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A. Ford
Pexip
C. Raiciu
U. Politechnica of Bucharest
M. Handley
U. College London
O. Bonaventure
U. catholique de Louvain
C. Paasch
Apple, Inc.
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TCP Extensions for Multipath Operation with Multiple Addresses
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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 it provides the components necessary to establish and use multiple TCP flows across potentially disjoint paths.

This document specifies v1 of Multipath TCP, obsoleting v0 as specified in RFC6824 [RFC6824] through clarifications and modifications primarily driven by deployment experience.

Status of This Memo

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1. Introduction

Multipath TCP (MPTCP) is a set of extensions to regular TCP [RFC0793] to provide a Multipath TCP [RFC6182] 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:

- o Architecture [RFC6182], 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.

- o Congestion control [RFC6356] presents a safe congestion control algorithm for coupling the behavior of the multiple paths in order to "do no harm" to other network users.

- o Application considerations [RFC6897] discusses 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.

This document is an update to, and obsoletes, the v0 specification of Multipath TCP [RFC6824]. This document specifies MPTCP v1, which is not backward compatible with MPTCP v0. This document additionally defines version negotiation procedures for implementations that support both versions.

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:

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

- o 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 utilization and resilience to a subset of node failures. The congestion control algorithms defined in [RFC6356] 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 Equal-Cost Multipath (ECMP) implementations

[RFC2992]), 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 [RFC6182]):

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.

Fallback: 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 [RFC6897] discusses the necessary features of an API to provide backwards-compatibility, as well as API extensions to convey the behavior 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 [RFC6182] and in [howhard].

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 [RFC6897].

```
                                 +-------------------------------+
                                 |           Application         |
    +---------------+            +-------------------------------+
    |  Application  |            |             MPTCP             |
    +---------------+            + - - - - - - - + - - - - - - - +
    |      TCP      |            | Subflow (TCP) | Subflow (TCP) |
    +---------------+            +-------------------------------+
    |      IP       |            |       IP      |      IP       |
    +---------------+            +-------------------------------+
```

Figure 1: Comparison of Standard TCP and MPTCP Protocol Stacks

1.3. Terminology

This document makes use of a number of terms that 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 similar 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: An 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.

- o To a non-MPTCP-aware application, MPTCP will behave the same as normal TCP. Extended APIs could provide additional control to MPTCP-aware applications [RFC6897]. An application begins by opening a TCP socket in the normal way. MPTCP signaling and operation are handled by the MPTCP implementation.

- o 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.

- o 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.

- o 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.

- o 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.

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

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

```
             Host A                               Host B
      ------------------------          ------------------------
      Address A1    Address A2          Address B1    Address B2
      ----------    ----------          ----------    ----------
          |             |                   |             |
          |     (initial connection setup)  |             |
          |-------------------------------->|             |
          |<--------------------------------|             |
          |             |                   |             |
          |            (additional subflow setup)         |
          |             |------------------>|             |
          |             |<------------------|             |
          |             |                   |             |
          |             |                   |             |
```

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 [RFC2119].

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 that follows in Section 3.

A Multipath TCP connection provides a bidirectional 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 subflow 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 initial ACK packets also carry the MP_CAPABLE option. This option is variable length and serves multiple purposes. Firstly, it verifies whether the remote host supports Multipath TCP; 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                ->
[flags]
                          <-            MP_CAPABLE
                                        [B's key, flags]
ACK + MP_CAPABLE (+ data) ->
[A's key, B's key, flags, (data-level details)]
```

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 set up. 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 Hash-based Message Authentication Code (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 are 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/port pair IP#-A1 and wants to open a second subflow starting at address/port pair 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/port pair (IP#-A2). Host B may later send an MP_JOIN to this new address. This option contains a HMAC to authenticate the address as having been sent from the originator of the connection. Further details are in Section 3.4.1.

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

There is a corresponding signal for address removal, making use of the Address ID that is signaled 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-bit sequence number space, utilising the regular TCP sequence number header, 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 acknowledgment (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 acknowledgment 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 only being 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 are 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 behavior 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 are 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:

- o 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.

- o 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.

- o To meet the threats identified in [RFC6181], the following steps are taken: keys are sent in the clear in the MP_CAPABLE messages; MP_JOIN messages are secured with HMAC-SHA256 ([RFC2104], [SHS]) using those keys; and standard TCP validity checks are made on the other messages (ensuring sequence numbers are in-window [RFC5961]).

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 signaled using optional TCP header fields. A single TCP option number ("Kind") has been assigned by IANA for MPTCP (see Section 8), and then individual messages will be determined by a "subtype", the values of which are also 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 subtype value of the symbolic name as defined in Section 8. This subtype is a 4-bit field -- the first 4 bits of the option payload, as shown in Figure 3. The MPTCP messages are defined in the following sections.

