]).

3.8.2. Delayed Subflow Start

Many TCP connections are short-lived and consist only of a few segments, and so the overheads of using MPTCP outweigh any benefits. A heuristic is required, therefore, to decide when to start using additional subflows in an MPTCP connection. We expect that

experience gathered from deployments will provide further guidance on this, and will be affected by particular application characteristics (which are likely to change over time). However, a suggested general-purpose heuristic that an implementation MAY choose to employ is as follows. Results from experimental deployments are needed in order to verify the correctness of this proposal.

If a host has data buffered for its peer (which implies that the application has received a request for data), the host opens one subflow for each initial window's worth of data that is buffered.

Consideration should also be given to limiting the rate of adding new subflows, as well as limiting the total number of subflows open for a particular connection. A host may choose to vary these values based on its load or knowledge of traffic and path characteristics.

Note that this heuristic alone is probably insufficient. Traffic for many common applications, such as downloads, is highly asymmetric and the host that is multihomed may well be the client which will never fill its buffers, and thus never use MPTCP. Advanced APIs that allow an application to signal its traffic requirements would aid in these decisions.

An additional time-based heuristic could be applied, opening additional subflows after a given period of time has passed. This would alleviate the above issue, and also provide resilience for low-bandwidth but long-lived applications.

This section has shown some of the considerations that an implementer should give when developing MPTCP heuristics, but is not intended to be prescriptive.

3.8.3. Failure Handling

Requirements for MPTCP's handling of unexpected signals have been given in Section 3.7. There are other failure cases, however, where a hosts can choose appropriate behaviour.

For example, Section 3.1 suggests that a host SHOULD fall back to trying regular TCP SYNs after one or more failures of MPTCP SYNs for a connection. A host may keep a system-wide cache of such information, so that it can back off from using MPTCP, firstly for that particular destination host, and eventually on a whole interface, if MPTCP connections continue failing.

Another failure could occur when the MP_JOIN handshake fails. Section 3.7 specifies that an incorrect handshake MUST lead to the subflow being closed with a RST. A host operating an active

intrusion detection system may choose to start blocking MP_JOIN packets from the source host if multiple failed MP_JOIN attempts are seen. From the connection initiator's point of view, if an MP_JOIN fails, it SHOULD NOT attempt to connect to the same IP address and port during the lifetime of the connection, unless the other host refreshes the information with another ADD_ADDR option. Note that the ADD_ADDR option is informational only, and does not guarantee the other host will attempt a connection.

In addition, an implementation may learn over a number of connections that certain interfaces or destination addresses consistently fail and may default to not trying to use MPTCP for these. Behaviour could also be learnt for particularly badly performing subflows or subflows that regularly fail during use, in order to temporarily choose not to use these paths.

4. Semantic Issues

In order to support multipath operation, the semantics of some TCP components have changed. To aid clarity, this section collects these semantic changes as a reference.

Sequence Number: The (in-header) TCP sequence number is specific to the subflow. To allow the receiver to reorder application data, an additional data-level sequence space is used. In this data-level sequence space, the initial SYN and the final DATA_FIN occupy one octet of sequence space. There is an explicit mapping of data sequence space to subflow sequence space, which is signalled through TCP options in data packets.

ACK: The ACK field in the TCP header acknowledges only the subflow sequence number, not the data-level sequence space. Implementations SHOULD NOT attempt to infer a data-level acknowledgement from the subflow ACKs. This separates subflow- and connection-level processing at an end host.

Duplicate ACK: A duplicate ACK that includes any MPTCP signaling (with the exception of the DSS option) MUST NOT be treated as a signal of congestion. To limit the chances of non-MPTCP-aware entities mistakenly interpreting duplicate ACKs as a signal of congestion, MPTCP SHOULD NOT send more than two duplicate ACKs containing (non-DSS) MPTCP signals in a row.

Receive Window: The receive window in the TCP header indicates the amount of free buffer space for the whole data-level connection (as opposed to for this subflow) that is available at the receiver. This is the same semantics as regular TCP, but to

maintain these semantics the receive window must be interpreted at the sender as relative to the sequence number given in the DATA_ACK rather than the subflow ACK in the TCP header. In this way the original flow control role is preserved. Note that some middleboxes may change the receive window, and so a host SHOULD use the maximum value of those recently seen on the constituent subflows for the connection-level receive window, and also needs to maintain a subflow-level window for subflow-level processing.

FIN: The FIN flag in the TCP header applies only to the subflow it is sent on, not to the whole connection. For connection-level FIN semantics, the DATA_FIN option is used.

RST: The RST flag in the TCP header applies only to the subflow it is sent on, not to the whole connection. The MP_FASTCLOSE option provides the fast-close functionality of a RST at the MPTCP connection level.

Address List: Address list management (i.e. knowledge of the local and remote hosts' lists of available IP addresses) is handled on a per-connection basis (as opposed to per-subflow, per host, or per pair of communicating hosts). This permits the application of per-connection local policy. Adding an address to one connection (either explicitly through an Add Address message, or implicitly through a Join) has no implication for other connections between the same pair of hosts.

5-tuple: The 5-tuple (protocol, local address, local port, remote address, remote port) presented by kernel APIs to the application layer in a non-multipath-aware application is that of the first subflow, even if the subflow has since been closed and removed from the connection. This decision, and other related API issues, are discussed in more detail in [6].

5. Security Considerations

As identified in [8], the addition of multipath capability to TCP will bring with it a number of new classes of threat. In order to prevent these, [2] presents a set of requirements for a security solution for MPTCP. The fundamental goal is for the security of MPTCP to be "no worse" than regular TCP today, and the key security requirements are:

- Provide a mechanism to confirm that the parties in a subflow handshake are the same as in the original connection setup.

- Provide verification that the peer can receive traffic at a new address before using it as part of a connection.

- Provide replay protection, i.e. ensure that a request to add/ remove a subflow is 'fresh'.

In order to achieve these goals, MPTCP includes a hash-based handshake algorithm documented in Section 3.1 and Section 3.2.

The security of the MPTCP connection hangs on the use of keys that are shared once at the start of the first subflow, and are never sent again over the network (unless used in the fast close mechanism, Section 3.5). To ease demultiplexing whilst not giving away any cryptographic material, future subflows use a truncated cryptographic hash of this key as the connection identification "token". The keys are concatenated and used as keys for creating Hash-based Message Authentication Codes (HMAC) used on subflow setup, in order to verify that the parties in the handshake are the same as in the original connection setup. It also provides verification that the peer can receive traffic at this new address. Replay attacks would still be possible when only keys are used, and therefore the handshakes use single-use random numbers (nonces) at both ends - this ensures the HMAC will never be the same on two handshakes. Guidance on generating random numbers suitable for use as keys is given in [13] and discussed in Section 3.1.

The use of crypto capability bits in the initial connection handshake to negotiate use of a particular algorithm allows the deployment of additional crypto mechanisms in the future. Note that this would be susceptible to bid-down attacks only if the attacker was on-path (and thus would be able to modify the data anyway). The security mechanism presented in this draft should therefore protect against all forms of flooding and hijacking attacks discussed in [8].

During normal operation, regular TCP protection mechanisms (such as ensuring sequence numbers are in-window) will provide the same level of protection against attacks on indivudal TCP subflows as exists for regular TCP today. Implementations will introduce additional buffers compared to regular TCP, to reassemble data at the connection level. The application of window sizing will minimize the risk of denial-of-service attacks consuming resources.

As discussed in Section 3.4.1, a host may advertise its private addresses, but these might point to different hosts in the receiver's network. The MP_JOIN handshake (Section 3.2) will ensure that this does not succeed in setting up a subflow to the incorrect host. However, it could still create unwanted TCP handshake traffic. This feature of MPTCP could be a target for denial-of-service exploits,

with malicious participants in MPTCP connections encouraging the recipient to target other hosts in the network. Therefore, implementations should consider heuristics (Section 3.8) at both the sender and receiver to reduce the impact of this.

A small security risk could theoretically exist with key reuse, but in order to accomplish a replay attack, both the sender and receiver keys, and the sender and receiver random numbers, in the MP_JOIN handshake (Section 3.2) would have to match.

Whilst this specification defines a "medium" security solution, meeting the criteria specified at the start of this section and the threat analyis ([8]), since attacks only ever get worse, it is likely that a future standards-track version of MPTCP would need to be able to support stronger security. There are several ways the security of MPTCP could potentially be improved; some of these would be compatible with MPTCP as defined in this document, whilst others may not be. For now, the best approach is to get experience with the current approach, establish what might work and check that the threat analysis is still accurate.

Possible ways of improving MPTCP security could include:

- defining a new MPCTP cryptographic algorithm, as negotiated in MP_CAPABLE. A sub-case could be to include an additional deployment assumption, such as stateful servers, in order to allow a more powerful algorithm to be used.

- defining how to secure data transfer with MPTCP, whilst not changing the signalling part of the protocol.

- defining security that requires more option space, perhaps in conjunction with a "long options" proposal for extending the TCP options space (such as those surveyed in [19]), or perhaps building on the current approach with a second stage of MPTCP-option-based security.

- re-visiting the working group’s decision to exclusively use TCP options for MPTCP signalling, and instead look at also making use of the TCP payloads.

MPTCP has been designed with several methods available to indicate a new security mechanism, including:

- available flags in MP_CAPABLE (Figure 4);

- available subtypes in the MPTCP Option Figure 3);

- o the version field in MP_CAPABLE (Figure 4);

6. Interactions with Middleboxes

Multipath TCP was designed to be deployable in the present world. Its design takes into account "reasonable" existing middlebox behaviour. In this section we outline a few representative middlebox-related failure scenarios and show how multipath TCP handles them. Next, we list the design decisions multipath has made to accommodate the different middleboxes.

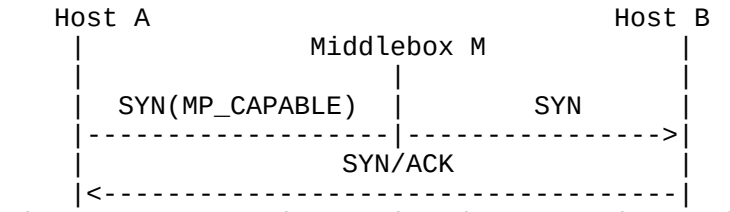
A primary concern is our use of a new TCP option. Middleboxes should forward packets with unknown options unchanged, yet there are some that don't. These we expect will either strip options and pass the data, drop packets with new options, copy the same option into multiple segments (e.g. when doing segmentation) or drop options during segment coalescing.

MPTCP uses a single new TCP option "Kind", and all message types are defined by "subtype" values (see Section 8). This should reduce the chances of only some types of MPTCP options being passed, and instead the key differing characteristics are different paths, and the presence of the SYN flag.