```
                     1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---------------+---------------+-------+-----------------------+
|     Kind      |    Length     |Subtype|                       |
+---------------+---------------+-------+                       |
|                     Subtype-specific data                     |
|                       (variable length)                       |
+---------------------------------------------------------------+
```

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 (selective acknowledgment) [RFC2018]) 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 [RFC5681] in regular TCP. Therefore, an MPTCP implementation receiving a duplicate ACK that 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 acknowledgment number are within window) MUST be undertaken before processing any MPTCP signals, as described in [RFC5961], and initial subfow sequence numbers SHOULD be generated according to the recommendations in [RFC6528].

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

The MP_CAPABLE exchange in this specification (v1) is different to that specified in v0 [RFC6824]. If a host supports multiple versions of MPTCP, the sender of the MP_CAPABLE option SHOULD signal the highest version number it supports. The passive opener, on receipt of this, will signal the version number it wishes to use, which MUST be equal to or lower than the version number indicated in the initial MP_CAPABLE. Given the SYN exchange is different between v1 and v0 the exchange cannot be immediately downgraded, and therefore if the far end has requested a lower version then the initiator SHOULD respond with an ACK without any MP_CAPABLE option, to fall back to regular TCP. If the initiator supports the requsted version, on future connections to the target host, the initiator MAY cache the version preference. Alternatively, the initiator MAY close the connection with a TCP RST and immediately re-establish with the requested version of MPTCP.

The MP_CAPABLE option is variable-length, with different fields included depending on which packet the option is used on. The full MP_CAPABLE option is shown in Figure 4.