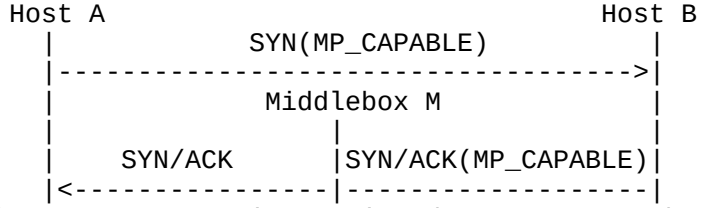
MPTCP SYN packets on the first subflow of a connection contain the MP_CAPABLE option (Section 3.1). If this is dropped, MPTCP SHOULD fall back to regular TCP. If packets with the MP_JOIN option (Section 3.2) are dropped, the paths will simply not be used.

If a middlebox strips options but otherwise passes the packets unchanged, MPTCP will behave safely. If an MP_CAPABLE option is dropped on either the outgoing or the return path, the initiating host can fall back to regular TCP, as illustrated in Figure 16 and discussed in Section 3.1.

Subflow SYNs contain the MP_JOIN option. If this option is stripped on the outgoing path the SYN will appear to be a regular SYN to host B. Depending on whether there is a listening socket on the target port, host B will reply either with SYN/ACK or RST (subflow connection fails). When host A receives the SYN/ACK it sends a RST because the SYN/ACK does not contain the MP_JOIN option and its token. Either way, the subflow setup fails, but otherwise does not affect the MPTCP connection as a whole.



a) MP_CAPABLE option stripped on outgoing path



b) MP_CAPABLE option stripped on return path

Figure 16: Connection Setup with Middleboxes that Strip Options from Packets

We now examine data flow with MPTCP, assuming the flow is correctly setup, which implies the options in the SYN packets were allowed through by the relevant middleboxes. If options are allowed through and there is no resegmentation or coalescing to TCP segments, multipath TCP flows can proceed without problems.

The case when options get stripped on data packets has been discussed in the Fallback section. If a fraction of options are stripped, behaviour is not deterministic. If some Data Sequence Mappings are lost, the connection can continue so long as mappings exist for the subflow-level data (e.g. if multiple maps have been sent that reinforce each other). If some subflow-level space is left unmapped, however, the subflow is treated as broken and is closed, through the process described in Section 3.6. MPTCP should survive with a loss of some Data ACKs, but performance will degrade as the fraction of stripped options increases. We do not expect such cases to appear in practice, though: most middleboxes will either strip all options or let them all through.

We end this section with a list of middlebox classes, their behaviour and the elements in the MPTCP design that allow operation through such middleboxes. Issues surrounding dropping packets with options or stripping options were discussed above, and are not included here:

- NATs [20] (Network Address (and Port) Translators) change the source address (and often source port) of packets. This means that a host will not know its public-facing address for signaling

in MPTCP. Therefore, MPTCP permits implicit address addition via the MP_JOIN option, and the handshake mechanism ensures that connection attempts to private addresses [17] do not cause problems. Explicit address removal is undertaken by an Address ID to allow no knowledge of the source address.

- Performance Enhancing Proxies (PEPs) [21] might pro-actively ACK data to increase performance. MPTCP, however, relies on accurate congestion control signals from the end host, and non-MPTCP-aware PEPs will not be able to provide such signals. MPTCP will therefore fall back to single-path TCP, or close the problematic subflow (see Section 3.6).

- Traffic Normalizers [22] may not allow holes in sequence numbers, and may cache packets and retransmit the same data. MPTCP looks like standard TCP on the wire, and will not retransmit different data on the same subflow sequence number. In the event of a retransmission, the same data will be retransmitted on the original TCP subflow even if it is additionally retransmitted at the connection-level on a different subflow.

- Firewalls [23] might perform initial sequence number randomization on TCP connections. MPTCP uses relative sequence numbers in data sequence mapping to cope with this. Like NATs, firewalls will not permit many incoming connections, so MPTCP supports address signaling (ADD_ADDR) so that a multi-addressed host can invite its peer behind the firewall/NAT to connect out to its additional interface.

- Intrusion Detection Systems look out for traffic patterns and content that could threaten a network. Multipath will mean that such data is potentially spread, so it is more difficult for an IDS to analyse the whole traffic, and potentially increases the risk of false positives. However, for an MPTCP-aware IDS, tokens can be read by such systems to correlate multiple subflows and re-assemble for analysis.

- Application level middleboxes such as content-aware firewalls may alter the payload within a subflow, such as re-writing URIs in HTTP traffic. MPTCP will detect these using the checksum and close the affected subflow(s), if there are other subflows that can be used. If all subflows are affected multipath will fallback to TCP, allowing such middleboxes to change the payload. MPTCP-aware middleboxes should be able to adjust the payload and MPTCP metadata in order not to break the connection.

In addition, all classes of middleboxes may affect TCP traffic in the following ways:

- o TCP Options may be removed, or packets with unknown options dropped, by many classes of middleboxes. It is intended that the initial SYN exchange, with a TCP Option, will be sufficient to identify the path capabilities. If such a packet does not get through, MPTCP will end up falling back to regular TCP.

- o Segmentation/Coalescing (e.g. TCP segmentation offloading) might copy options between packets and might strip some options. MPTCP's data sequence mapping includes the relative subflow sequence number instead of using the sequence number in the segment. In this way, the mapping is independent of the packets that carry it.

- o The Receive Window may be shrunk by some middleboxes at the subflow level. MPTCP will use the maximum window at data-level, but will also obey subflow specific windows.

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8. IANA Considerations

This document defines a new TCP option for MPTCP, assigned a value of 30 (decimal) from the TCP Option space. This value is the value of "Kind" as seen in all MPTCP options in this document. This value is defined as:

Kind	Length	Meaning	Reference
30	N	Multipath TCP	(This document)

Table 1: TCP Option Kind Numbers

This document also defines a four-bit subtype field, for which IANA is to create and maintain a new sub-registry entitled "MPTCP option subtype values" under the TCP Parameters registry. Initial values for the MPTCP option subtype registry are given below; future assignments are to be defined by Standards Action as defined by [24]. Assignments consist of the MPTCP subtype's symbolic name and its associated value, as per the following table.

Symbol	Name	Reference	Value
MP_CAPABLE	Multipath Capable	Section 3.1	0x0
MP_JOIN	Join Connection	Section 3.2	0x1
DSS	Data Sequence Signal (Data ACK and Data Sequence Mapping)	Section 3.3	0x2
ADD_ADDR	Add Address	Section 3.4.1	0x3
REMOVE_ADDR	Remove Address	Section 3.4.2	0x4
MP_PRIO	Change Subflow Priority	Section 3.3.8	0x5
MP_FAIL	Fallback	Section 3.6	0x6
MP_FASTCLOSE	Fast Close	Section 3.5	0x7

Table 2: MPTCP Option Subtypes

The value 0xf is reserved for Private Use within controlled testbeds.

This document also requests that IANA creates another sub-registry, "MPTCP handshake algorithms" under the TCP Paramers registry, based on the flags in MP_CAPABLE (Section 3.1). The flags consist of eight bits, labelled "A" through "H", and this document assigns the bits as follows, where "(available)" means that the bit is available for future assignment:

Flag Bit	Meaning	Reference
A	Checksum required	This document, Section 3.1
B	Extensibility	This document, Section 3.1
C	(available)	
D	(available)	
E	(available)	
F	(available)	
G	(available)	
H	HMAC-SHA1	This document, Section 3.2

Table 3: MPTCP Handshake Algorithms

Note that the meanings of bits C through H can be dependent upon bit B, depending on how Extensibility is defined in future specifications; see Section 3.1 for more information.

Future assignments in this registry are also to be defined by Standards Action as defined by [24]. Assignments consist of the value of the flags, a symbolic name for the algorithm, and a reference to its specification.

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[24] Narten, T. and H. Alvestrand, "Guidelines for Writing an IANA Considerations Section in RFCs", BCP 26, RFC 5226, May 2008.

Appendix A. Notes on use of TCP Options

The TCP option space is limited due to the length of the Data Offset field in the TCP header (4 bits), which defines the TCP header length in 32 bit words. With the standard TCP header being 20 bytes, this leaves a maximum of 40 bytes for options, and many of these may already be used by options such as timestamp and SACK.

We have performed a brief study on the commonly used TCP options in SYN, data, and pure ACK packets, and found that there is enough room to fit all the options we propose using in this draft.

SYN packets typically include MSS (4 bytes), window scale (3 bytes), SACK permitted (2 bytes) and timestamp (10 bytes) options. Together these sum to 19 bytes. Some operating systems appear to pad each option up to a word boundary, thus using 24 bytes (a brief survey suggests Windows XP and Mac OS X do this, whereas Linux does not). Optimistically, therefore, we have 21 bytes spare, or 16 if it has to be word-aligned. In either case, however, the SYN versions of Multipath Capable (12 bytes) and Join (12 or 16 bytes) options will fit in this remaining space.

TCP data packets typically carry timestamp options in every packet, taking 10 bytes (or 12 with padding). That leaves 30 bytes (or 28, if word-aligned). The Data Sequence Signal (DSS) option varies in length depending on whether the Data Sequence Mapping and DATA_ACK are included, and whether the sequence numbers in use are 4 or 8 octets. The maximum size of the DSS option is 28 bytes, so even that will fit in the available space. But unless a connection is both bi-

directional and high-bandwidth, it is unlikely that all that option space will be required on each DSS option.

Within the DSS option, it is not necessary to include the Data Sequence Mapping and DATA_ACK in each packet, and in many cases it may be possible to alternate their presence (so long as the mapping covers the data being sent in the following packet). It would also be possible to alternate between 4 and 8 byte sequence numbers in each option.

On subflow and connection setup, an MPTCP option is also set on the third packet (an ACK). These are 20 bytes (for Multipath Capable) and 24 bytes (for Join), both of which will fit in the available option space.

Pure ACKs in TCP typically contain only timestamps (10 bytes). Here, multipath TCP typically needs to encode only the DATA_ACK (maximum of 12 bytes). Occasionally ACKs will contain SACK information. Depending on the number of lost packets, SACK may utilize the entire option space. If a DATA_ACK had to be included, then it is probably necessary to reduce the number of SACK blocks to accommodate the DATA_ACK. However, the presence of the DATA_ACK is unlikely to be necessary in a case where SACK is in use, since until at least some of the SACK blocks have been retransmitted, the cumulative data-level ACK will not be moving forward (or if it does, due to retransmissions on another path, then that path can also be used to transmit the new DATA_ACK).

The ADD_ADDR option can be between 8 and 22 bytes, depending on whether IPv4 or IPv6 is used, and whether the port number is present or not. It is unlikely that such signaling would fit in a data packet (although if there is space, it is fine to include it). It is recommended to use duplicate ACKs with no other payload or options in order to transmit these rare signals. Note this is the reason for mandating that duplicate ACKs with MPTCP options are not taken as a signal of congestion.

Finally, there are issues with reliable delivery of options. As options can also be sent on pure ACKs, these are not reliably sent. This is not an issue for DATA_ACK due to their cumulative nature, but may be an issue for ADD_ADDR/REMOVE_ADDR options. Here, it is recommended to send these options redundantly (whether on multiple paths, or on the same path on a number of ACKs - but interspersed with data in order to avoid interpretation as congestion). The cases where options are stripped by middleboxes are discussed in Section 6.

Appendix B. Control Blocks

Conceptually, an MPTCP connection can be represented as an MPTCP control block that contains several variables that track the progress and the state of the MPTCP connection and a set of linked TCP control blocks that correspond to the subflows that have been established.

RFC793 [1] specifies several state variables. Whenever possible, we reuse the same terminology as RFC793 to describe the state variables that are maintained by MPTCP.

B.1. MPTCP Control Block

The MPTCP control block contains the following variable per-connection.

B.1.1. Authentication and Metadata

Local.Token (32 bits): This is the token chosen by the local host on this MPTCP connection. The token MUST be unique among all established MPTCP connections, generated from the local key.

Local.Key (64 bits): This is the key sent by the local host on this MPTCP connection.

Remote.Token (32 bits): This is the token chosen by the remote host on this MPTCP connection, generated from the remote key.

Remote.Key (64 bits): This is the key chosen by the remote host on this MPTCP connection

MPTCP.Checksum (flag): This flag is set to true if at least one of the hosts has set the C bit in the MP_CAPABLE options exchanged during connection establishment, and is set to false otherwise. If this flag is set, the checksum must be computed in all DSS options.

B.1.2. Sending Side

SND.UNA (64 bits): This is the Data Sequence Number of the next byte to be acknowledged, at the MPTCP connection level. This variable is updated upon reception of a DSS option containing a DATA_ACK.

SND.NXT (64 bits): This is the Data Sequence Number of the next byte to be sent. SND.NXT is used to determine the value of the DSN in the DSS option.

SND.WND (32 bits with RFC1323, 16 bits without): This is the sending window. MPTCP maintains the sending window at the MPTCP connection level and the same window is shared by all subflows. All subflows use the MPTCP connection level SND.WND to compute the SEQ.WND value which is sent in each transmitted segment.

B.1.3. Receiving Side

RCV.NXT (64 bits): This is the Data Sequence Number of the next byte which is expected on the MPTCP connection. This state variable is modified upon reception of in-order data. The value of RCV.NXT is used to specify the DATA_ACK which is sent in the DSS option on all subflows.

RCV.WND (32bits with RFC1323, 16 bits otherwise): This is the connection-level receive window, which is the maximum of the RCV.WND on all the subflows.

B.2. TCP Control Blocks

The MPTCP control block also contains a list of the TCP control blocks that are associated to the MPTCP connection.

Note that the TCP control block on the TCP subflows does not contain the RCV.WND and SND.WND state variables as these are maintained at the MPTCP connection level and not at the subflow level.

Inside each TCP control block, the following state variables are defined:

B.2.1. Sending Side

SND.UNA (32 bits): This is the sequence number of the next byte to be acknowledged on the subflow. This variable is updated upon reception of each TCP acknowledgement on the subflow.

SND.NXT (32 bits): This is the sequence number of the next byte to be sent on the subflow. SND.NXT is used to set the value of SEG.SEQ upon transmission of the next segment.

B.2.2. Receiving Side

RCV.NXT (32 bits): This is the sequence number of the next byte which is expected on the subflow. This state variable is modified upon reception of in-order segments. The value of RCV.NXT is copied to the SEG.ACK field of the next segments transmitted on the subflow.

RCV.WND (32 bits with RFC1323, 16 bits otherwise): This is the subflow-level receive window which is updated with the window field from the segments received on this subflow.

Appendix C. Finite State Machine

The diagram in Figure 17 shows the Finite State Machine for connection-level closure. This illustrates how the DATA_FIN connection-level signal (indicated as the DFIN flag on a DATA_ACK) interacts with subflow-level FINs, and permits "break-before-make" handover between subflows.

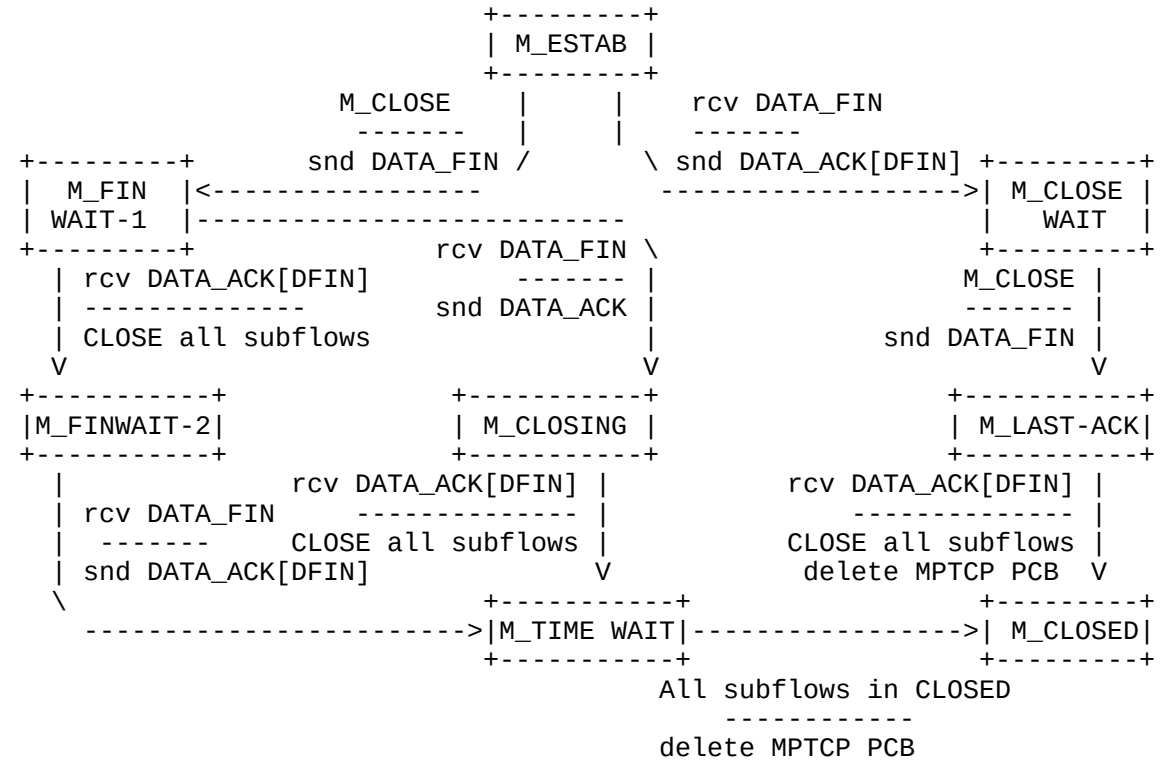

Figure 17: Finite State Machine for Connection Closure

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Threat Analysis for TCP Extensions for Multi-path Operation with Multiple Addresses
draft-ietf-mptcp-threat-08

Abstract

Multi-path TCP (MPTCP for short) describes the extensions proposed for TCP so that endpoints of a given TCP connection can use multiple paths to exchange data. Such extensions enable the exchange of segments using different source-destination address pairs, resulting in the capability of using multiple paths in a significant number of scenarios. Some level of multihoming and mobility support can be achieved through these extensions. However, the support for multiple IP addresses per endpoint may have implications on the security of the resulting MPTCP protocol. This note includes a threat analysis for MPTCP.

Table of Contents

1. Introduction

Multi-path TCP (MPTCP for short) describes the extensions proposed for TCP [RFC0793] so that endpoints of a given TCP connection can use multiple paths to exchange data. Such extensions enable the exchange of segments using different source-destination address pairs, resulting in the capability of using multiple paths in a significant number of scenarios. Some level of multihoming and mobility support can be achieved through these extensions. However, the support for multiple IP addresses per endpoint may have implications on the security of the resulting MPTCP protocol. This note includes a threat analysis for MPTCP. There are there may other ways to provide multiple paths for a TCP connection other than the usage of multiple addresses. The threat analysis performed in this document is limited to the specific case of using multiple addresses per endpoint.

2. Scope

There are multiple ways to achieve Multi-path TCP. Essentially what is needed is for different segments of the communication to be forwarded through different paths by enabling the sender to specify some form of path selector. There are multiple options for such a path selector, including the usage of different next hops, using tunnels to different egress points and so on. In this note, we will focus on a particular approach, namely MPTCP, that rely on the usage of multiple IP address per endpoint and that uses different source-destination address pairs as a mean to express different paths. So, in the rest of this note, the MPTCP expression will refer to this Multi-addressed flavour of Multi-path TCP [I-D.ietf-mptcp-multiaddressed].

In this note we perform a threat analysis for MPTCP. Introducing the support of multiple addresses per endpoint in a single TCP connection may result in additional vulnerabilities compared to single-path TCP. The scope of this note is to identify and characterize these new vulnerabilities. So, the scope of the analysis is limited to the additional vulnerabilities resulting from the multi-address support compared to the current TCP protocol (where each endpoint only has one address available for use per connection). A full analysis of the complete set of threats is explicitly out of the scope. The goal of this analysis is to help the MPTCP protocol designers create an MPTCP specification that is as secure as the current TCP. It is a non-goal of this analysis to help in the design of MPTCP that is more secure than regular TCP.

We will focus on attackers that are not along the path, at least not during the whole duration of the connection. In the current single

path TCP, an on-path attacker can launch a significant number of attacks, including eavesdropping, connection hijacking Man-in-the-Middle attacks and so on. However, it is not possible for the off-path attackers to launch such attacks. There is a middle ground in case the attacker is located along the path for a short period of time to launch the attack and then moves away, but the attack effects still apply. These are the so-called time-shifted attacks. Since these are not possible in today's TCP, we will also consider them as part of the analysis. So, summarizing, we will consider both attacks launched by off-path attackers and time-shifted attacks. Attacks launched by on-path attackers are out of scope, since they also apply to current single-path TCP.

However, that some current on-path attacks may become more difficult with multi-path TCP, since an attacker (on a single path) will not have visibility of the complete data stream.

3. Related work

There is a significant amount of previous work in terms of analysis of protocols that support address agility. In this section we present the most relevant ones and we relate them to the current MPTCP effort.

Most of the problems related to address agility have been deeply analyzed and understood in the context of Route Optimization support in Mobile IPv6 (MIPv6 RO) [RFC3775]. [RFC4225] includes the rational for the design of the security of MIPv6 RO. All the attacks described in the aforementioned analysis apply here and are an excellent basis for our own analysis. The main differences are:

- In MIPv6 RO, the address binding affects all the communications involving an address, while in the MPTCP case, a single connection is at stake. If a binding between two addresses is created at the IP layer, this binding can and will affect all the connections that involve those addresses. However, in MPTCP, if an additional address is added to an ongoing TCP connection, the additional address will/can only affect the connection at hand and not other connections even if the same address is being used for those other connections. The result is that in MPTCP there is much less at stake and the resulting vulnerabilities are less. On the other hand, it is very important to keep the assumption valid that the address bindings for a given connection do not affect other connections. If reusing of binding or security information is to be considered, this assumption could be no longer valid and the full impact of the vulnerabilities must be assessed.

- o In MIPv6 there is a trusted third party, called the Home Agent that can help with some security problems, as expanded in the next bullet.
- o In MIPv6 RO, there is the assumption that the original address (Home Address) through which the connection has been established is always available and in case it is not, the communication will be lost. This is achieved by leveraging in the on the trusted party (the Home Agent) to rely the packets to the current location of the Mobile Node. In MPTCP, it is an explicit goal to provide communication resilience when one of the address pairs is no longer usable, so it is not possible to leverage on the original address pair to be always working.
- o MIPv6 RO is of course designed for IPv6 and it is an explicit goal of MPTCP to support both IPv6 and IPv4. Some MIPv6 RO security solutions rely on the usage of some characteristics of IPv6 (such as the usage of CGAs [RFC3972]), which will not be usable in the context of MPTCP.
- o As opposed to MPTCP, MIPv6 RO does not have a connection state information, including sequence numbers, port numbers that could be leveraged to provide security in some form.

In the Shim6 [RFC5533] design, similar issues related to address agility were considered and a threat analysis was also performed [RFC4218]. The analysis performed for Shim6 also largely applies to the MPTCP context, the main difference being:

- o The Shim6 protocol is a layer 3 protocol so all the communications involving the target address are at stake; in MPTCP, the impact can be limited to a single TCP connection.
- o Similarly to MIPv6 RO, Shim6 only uses IPv6 addresses as identifiers and leverages on some of their properties to provide the security, such as relying on CGAs or HBAs [RFC5535], which is not possible in the MPTCP case where IPv4 addresses must be supported.
- o Similarly to MIPv6 RO, Shim6 does not have a connection state information, including sequence numbers, port that could be leveraged to provide security in some form.

SCTP [RFC4960]is a transport protocol that supports multiple addresses per endpoint and the security implications are very close to the ones of MPTCP. A security analysis, identifying a set of attacks and proposed solutions was performed in [RFC5062]. The results of this analysis apply directly to the case of MPTCP. However, the analysis was performed after the base SCTP protocol was designed and the goal of the document was essentially to improve the security of SCTP. As such, the document is very specific to the actual SCTP specification and relies on the SCTP messages and behaviour to characterize the issues. While some them can be translated to the MPTCP case, some may be caused by specific

behaviour of SCTP.

So, the conclusion is that while we do have a significant amount of previous work that is closely related and we can and will use it as a basis for this analysis, there is a set of characteristics that are specific to MPTCP that grant the need for a specific analysis for MPTCP. The goal of this analysis is to help MPTCP protocol designers to include a set of security mechanisms that prevent the introduction of new vulnerabilities to the Internet due to the adoption of MPTCP.

4. Basic MPTCP.

The goal of this document is to serve as input for MPTCP protocol designers to properly take into account the security issues. As such, the analysis cannot be performed for a specific MPTCP specification, but must be a general analysis that applies to the widest possible set of MPTCP designs. We will characterize what are the fundamental features that any MPTCP protocol must provide and attempt to perform the security implications only assuming those. In some cases, we will have a design choice that will significantly influence the security aspects of the resulting protocol. In that case we will consider both options and try to characterize both designs.

We assume that any MPTCP will behave in the case of a single address per endpoint as TCP. This means that a MPTCP connection will be established by using the TCP 3-way handshake and will use a single address pair.

The addresses used for the establishment of the connection do have a special role in the sense that this is the address used as identifier by the upper layers. The address used as destination address in the SYN packet is the address that the application is using to identify the peer and has been obtained either through the DNS (with or without DNSSEC validation) or passed by a referral or manually introduced by the user. As such, the initiator does have a certain amount of trust in the fact that it is establishing a communication with that particular address. If due to MPTCP, packets end up being delivered to an alternative address, the trust that the initiator has placed on that address would be deceived. In any case, the adoption of MPTCP necessitates a slight evolution of the traditional TCP trust model, in that the initiator is additionally trusting the peer to provide additional addresses which it will trust to the same degree as the original pair. An application or implementation that cannot trust the peer in this way should not make use of multiple paths.

During the 3-way handshake, the sequence number will be synchronized

for both ends, as in regular TCP. We assume that a MPTCP connection will use a single sequence number for the data, even if the data is exchanged through different paths, as MPTCP provides an in-order delivery service of bytes

Once the connection is established, the MPTCP extensions can be used to add addresses for each of the endpoints. This is achieved by each end sending a control message containing the additional address(es). In order to associate the additional address to an ongoing connection, the connection needs to be identified. We assume that the connection can be identified by the 4-tuple of source address, source port, destination address, destination port used for the establishment of the connection. So, at least, the control message that will convey the additional address information can also contain the 4-tuple in order to inform about what connection the address belong to (if no other connection identifier is defined). There are two different ways to convey address information:

- Explicit mode: the control message contain a list of addresses.
- Implicit mode: the address added is the one included in the source address field of the IP header

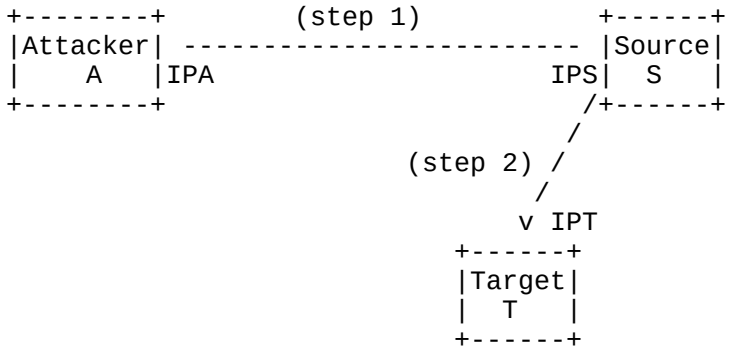
These two modes have different security properties for some type of attacks. The explicit mode seems to be the more vulnerable to abuse. The implicit mode may benefit from forms of ingress filtering security, which would reduce the possibility of an attacker to add any arbitrary address to an ongoing connection. However, ingress filtering deployment is far from universal, and it is unwise to rely on it as a basis for the protection of MPTCP.

Further consideration about the interaction between ingress filtering and implicit mode signaling is needed in the case that we need to remove an address that is no longer available from the MPTCP connection. A host attached to a network that performs ingress filtering and using implicit signaling would not be able to remove an address that is no longer available (either because of a failure or due to a mobility event) from an ongoing MPTCP connection.

We will assume that MPTCP will use all the address pairs that it has available for sending packets and that it will distribute the load based on congestion among the different paths.

5. Flooding attacks

The first type of attacks that are introduced by address agility are the flooding (or bombing) attacks. The setup for this attack is depicted in the following figure:



The scenario consists of an attacker A who has an IP address IPA. A server that can generate a significant amount of traffic (such as a streaming server), called source S and that has IP address IPS. Target T has an IP address IPT.

In step 1 of this attack, the attacker A establishes a MPTCP connection with the source of the traffic server S and starts downloading a significant amount of traffic. The initial connection only involves one IP address per endpoint, IPA and IPS. Once that the download is on course, in the step 2 of the attack is that the attacker A adds IPT as one of the available addresses for the communication. How the additional address is added depends on the MPTCP address management mode. In explicit address management, the attacker A only needs to send a signaling packet conveying address IPT. In implicit mode, the attacker A would need to send a packet with IPT as the source address. Depending on whether ingress filtering is deployed and the location of the attacker, it may be possible or not for the attacker to send such a packet. At this stage, the MPTCP connection still has a single address for the Source S i.e. IPS but has two addresses for the Attacker A, IPA and IPT. The attacker now attempts to get the Source S to send the traffic of the ongoing download to the Target T IP address i.e. IPT. The attacker can do that by pretending that the path between IPA and IPT is congested but that the path between IPS and IPT is not. In order to do that, it needs to send ACKs for the data that flows through the path between IPS and IPT and do not send ACKs for the data that is sent to IPA. The details of this will depend on how the data sent through the different paths is ACKed. One possibility is that ACKs for the data sent using a given address pair should come in packets containing the same address pair. If so, the attacker would need to send ACKs using packets containing IPT as the source address to keep the attack flowing. This may be possible or not depending on the deployment of ingress filtering and the location of the attacker. The attacker would also need to guess the sequence number of the data

being sent to the Target. Once the attacker manages to perform these actions the attack is on place and the download will hit the target. In this type of attacks, the Source S still thinks it is sending packets to the Attacker A while in reality it is sending the packet to Target T.

Once that the traffic from the Source S start hitting the Target T, the target will react. Since the packets are likely to belong to a non existent TCP connection, the Target T will issue RST packets. It is relevant then to understand how MPTCP reacts to incoming RST packets. It seems that the at least the MPTCP that receives a RST packet should terminate the packet exchange corresponding to the particular address pair (maybe not the complete MPTCP connection, but at least it should not send more packets with the address pair involved in the RST packet). However, if the attacker, before redirecting the traffic has managed to increase the window size considerably, the flight size could be enough to impose a significant amount of traffic to the Target node. There is a subtle operation that the attacker needs to achieve in order to launch a significant attack. On the one hand it needs to grow the window enough so that the flight size is big enough to cause enough effect and on the other hand the attacker needs to be able to simulate congestion on the IPA-IPS path so that traffic is actually redirected to the alternative path without significantly reducing the window. This will heavily depend on how the coupling of the windows between the different paths works, in particular how the windows are increased. Some designs of the congestion control window coupling could render this attack ineffective. If the MPTCP protocol requires performing slow start per subflow, then the flooding will be limited by the slow-start initial window size.

Previous protocols, such as MIPv6 RO and SCTP, that have to deal with this type of attacks have done so by adding a reachability check before actually sending data to a new address. The solution used in other protocols, would include the Source S to explicitly asking the host sitting in the new address (the Target T sitting in IPT) whether it is willing to accept packets from the MPTCP connection identified by the 4-tuple IPA, port A, IPS, port S. Since this is not part of the established connection that Target T has, T would not accept the request and Source S would not use IPT to send packets for this MPTCP connection. Usually, the request also includes a nonce that cannot be guessed by the attacker A so that it cannot fake the reply to the request easily. In the case of SCTP, it sends a message with a 64-bit nonce (in a HEARTBEAT).

One possible approach to do this reachability test would be to perform a 3-way handshake for each new address pair that is going to be used in a MPTCP connection. While there are other reasons for

doing this (such as NAT traversal), such approach would also act as a reachability test and would prevent the flooding attacks described in this section.

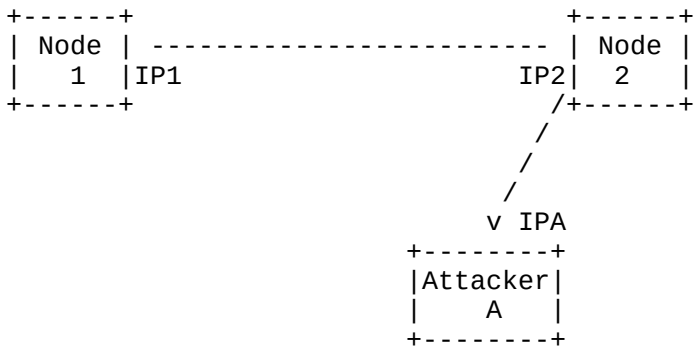
Another type of flooding attack that could potentially be performed with MPTCP is one where the attacker initiates a communication with a peer and includes a long list of alternative addresses in explicit mode. If the peer decides to establish subflows with all the available addresses, the attacker have managed to achieve an amplified attack, since by sending a single packet containing all the alternative addresses it triggers the peer to generate packets to all the destinations.

6. Hijacking attacks

6.1. Hijacking attacks to the Basic MPTCP protocol

The hijacking attacks essentially use the MPTCP address agility to allow an attacker to hijack a connection. This means that the victim of a connection thinks that it is talking to a peer, while it is actually exchanging packets with the attacker. In some sense it is the dual of the flooding attacks (where the victim thinks it is exchanging packets with the attacker but in reality is sending the packets to the target).

The scenario for a hijacking attack is described in the next figure.



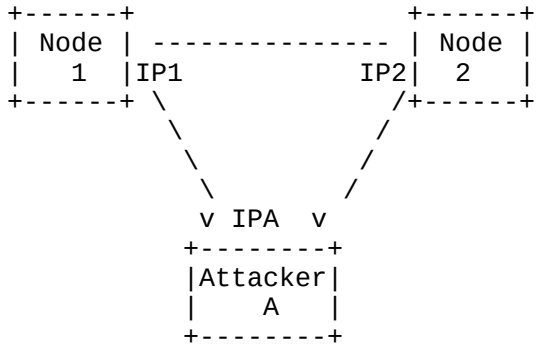
We have a MPTCP connection established between Node 1 and Node 2. The connection is using only one address per endpoint, IP1 and IP2. The attacker then launches the hijacking attack by adding IPA as an additional address for Node 1. There is not much difference between

explicit or implicit address management, since in both cases the Attacker A could easily send a control packet adding the address IPA, either as control data or as the source address of the control packet. In order to be able to hijack the connection, the attacker needs to know the 4-tuple that identifies the connection, including the pair of addresses and the pair of ports. It seems reasonable to assume that knowing the source and destination IP addresses and the port of the server side is fairly easy for the attacker. Learning the port of the client (i.e. of the initiator of the connection) may prove to be more challenging. The attacker would need to guess what the port is or to learn it by intercepting the packets. Assuming that the attacker can gather the 4-tuple and issue the message adding IPA to the addresses available for the MPTCP connection, then the attacker A has been able to participate in the communication. In particular:

- Segments flowing from the Node 2:Depending how the usage of addresses is defined, Node 2 will start using IPA to send data to. In general, since the main goal is to achieve multi-path capabilities, we can assume that unless there are already many IP address pairs in use in the MPTCP connection, Node 2 will start sending data to IPA. This means that part of the data of the communication will reach the Attacker but probably not all of it. This already has negative effects, since Node 1 will not receive all the data from Node 2. Moreover, from the application perspective, this would result in DoS attack, since the byte flow will stop waiting for the missing data. However, it is not enough to achieve full hijacking of the connection, since part of data will be still delivered to IP1, so it would reach Node 1 and not the Attacker. In order for the attacker to receive all the data of the MPTCP connection, the Attacker must somehow remove IP1 of the set of available addresses for the connection. in the case of implicit address management, this operation is likely to imply sending a termination packet with IP1 as source address, which may or not be possible for the attacker depending on whether ingress filtering is in place and the location of the attacker. If explicit address management is used, then the attacker will send a remove address control packet containing IP1. Once IP1 is removed, all the data sent by Node 2 will reach the Attacker and the incoming traffic has been hijacked.
- Segments flowing to the Node 2: As soon as IPA is accepted by Node 2 as part of the address set for the MPTCP connection, the Attacker can send packets using IPA and those packets will be considered by Node 2 as part of MPTCP connection. This means that the attacker will be able to inject data into the MPTCP connection, so from this perspective, the attacker has hijacked part of the outgoing traffic. However, Node 1 would still be able to send traffic that will be received by Node 2 as part of the MPTCP connection. This means that there will be two sources of

data i.e. Node 1 and the attacker, potentially preventing the full hijacking of the outgoing traffic by the attacker. In order to achieve a full hijacking, the attacker would need to remove IP1 from the set of available addresses. This can be done using the same techniques described in the previous paragraph.

A related attack that can be achieved using similar techniques would be a Man-in-the-Middle (MitM) attack. The scenario for the attack is depicted in the figure below.



There is an established connection between Node 1 and Node 2. The Attacker A will use the MPTCP address agility capabilities to place itself as a MitM. In order to do so, it will add IP address IPA as an additional address for the MPTCP connection on both Node 1 and Node 2. This is essentially the same technique described earlier in this section, only that it is used against both nodes involved in the communication. The main difference is that in this case, the attacker can simply sniff the content of the communication that is forwarded through it and in turn forward the data to the peer of the communication. The result is that the attacker can place himself in the middle of the communication and sniff part of the traffic unnoticed. Similar considerations about how the attacker can manage to get to see all the traffic by removing the genuine address of the peer apply.

6.2. Time-shifted hijacking attacks

A simple way to prevent off-path attackers to launch hijacking attacks is to provide security of the control messages that add and remove addresses by the usage of a cookie. In this type of approaches, the peers involved in the MPTCP connection agree on a cookie, that is exchanged in plain text during the establishment of

the connection and that needs to be presented in every control packet that adds or removes an address for any of the peers. The result is that the attacker needs to know the cookie in order to launch any of the hijacking attacks described earlier. This implies that off path attackers can no longer perform the hijacking attacks and that only on-path attackers can do so, so one may consider that a cookie based approach to secure MPTCP connection results in similar security than current TCP. While it is close, it is not entirely true.

The main difference between the security of a MPTCP protocol secured through cookies and the current TCP protocol are the time shifted attacks. As we described earlier, a time shifted attack is one where the attacker is along the path during a period of time, and then moves away but the effects of the attack still remains, after the attacker is long gone. In the case of a MPTCP protocol secured through the usage of cookies, the attacker needs to be along the path until the cookie is exchanged. After the attacker has learnt the cookie, it can move away from the path and can still launch the hijacking attacks described in the previous section.

There are several types of approaches that provide some protection against hijacking attacks and that are vulnerable to some forms of time-shifted attacks. We will next present some general taxonomy of solutions and we describe the residual threats:

- Cookie-based solution: As we described earlier, one possible approach is to use a cookie, that is sent in clear text in every MPTCP control message that adds a new address to the existing connection. The residual threat in this type of solution is that any attacker that can sniff any of these control messages will learn the cookie and will be able to add new addresses at any given point in the lifetime of the connection. Moreover, the endpoints will not detect the attack since the original cookie is being used by the attacker. Summarizing, the vulnerability window of this type of attacks includes all the flow establishment exchanges and it is undetectable by the endpoints.
- Shared secret exchanged in plain text: An alternative option that is more secure than the cookie based approach is to exchange a key in clear text during the establishment of the first subflow and then validate the following subflows by using a keyed HMAC signature using the shared key. This solution would be vulnerable to attackers sniffing the message exchange for the establishment of the first subflow, but after that, the shared key is not transmitted any more, so the attacker cannot learn it through sniffing any other message. Unfortunately, in order to be compatible with NATs (see analysis below) even though this approach includes a keyed HMAC signature, this signature cannot cover the IP address that is being added. This means that this type of approaches are also vulnerable to integrity attacks of the

exchanged messages. This means that even though the attacker cannot learn the shared key by sniffing the subsequent subflow establishment, the attacker can modify the subflow establishment message and change the address that is being added. So, the vulnerability window for confidentially to the shared key is limited to the establishment of the first subflow, but the vulnerability window for integrity attacks still includes all the subflow establishment exchanges. These attacks are still undetectable by the endpoints. The SCTP security falls in this category.

- o Strong crypto anchor exchange. Another approach that could be used would be to exchange some strong crypto anchor while the establishment of the first subflow, such as a public key or a hash chain anchor. Subsequent subflows could be protected by using the crypto material associated to that anchor. An attacker in this case would need to change the crypto material exchanged in the connection establishment phase. As a result the vulnerability window for forging the crypto anchor is limited to the initial connection establishment exchange. Similarly to the previous case, due to NAT traversal considerations, the vulnerability window for integrity attacks include all the subflow establishment exchanges. Because the attacker needs to change the crypto anchor, this approach are detectable by the endpoints, if they communicate directly.

6.3. NAT considerations

In order to be widely adopted MPTCP must work through NATs. NATs are an interesting device from a security perspective. In terms of MPTCP they essentially behave as a Man-in-the-Middle attacker. MPTCP security goal is to prevent from any attacker to insert their addresses as valid addresses for a given MPTCP connection. But that is exactly what a NAT does, they modify the addresses. So, if MPTCP is to work through NATs, MPTCP must accept address rewritten by NATs as valid addresses for a given session. The most direct corollary is that the MPTCP messages that add addresses in the implicit mode (i.e. the SYN of new subflows) cannot be protected against integrity attacks, since they must allow for NATs to change their addresses. This rules out any solution that would rely on providing integrity protection to prevent an attacker from changing the address used in a subflow establishment exchange. This implies that alternative creative mechanisms are needed to protect from integrity attacks to the MPTCP signaling that adds new addresses to a connection. It is far from obvious how one such creative approach could look like at this point.

In the case of explicit mode, you could protect the address included in the MPTCP option. Now the question is what address to include in

the MPTCP option that conveys address information. If the address included is the address configured in the host interface and that interface is behind a NAT, the address information is useless, as the address is not actually reachable from the other end so there is no point in conveying it and even less in securing it. It would be possible to envision the usage of NAT traversal techniques such as STUN to learn the address and port that the NAT has assigned and convey that information in a secure. While this is possible, it relies on using NAT traversal techniques and also tools to convey the address and the port in a secure manner.

7. Recommendation

The presented analysis shows that there is a tradeoff between the complexity of the security solution and the residual threats. In order to define a proper security solution, we need to assess the tradeoff and make a recommendation. After evaluating the different aspects in the MPTCP WG, our conclusion are as follows:

MPTCP should implement some form of reachability check using a random nonce (e.g. TCP 3-way handshake) before adding a new address to an ongoing communication in order to prevent flooding attacks.

The default security mechanisms for MPTCP should be to exchange a key in clear text in the establishment of the first subflow and then secure following address additions by using a keyed HMAC using the exchanged key.

MPTCP security mechanism should support using a pre-shared key to be used in the keyed HMAC, providing a higher level of protection than the previous one.

A mechanism to prevent replay attacks using these messages should be provided e.g. a sequence number protected by the HMAC.

The MPTCP protocol should be extensible and it should able to accommodate multiple security solutions, in order to enable the usage of more secure mechanisms if needed.

8. Security Considerations

This note contains a security analysis for MPTCP, so no further security considerations need to be described in this section.

9. IANA Considerations

This document does not require any action from IANA.

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MPTCP Application Interface Considerations
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Abstract

Multipath TCP (MPTCP) adds the capability of using multiple paths to a regular TCP session. Even though it is designed to be totally backward compatible to applications, the data transport differs compared to regular TCP, and there are several additional degrees of freedom that applications may wish to exploit. This document summarizes the impact that MPTCP may have on applications, such as changes in performance. Furthermore, it discusses compatibility issues of MPTCP in combination with non-MPTCP-aware applications. Finally, the document describes a basic application interface for MPTCP-aware applications that provides access to multipath address information and a level of control equivalent to regular TCP.

Table of Contents

1. Introduction

Multipath TCP adds the capability of using multiple paths to a regular TCP session [1]. The motivations for this extension include increasing throughput, overall resource utilisation, and resilience to network failure, and these motivations are discussed, along with high-level design decisions, as part of the Multipath TCP architecture [4]. The MPTCP protocol [5] offers the same reliable, in-order, byte-stream transport as TCP, and is designed to be backward compatible with both applications and the network layer. It requires support inside the network stack of both endpoints.

This document first presents the impacts that MPTCP may have on applications, such as performance changes compared to regular TCP. Second, it defines the interoperation of MPTCP and applications that are unaware of the multipath transport. MPTCP is designed to be usable without any application changes, but some compatibility issues have to be taken into account. Third, this memo specifies a basic Application Programming Interface (API) for MPTCP-aware applications. The API presented here is an extension to the regular TCP API to allow an MPTCP-aware application the equivalent level of control and access to information of an MPTCP connection that would be possible with the standard TCP API on a regular TCP connection.

An advanced API for MPTCP is outside the scope of this document. Such an advanced API could offer a more fine-grained control over multipath transport functions and policies. The appendix includes a brief, non-compulsory list of potential features of such an advanced API.

The de facto standard API for TCP/IP applications is the "sockets" interface. This document defines experimental MPTCP-specific extensions, using additional socket options. It is up to the applications, high-level programming languages, or libraries to decide whether to use these optional extensions. For instance, an application may want to turn on or off the MPTCP mechanism for certain data transfers, or limit its use to certain interfaces. The syntax and semantics of the specification is in line with the Posix standard [8] as much as possible.

There are also various related extensions of the sockets interface: [12] specifies sockets API extensions for a multihoming shim layer. The API enables interactions between applications and the multihoming shim layer for advanced locator management and for access to information about failure detection and path exploration. Experimental extensions to the sockets API are also defined for the Host Identity Protocol (HIP) [13] in order to manage the bindings of identifiers and locator. Further related API extensions exist for

IPv6 [10], Mobile IP [11], and SCTP [14]. There can be interactions or incompatibilities of these APIs with MPTCP, which are discussed later in this document.

Some network stack implementations, specially on mobile devices, have centralized connection managers or other higher-level APIs to solve multi-interface issues, as surveyed in [16]. Their interaction with MPTCP is outside the scope of this note.

The target readers of this document are application developers whose software may benefit significantly from MPTCP. This document also provides the necessary information for developers of MPTCP to implement the API in a TCP/IP network stack.

2. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [3].

This document uses the terminology introduced in [5].

3. Comparison of MPTCP and Regular TCP

This section discusses the impact that the use of MPTCP will have on applications, in comparison to what may be expected from the use of regular TCP.

3.1. Performance Impact

One of the key goals of adding multipath capability to TCP is to improve the performance of a transport connection by load distribution over separate subflows across potentially disjoint paths. Furthermore, it is an explicit goal of MPTCP that it should not provide a worse performing connection that would have existed through the use of single-path TCP. A corresponding congestion control algorithm is described in [7]. The following sections summarize the performance impact of MPTCP as seen by an application.

3.1.1. Throughput

The most obvious performance improvement that will be gained with the use of MPTCP is an increase in throughput, since MPTCP will pool more than one path (where available) between two endpoints. This will provide greater bandwidth for an application. If there are shared bottlenecks between the flows, then the congestion control algorithms will ensure that load is evenly spread amongst regular and multipath TCP sessions, so that no end user receives worse performance than

single-path TCP.

This performance increase additionally means that an MPTCP session could achieve throughput that is greater than the capacity of a single interface on the device. If any applications make assumptions about interfaces due to throughput (or vice versa), they must take this into account (although an MPTCP implementation must always respect an application's request for a particular interface).

The transport of MPTCP signaling information results in a small overhead. If multiple subflows share a same bottleneck, this overhead slightly reduces the capacity that is available for data transport. Yet, this potential reduction of throughput will be neglectible in many usage scenarios, and the protocol contains optimisations in its design so that this overhead is minimal.

3.1.2. Delay

If the delays on the constituent subflows of an MPTCP connection differ, the jitter perceivable to an application may appear higher as the data is spread across the subflows. Although MPTCP will ensure in-order delivery to the application, the application must be able to cope with the data delivery being burstier than may be usual with single-path TCP. Since burstiness is commonplace on the Internet today, it is unlikely that applications will suffer from such an impact on the traffic profile, but application authors may wish to consider this in future development.

In addition, applications that make round trip time (RTT) estimates at the application level may have some issues. Whilst the average delay calculated will be accurate, whether this is useful for an application will depend on what it requires this information for. If a new application wishes to derive such information, it should consider how multiple subflows may affect its measurements, and thus how it may wish to respond. In such a case, an application may wish to express its scheduling preferences, as described later in this document.

3.1.3. Resilience

The use of multiple subflows simultaneously means that, if one should fail, all traffic will move to the remaining subflow(s), and additionally any lost packets can be retransmitted on these subflows.

Subflow failure may be caused by issues within the network, which an application would be unaware of, or interface failure on the node. An application may, under certain circumstances, be in a position to be aware of such failure (e.g. by radio signal strength, or simply an

interface enabled flag), and so must not make assumptions of an MPTCP flow's stablity based on this. An MPTCP implementation must never override an application's request for a given interface, however, so the cases where this issue may be applicable are limited.

3.2. Potential Problems

3.2.1. Impact of Middleboxes

MPTCP has been designed in order to pass through the majority of middleboxes. Empirical evidence suggests that new TCP options can successfully be used on most paths in the Internet. Nevertheless some middleboxes may still refuse to pass MPTCP messages due to the presence of TCP options, or they may strip TCP options. If this is the case, MPTCP should fall back to regular TCP. Although this will not create a problem for the application (its communication will be set up either way), there may be additional (and indeed, user-perceivable) delay while the first handshake fails. A detailed discussion of the various fallback mechanisms, for failures occurring at different points in the connection, is presented in [5].

There may also be middleboxes that transparently change the length of content. If such middleboxes are present, MPTCP's reassembly of the byte stream in the receiver is difficult. Still, MPTCP can detect such middleboxes and then fall back to regular TCP. An overview of the impact of middleboxes is presented in [4] and MPTCP's mechanisms to work around these are presented and discussed in [5].

MPTCP can also have other unexpected implications. For instance, intrusion detection systems could be triggered. A full analysis of MPTCP's impact on such middleboxes is for further study after deployment experiments.

3.2.2. Outdated Implicit Assumptions

In regular TCP, there is a one-to-one mapping of the socket interface to a flow through a network. Since MPTCP can make use of multiple flows, applications cannot implicitly rely on this one-to-one mapping any more. Applications that require the transport along a single path can disable the use of MPTCP as described later in this document. Examples include monitoring tools that want to measure the available bandwidth on a path, or routing protocols such as BGP that require the use of a specific link.

3.2.3. Security Implications

The support for multiple IP addresses within one MPTCP connection can result in additional security vulnerabilities, such as possibilities

for attackers to hijack connections. The protocol design of MPTCP minimizes this risk. An attacker on one of the paths can cause harm, but this is hardly an additional security risk compared to single-path TCP, which is vulnerable to man-in-the-middle attacks, too. A detailed thread analysis of MPTCP is published in [6].

4. Operation of MPTCP with Legacy Applications

4.1. Overview of the MPTCP Network Stack

MPTCP is an extension of TCP, but it is designed to be backward compatible for legacy applications. TCP interacts with other parts of the network stack by different interfaces. The de facto standard API between TCP and applications is the sockets interface. The position of MPTCP in the protocol stack can be illustrated in Figure 1.

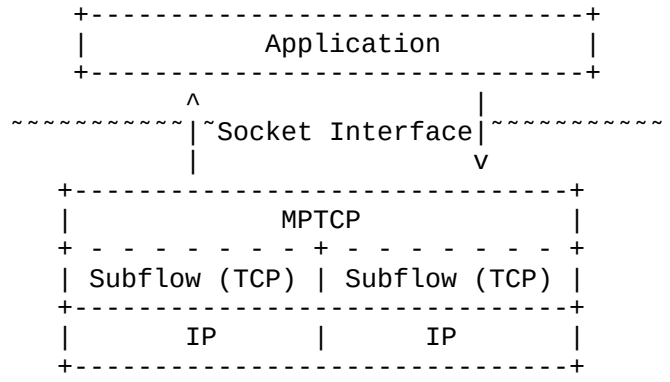

Figure 1: MPTCP protocol stack

In general, MPTCP can affect all interfaces that make assumptions about the coupling of a TCP connection to a single IP address and TCP port pair, to one sockets endpoint, to one network interface, or to a given path through the network.

This means that there are two classes of applications:

- Legacy applications: These applications are unaware of MPTCP and use the existing API towards TCP without any changes. This is the default case.

- MPTCP-aware applications: These applications indicate support for an enhance MPTCP interface. This document specified a minimum set of API extensions for such applications.

In the following, it is discussed to which extent MPTCP affects legacy applications using the existing sockets API. The existing sockets API implies that applications deal with data structures that store, amongst others, the IP addresses and TCP port numbers of a TCP connection. A design objective of MPTCP is that legacy applications can continue to use the established sockets API without any changes. However, in MPTCP there is a one-to-many mapping between the socket endpoint and the subflows. This has several subtle implications for legacy applications using sockets API functions.

4.2. Address Issues

4.2.1. Specification of Addresses by Applications

During binding, an application can either select a specific address, or bind to INADDR_ANY. Furthermore, on some systems other socket options (e.g., SO_BINDTODEVICE) can be used to bind to a specific interface. If an application uses a specific address or binds to a specific interface, then MPTCP MUST respect this and not interfere in the application's choices. If an application binds to INADDR_ANY, it is assumed that the application does not care which addresses to use locally. In this case, a local policy MAY allow MPTCP to automatically set up multiple subflows on such a connection.

The basic sockets API of MPTCP-aware applications allows to express further preferences in an MPTCP-compatible way (e.g. bind to a subset of interfaces only).

4.2.2. Querying of Addresses by Applications

Applications can use the getpeername() or getsockname() functions in order to retrieve the IP address of the peer or of the local socket. These functions can be used for various purposes, including security mechanisms, geo-location, or interface checks. The socket API was designed with an assumption that a socket is using just one address, and since this address is visible to the application, the application may assume that the information provided by the functions is the same during the lifetime of a connection. However, in MPTCP, unlike in TCP, there is a one-to-many mapping of a connection to subflows, and subflows can be added and removed while the connections continues to exist. Therefore, MPTCP cannot expose addresses by getpeername() or getsockname() that are both valid and constant during the connection's lifetime.

This problem is addressed as follows: If used by a legacy application, the MPTCP stack MUST always return the addresses of the first subflow of an MPTCP connection, in all circumstances, even if that particular subflow is no longer in use.

As this address may not be valid any more if the first subflow is closed, the MPTCP stack MAY close the whole MPTCP connection if the first subflow is closed (i.e. fate sharing between the initial subflow and the MPTCP connection as a whole). Whether to close the whole MPTCP connection by default SHOULD be controlled by a local policy. Further experiments are needed to investigate its implications.

The functions getpeername() and getsockname() SHOULD also always return the addresses of the first subflow if the socket is used by an MPTCP-aware application, in order to be consistent with MPTCP-unaware applications, and, e. g., also with SCTP. Instead of getpeername() or getsockname(), MPTCP-aware applications can use new API calls, documented later, in order to retrieve the full list of address pairs for the subflows in use.

4.3. Socket Option Issues

4.3.1. General Guideline

The existing sockets API includes options that modify the behavior of sockets and their underlying communications protocols. Various socket options exist on socket, TCP, and IP level. The value of an option can usually be set by the setsockopt() system function. The getsockopt() function gets information. In general, the existing sockets interface functions cannot configure each MPTCP subflow individually. In order to be backward compatible, existing APIs therefore SHOULD apply to all subflows within one connection, as far as possible.

4.3.2. Disabling of the Nagle Algorithm

One commonly used TCP socket option (TCP_NODELAY) disables the Nagle algorithm as described in [2]. This option is also specified in the Posix standard [8]. Applications can use this option in combination with MPTCP exactly in the same way. It then SHOULD disable the Nagle algorithm for the MPTCP connection, i.e., all subflows.

In addition, the MPTCP protocol instance MAY use a different path scheduler algorithm if TCP_NODELAY is present. For instance, it could use an algorithm that is optimized for latency-sensitive traffic. Specific algorithms are outside the scope of this document.

4.3.3. Buffer Sizing

Applications can explicitly configure send and receive buffer sizes by the sockets API (SO_SNDBUF, SO_RCVBUF). These socket options can also be used in combination with MPTCP and then affect the buffer

size of the MPTCP connection. However, when defining buffer sizes, application programmers should take into account that the transport over several subflows requires a certain amount of buffer for resequencing in the receiver. MPTCP may also require more storage space in the sender, in particular, if retransmissions are sent over more than one path. In addition, very small send buffers may prevent MPTCP from efficiently scheduling data over different subflows. Therefore, it does not make sense to use MPTCP in combination with small send or receive buffers.

An MPTCP implementation MAY set a lower bound for send and receive buffers and treat a small buffer size request as an implicit request not to use MPTCP.

4.3.4. Other Socket Options

Some network stacks also provide other implementation-specific socket options or interfaces that affect TCP's behavior. If a network stack supports MPTCP, it must be ensured that these options do not interfere.

4.4. Default Enabling of MPTCP

It is up to a local policy at the end system whether a network stack should automatically enable MPTCP for sockets even if there is no explicit sign of MPTCP awareness of the corresponding application. Such a choice may be under the control of the user through system preferences.

The enabling of MPTCP, either by application or by system defaults, does not necessarily mean that MPTCP will always be used. Both endpoints must support MPTCP, and there must be multiple addresses at at least one endpoint, for MPTCP to be used. Even if those requirements are met, however, MPTCP may not be immediately used on a connection. It may make sense for multiple paths to be brought into operation only after a given period of time, or if the connection is saturated.

4.5. Summary of Advices to Application Developers

- Using the default MPTCP configuration: Like TCP, MPTCP is designed to be efficient and robust in the default configuration. Application developers should not explicitly configure TCP (or MPTCP) features unless this is really needed.

- Socker buffet dimensioning: Multipath transport requires larger buffers in the receiver for resequencing, as already explained. Applications should use reasonably buffer sizes (such as the

operating system default values) in order to fully benefit from MPTCP. A full discussion of buffer sizing issues is given in [5].

- o Facilitating stack-internal heuristics: The path management and data scheduling by MPTCP is realized by stack-internal algorithms that may implicitly try to self-optimize their behavior according to assumed application needs. For instance, an MPTCP implementation may use heuristics to determine whether an application requires delay-sensitive or bulk data transport, using for instance port numbers, the TCP_NODELAY socket options, or the application's read/write patterns as input parameters. An application developer can facilitate the operation of such heuristics by avoiding atypical interface use cases. For instance, for long bulk data transfers, it does neither make sense to enable the TCP_NODELAY socket option, nor is it reasonable to use many small subsequent socket "send()" calls with small amounts of data only.

5. Basic API for MPTCP-aware Applications

5.1. Design Considerations

While applications can use MPTCP with the unmodified sockets API, multipath transport results in many degrees of freedom. MPTCP manages the data transport over different subflows automatically. By default, this is transparent to the application, but an application could use an additional API to interface with the MPTCP layer and to control important aspects of the MPTCP implementation's behaviour.

This document describes a basic MPTCP API. The API uses non-mandatory socket options and only includes a minimum set of functions that provide an equivalent level of control and information as exists for regular TCP. It maintains backward compatibility with legacy applications.

An advanced MPTCP API is outside the scope of this document. The basic API does not allow a sender or a receiver to express preferences about the management of paths or the scheduling of data, even if this can have a significant performance impact and if an MPTCP implementation could benefit from additional guidance by applications. A list of potential further API extensions is provided in the appendix. The specification of such an advanced API is for further study and may partly be implementation-specific.

MPTCP mainly affects the sending of data. Therefore, the basic API only affects the sender side of a data transfer. A receiver may also have preferences about data transfer choices, and it may have performance requirements, too. Yet, the signaling of the receiver's

needs is outside of the scope of this document.

As this document specifies sockets API extensions, it is written so that the syntax and semantics are in line with the Posix standard [8] as much as possible.

5.2. Requirements on the Basic MPTCP API

Because of the importance of the sockets interface there are several fundamental design objectives for the basic interface between MPTCP and applications:

- Consistency with existing sockets APIs must be maintained as far as possible. In order to support the large base of applications using the original API, a legacy application must be able to continue to use standard socket interface functions when run on a system supporting MPTCP. Also, MPTCP-aware applications should be able to access the socket without any major changes.

- Sockets API extensions must be minimized and independent of an implementation.

- The interface should both handle IPv4 and IPv6.

The following is a list of the core requirements for the basic API:

REQ1: Turn on/off MPTCP: An application should be able to request to turn on or turn off the usage of MPTCP. This means that an application should be able to explicitly request the use of MPTCP if this is possible. Applications should also be able to request not to enable MPTCP and to use regular TCP transport instead. This can be implicit in many cases, since MPTCP must disabled by the use of binding to a specific address. MPTCP may also be enabled if an application uses a dedicated multipath address family (such as AF_MULTIPATH, [9]).

REQ2: An application should be able to restrict MPTCP to binding to a given set of addresses.

REQ3: An application should be able obtain information on the addresses used by the MPTCP subflows.

REQ4: An application should be able to extract a unique identifier for the connection (per endpoint).

The first requirement is the most important one, since some applications could benefit a lot from MPTCP, but there are also cases

in which it hardly makes sense. The existing sockets API provides similar mechanisms to enable or disable advanced TCP features. The second requirement corresponds to the binding of addresses with the bind() socket call, or, e.g., explicit device bindings with a SO_BINDTODEVICE option. The third requirement ensures that there is an equivalent to getpeername() or getsockname() that is able to deal with more than one subflow. Finally, it should be possible for the application to retrieve a unique connection identifier (local to the endpoint on which it is running) for the MPTCP connection. This is equivalent to using the (address, port) pair for a connection identifier in single-path TCP, which is no longer static in MPTCP.

An application can continue to use getpeername() or getsockname() in addition to the basic MPTCP API. In that case, both functions return the corresponding addresses of the first subflow, as already explained.

5.3. Sockets Interface Extensions by the Basic MPTCP API

5.3.1. Overview

The basic MPTCP API consist of four new socket options that are specific to MPTCP. All of these socket options are defined at TCP level (IPPROTO_TCP).

- TCP_MULTIPATH_ENABLE: Enable/disable MPTCP
- TCP_MULTIPATH_BIND: Bind MPTCP to a set of given local addresses
- TCP_MULTIPATH_SUBFLOWS: Get the addresses currently used by the MPTCP subflows
- TCP_MULTIPATH_CONNID: Get the local connection identifier for this MPTCP connection

Table Table 1 shows a list of the socket options for the general configuration of MPTCP. The first column gives the name of the option. The second and third columns indicate whether the option can be handled by the getsockopt() system call and/or by the setsockopt() system call. The fourth column lists the type of data structure specified along with the socket option.

Option name	Get	Set	Data type
TCP_MULTIPATH_ENABLE	o	o	int
TCP_MULTIPATH_BIND		o	list of "struct sockaddr"
TCP_MULTIPATH_SUBFLOWS	o		list of pairs of "struct sockaddr"
TCP_MULTIPATH_CONNID	o		uint32

Table 1: Socket options for MPTCP

There are restrictions when these new socket options can be used:

- o TCP_MULTIPATH_ENABLE: This option SHOULD only be set before the establishment of a TCP connection. Its value SHOULD only be read after the establishment of a connection.

- o TCP_MULTIPATH_BIND: This option MAY be both applied before connection setup or during a connection. In the latter case, it allows MPTCP to use a new address, if there has been a restriction before connection setup.

- o TCP_MULTIPATH_SUBFLOWS: This option is read-only and SHOULD only be used after connection setup.

- o TCP_MULTIPATH_CONNID: This option is read-only and SHOULD only be used after connection setup.

5.3.2. Enabling and Disabling of MPTCP

An application can explicitly indicate multipath capability by setting the TCP_MULTIPATH_ENABLE option with a value larger than 0. In this case, the MPTCP implementation SHOULD try to negitiate MPTCP for that connection. Note that multipath transport will not necessarily be enabled, as it requires multiple addresses and support in the other end-system and potentially also on middleboxes.

An application can disable MPTCP setting the option with a value of 0. In that case, MPTCP MUST NOT be used on that connection.

After connection establishment, an application can get the value of the TCP_MULTIPATH_ENABLE option. A value of 0 then means lack of MPTCP support. Any value equal to or larger than 1 means that MPTCP is supported.

As alternative to setting a socket option, an application can also use a new, separate address family called AF_MULTIPATH [9]. This

separate address family can be used to exchange multiple addresses between an application and the standard sockets API, and additionally acts as an explicit indication that an application is MPTCP-aware, i.e., that it can deal with the semantic changes of the sockets API, in particular concerning getpeername() and getsockname(). The usage of AF_MULTIPATH is also more flexible with respect to multipath transport, either IPv4 or IPv6, or both in parallel [9].

5.3.3. Binding MPTCP to Specified Addresses

An application can set the TCP_MULTIPATH_BIND socket option to announce a set of local IP addresses that MPTCP may bind to. The parameter of the option is a list of data structures of type "sockaddr". A MPTCP implementation must iterate over this list since the length of the structures may vary and will be deteremined by the address families. By extension, this option will also control the list of addresses that can be advertised to the peer via MPTCP signalling.

If used during the lifetime of a connection, an application MUST always provide the full list of addresses that MPTCP is allowed to use. If the option is set, MPTCP MUST only establish subflows using one of the addresses in that list as source addresses. MPTCP MUST also use the list as the only set of addresses it can signal to its peer. It should be noted that this signal is only a hint, and an MPTCP implementation may only use a subset of the addresses.

If an address is not present in the new list, MPTCP MUST close any corresponding subflows (i.e. those using the local address that is no longer present), and signal the removal of the address to the peer. If alternative paths are available using the supplied address list but MPTCP is not currently using them, an MPTCP implementation SHOULD establish alternative subflows before undertaking the address removal.

TBD: If it is unreasonable or difficult for an application to keep track of addresses to provide full lists for every time TCP_MULTIPATH_BIND is set, we could also provide separate TCP_MULTIPATH_ADDR_ADD and TCP_MULTIPATH_ADDR_REMOVE options. Would this be preferable? (The ADD option would provide the same functionality as bind() before connection setup.)

5.3.4. Querying the MPTCP Subflow Addresses

An application can get a list of the addresses used by the currently established subflows by means of the TCP_MULTIPATH_SUBFLOWS option, which cannot be set. The return value is a list of pairs of "sockaddr" data structures. In one pair, the first data structure

refers to the local IP address and the second one to the remote IP address used by the subflow. The list MUST only include established subflows.

The length of the data structure depends on the number of subflows, and so an application must iterate over the list for its length, determining the length of each "sockaddr" data structure by its address family.

5.3.5. Getting a Unique Connection Identifier

An application that wants a unique identifier for the connection, analogous to an (address, port) pair in regular TCP, can use the TCP_MULTIPATH_CONNID option to get a local connection identifier for the MPTCP connection.

This is a 32-bit number, and SHOULD be the same as the local connection identifier sent in the MPTCP handshake.

5.4. Usage Examples

TODO: Example C code for the API functions

6. Other Compatibility Issues

6.1. Usage of the SCTP Socket API

For dealing with multi-homing, several socket API extensions have been defined for SCTP [14]. As MPTCP realizes multipath transport from and to multi-homed endsystems, some of these interface function calls are actually applicable to MPTCP in a similar way.

The following functions that are defined for SCTP have similar functionality to the MPTCP API extensions defined earlier:

- sctp_bindx()
- sctp_connectx()
- sctp_getladdrs()
- sctp_getpaddrs()

The syntax and semantics of these functions are described in [14].

API developers MAY wish to integrate SCTP and MPTCP calls to provide a consistent interface to the application. Yet, it must be emphasized that the transport service provided by MPTCP is different

to SCTP, and this is why not all SCTP API functions can be mapped directly to MPTCP. Furthermore, a network stack implementing MPTCP does not necessarily support SCTP and its specific socket interface extensions. This is why the basic API of MPTCP defines additional socket options only, which are a backward compatible extension of TCP's application interface.

6.2. Incompatibilities with other Multihoming Solutions

The use of MPTCP can interact with various related sockets API extensions. The use of a multihoming shim layer conflicts with multipath transport such as MPTCP or SCTP [12]. Care should be taken for the usage not to confuse with the overlapping features of other APIs:

- SHIM API [12]: This API specifies sockets API extensions for the multihoming shim layer.
- HIP API [13]: The Host Identity Protocol (HIP) also results in a new API.
- API for Mobile IPv6 [11]: For Mobile IPv6, a significantly extended socket API exists as well.

In order to avoid any conflict, multiaddressed MPTCP SHOULD NOT be enabled if a network stack uses SHIM6, HIP, or Mobile IPv6. Furthermore, applications should not try to use both the MPTCP API and another multihoming or mobility layer API.

It is possible, however, that some of the MPTCP functionality, such as congestion control, could be used in a SHIM6 or HIP environment. Such operation is outside the scope of this document.

6.3. Interactions with DNS

In multihomed or multiaddressed environments, there are various issues that are not specific to MPTCP, but have to be considered, too. These problems are summarized in [15].

Specifically, there can be interactions with DNS. Whilst it is expected that an application will iterate over the list of addresses returned from a call such as getaddrinfo(), MPTCP itself MUST NOT make any assumptions about multiple A or AAAA records from the same DNS query referring to the same host, as it is very likely that multiple addresses refer to multiple servers for load balancing purposes.

TODO: Elaborate on DNS

7. Security Considerations

Will be added in a later version of this document.

8. IANA Considerations

No IANA considerations.

9. Conclusion

This document discusses MPTCP's application implications and specifies a basic MPTCP API. For legacy applications, it is ensured that the existing sockets API continues to work. MPTCP-aware applications can use the basic MPTCP API that provides some control over the transport layer equivalent to regular TCP. A more fine-granular interaction between applications and MPTCP requires an advanced MPTCP API, which is not specified in this document.

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Appendix A. Requirements on a Future Advanced MPTCP API

A.1. Design Considerations

Multipath transport results in many degrees of freedom. The basic MPTCP API only defines a minimum set of the sockets API extensions for the interface between the MPTCP layer and applications, which does not offer much control of the MPTCP implementation's behaviour. A future, advanced API could address further features of MPTCP and provide more control.

Applications that use TCP may have different requirements on the transport layer. While developers have become used to the characteristics of regular TCP, new opportunities created by MPTCP could allow the service provided to be optimised further. An advanced API could enable MPTCP-aware applications to specify preferences and control certain aspects of the behavior, in addition to the simple control provided by the basic interface. An advanced API could also address aspects that are completely out-of-scope of the basic API, for example, the question whether a receiving application could influence the sending policy.

Furthermore, an advanced MPTCP API could be part of a new overall interface between the network stack and applications that addresses other issues as well, such as the split between identifiers and locators. An API that does not use IP addresses (but, instead e.g. a connectbyname() function) would be useful for numerous purposes, independent of MPTCP.

This appendix documents a list of potential usage scenarios and requirements for the advanded API. The specification and implementation of a corresponding API is outside the scope of this document.

A.2. MPTCP Usage Scenarios and Application Requirements

There are different MPTCP usage scenarios. An application that wishes to transmit bulk data will want MPTCP to provide a high throughput service immediately, through creating and maximising utilisation of all available subflows. This is the default MPTCP use case.

But at the other extreme, there are applications that are highly interactive, but require only a small amount of throughput, and these

are optimally served by low latency and jitter stability. In such a situation, it would be preferable for the traffic to use only the lowest latency subflow (assuming it has sufficient capacity), maybe with one or two additional subflows for resilience and recovery purposes. The key challenge for such a strategy is that the delay on a path may fluctuate significantly and that just always selecting the path with the smallest delay might result in instability.

The choice between bulk data transport and latency-sensitive transport affects the scheduler in terms of whether traffic should be, by default, sent on one subflow or across several ones. Even if the total bandwidth required is less than that available on an individual path, it is desirable to spread this load to reduce stress on potential bottlenecks, and this is why this method should be the default for bulk data transport. However, that may not be optimal for applications that require latency/jitter stability.

In the case of the latter option, a further question arises: Should additional subflows be used whenever the primary subflow is overloaded, or only when the primary path fails (hot-standby)? In other words, is latency stability or bandwidth more important to the application? This results in two different options: Firstly, there is the single path which can overflow into an additional subflow; and secondly there is single-path with hot-standby, whereby an application may want an alternative backup subflow in order to improve resilience. In case that data delivery on the first subflow fails, the data transport could immediately be continued on the second subflow, which is idle otherwise.

A further, mostly orthogonal question is whether data should be duplicated over the different subflows, in particular if there is spare capacity. This could improve both the timeliness and reliability of data delivery.

In summary, there are at least three possible performance objectives for multipath transport (not necessarily disjoint):

1. High bandwidth

2. Low latency and jitter stability

3. High reliability

In an advanced API, applications could provide high-level guidance to the MPTCP implementation concerning these performance requirements, for instance, which is considered to be the most important one. The MPTCP stack would then use internal mechanisms to fulfill this abstract indication of a desired service, as far as possible. This

would both affect the assignment of data (including retransmissions) to existing subflows (e.g., 'use all in parallel', 'use as overflow', 'hot standby', 'duplicate traffic') as well as the decisions when to set up additional subflows to which addresses. In both cases different policies can exist, which can be expected to be implementation-specific.

Therefore, an advanced API could provide a mechanism how applications can specify their high-level requirements in an implementation-independent way. One possibility would be to select one "application profile" out of a number of choices that characterize typical applications. Yet, as applications today do not have to inform TCP about their communication requirements, it requires further studies whether such an approach would be realistic.

Of course, independent of an advanced API, such functionality could also partly be achieved by MPTCP-internal heuristics that infer some application preferences e.g. from existing socket options, such as TCP_NODELAY. Whether this would be reliable, and indeed appropriate, is for further study, too.

A.3. Potential Requirements on an Advanced MPTCP API

The following is a list of potential requirements for an advanced MPTCP API beyond the features of the basic API. It is included here for information only:

REQ5: An application should be able to establish MPTCP connections without using IP addresses as locators.

REQ6: An application should be able obtain usage information and statistics about all subflows (e.g., ratio of traffic sent via this subflow).

REQ7: An application should be able to request a change in the number of subflows in use, thus triggering removal or addition of subflows. An even finer control granularity would be a request for the establishment of a new subflow to a provided destination, or a request for the termination of a specified, existing subflow.

REQ8: An application should be able to inform the MPTCP implementation about its high-level performance requirements, e.g., in form of a profile.

REQ9: An application should be able to control the automatic establishment/termination of subflows. This would imply a selection among different heuristics of the path manager, e.g., 'try as soon as possible', 'wait until there is a bunch of data', etc.

REQ10: An application should be able to set preferred subflows or subflow usage policies. This would result in a selection among different configurations of the multipath scheduler.

REQ11: An application should be able to control the level of redundancy by telling whether segments should be sent on more than one path in parallel.

An advanced API fulfilling these requirements would allow application developers to more specifically configure MPTCP. It could avoid suboptimal decisions of internal, implicit heuristics. However, it is unclear whether all of these requirements would have a significant benefit to applications, since they are going above and beyond what the existing API to regular TCP provides.

Appendix B. Change History of the Document

Changes compared to version 02:

- Definition of the behavior of getpeername() and getsockname() when being called by an MPTCP-aware application.
- Discussion of the possiblity that an MPTCP implementation could support the SCTP API, as far as it is applicable to MPTCP.
- Various editorial fixes.

Changes compared to version 01:

- Second half of the document completely restructured
- Separation between a basic API and an advanced API: The focus of the document is the basic API only; all text concerning a potential extended API is moved to the appendix
- Several clarifications, e. g., concerning buffer sizeing and the use of different scheduling strategies triggered by TCP_NODELAY
- Additional references

Changes compared to version 00:

- Distinction between legacy and MPTCP-aware applications
- Guidance concerning default enabling, reaction to the shutdown of the first subflow, etc.
- Reference to a potential use of AF_MULTIPATH
- Additional references to related work

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