```
                     1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---------------+---------------+-------+-------+---------------+
|     Kind      |    Length     |Subtype|Version|A|B|C|D|E|F|G|H|
+---------------+---------------+-------+-------+---------------+
|                   Option Sender's Key (64 bits)               |
|                      (if option Length > 4)                   |
|                                                               |
+---------------------------------------------------------------+
|                  Option Receiver's Key (64 bits)              |
|                      (if option Length > 12)                  |
|                                                               |
+-------------------------------+-------------------------------+
|  Data-Level Length (16 bits)  |  Checksum (16 bits, optional) |
+-------------------------------+-------------------------------+
```

Figure 4: Multipath Capable (MP_CAPABLE) Option

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

- o SYN (A->B): only the first four octets (Length = 4).
- o SYN/ACK (B->A): B's Key for this connection (Length = 12).
- o ACK (no data) (A->B): A's Key followed by B's Key (Length = 20).
- o ACK (with first data) (A->B): A's Key followed by B's Key followed by Data-Level Length, and optional Checksum (Length = 22 or 24).

The contents of the option is determined by the SYN and ACK flags of the packet, along with 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).

The initial SYN, containing just the MP_CAPABLE header, is used to define the version of MPTCP being requested, as well as exchanging flags to negotiate connection features, described later.

This option is used to declare the 64-bit keys that the end hosts have 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.

Upon reception of the initial SYN-segment, a stateful server generates a random key and replies with a SYN/ACK. The key's 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 [RFC4086]. 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, and if there is, then it should generate a new key. 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.

Since key generation is implementation-specific, there is no requirement that they be simply random numbers. An implemention is free to exchange cryptographic material out-of-band and generate these keys from this, in order to provide additional mechanisms by which to verify the identity of the communicating entities. For example, an implementation could choose to link its MPTCP keys to those used in higher-layer TLS or SSH connections.

If the server behaves in a stateless manner, it has to generate its own key in a verifiable fashion. This verifiable way of generating the key can be done by using a hash of the 4-tuple, sequence number and a local secret (similar to what is done for the TCP-sequence number [RFC4987]). It will thus be able to verify whether it is indeed the originator of the key echoed back in the later MP_CAPABLE option. As for a stateful server, the tokens SHOULD be checked for uniqueness, however if uniqueness is not met, and there is no way to generate an alternative verifiable key, then the connection MUST fall back to using regular TCP by not sending a MP_CAPABLE in the SYN/ACK.

The ACK carries both A's key and B's key. This is the first time that A's key is seen on the wire, although it is expected that A will have generated a key locally before the initial SYN. The echoing of B's key allows B to operate statelessly, as described above. Therefore, A's key must be delivered reliably to B, and in order to do this, the transmission of this packet must be made reliable.

If B has data to send first, then the reliable delivery of the ACK can be inferred by the receipt of this data with a MPTCP Data Sequence Signal (DSS) option (Section 3.3). If, however, A wishes to send data first, it would not know whether the ACK has successfully been received, and thus whether the MPTCP is successfully established. Therefore, on the first data A has to send (if it has not received any data from B), it MUST also include a MP_CAPABLE option, with additional data parameters (the Data-Level Length and optional Checksum as shown in Figure 4). This packet may be the third ACK if data is ready to be sent by the application, or may be a later packet if the application only later has data to send. This MP_CAPABLE option is in place of the DSS, and simply specifies the data-level length of the payload, and the checksum (if the use of checksums is negotiated). This is the minimal data required to establish a MPTCP connection - it allows validation of the payload, and given it is the first data, the Initial Data Sequence Number (IDSN) is also known (as it is generated from the key, as described below). Conveying the keys on the first data packet allows the TCP reliability mechanisms to ensure the packet is successfully delivered. The receiver will acknowledge this data a the connection level with a Data ACK, as if a DSS option has been received.

There could be situations where both A and B attempt to transmit initial data at the same time. For example, if A did not initially have data to send, but then needed to transmit data before it had received anything from B, it would use a MP_CAPABLE option with data parameters (since it would not know if the MP_CAPABLE on the ACK was received). In such a situation, B may also have transmitted data with a DSS option, but it had not yet been received at A. Therefore, B has received data with a MP_CAPABLE mapping after it has sent data with a DSS option. To ensure these situations can be handled, it follows that the data parameters in a MP_CAPABLE are semantically equivalent to those in a DSS option and can be used interchangeably. Similar situations could occur when the MP_CAPABLE with data is lost and retransmitted. Furthermore, in the case of TCP Segmentation Offloading, the MP_CAPABLE with data parameters may be duplicated across multiple packets, and implementations must also be able to cope with duplicate MP_CAPABLE mappings as well as duplicate DSS mappings.

Additionally, the MP_CAPABLE 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 3.8. Note that new subflows MUST NOT be established (using the process documented in Section 3.2) until a Data Sequence Signal (DSS) option has been successfully received across the path (as documented in Section 3.3).

The first 4 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 4 bits of this octet specify the MPTCP version in use (for this specification, this is 1).

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

A: The leftmost bit, labeled "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, labeled "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: The third bit, labeled "C", is set to "1" to indicate that the sender of this option will not accept additional MPTCP subflows to the source address and port, and therefore the receiver MUST NOT try to open any additional subflows towards this address and port. This is an efficiency improvement for situations where the sender knows a restriction is in place, for example if the sender is behind a strict NAT, or operating behind a legacy Layer 4 load balancer.

D through H: The remaining bits, labeled "D" through "H", are used for crypto algorithm negotiation. Currently only the rightmost bit, labeled "H", is assigned. Bit "H" indicates the use of HMAC-SHA256 (as defined in Section 3.2). An implementation that only supports this method MUST set bit "H" to 1, and bits "D" through "G" to 0.

A crypto algorithm MUST be specified. If flag bits D 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 (IDSN). In this specification, with only the SHA-256 algorithm (bit "H") specified and selected, the token MUST be a truncated (most significant 32 bits) SHA-256 hash ([SHS], [RFC6234]) of the key. A different, 64-bit truncation (the least significant 64 bits) of the SHA-256 hash of the key MUST be used as the IDSN. 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-256 digest, as per [SHS]. 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 (labeled "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. If A=1 is received by a host that does not want to use checksums, it MUST fall back to regular TCP by ignoring the MP_CAPABLE option as if it was invalid.

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 1 bit set -- this is the chosen algorithm. The rationale for this behavior 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 fallback 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; 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 3.10).

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 reordered 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 3.8.

The initial data sequence number on an 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-256 algorithm specified and selected, the IDSN of a host MUST be the least significant 64 bits of the SHA-256 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 3.10 for heuristics).

A new subflow is started as a normal TCP SYN/ACK exchange. The Join Connection (MP_JOIN) MPTCP 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 behavior of MP_JOIN using the HMAC-SHA256 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-256 ([SHS], [RFC6234]) 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 sends not only 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 [RFC4086].

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 an 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, labeled "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.

```
                         1                   2                   3
     0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
    +---------------+---------------+-------+-----+-+---------------+
    |     Kind      |  Length = 12  |Subtype|     |B|   Address ID  |
    +---------------+---------------+-------+-----+-+---------------+
    |                   Receiver's Token (32 bits)                  |
    +---------------------------------------------------------------+
    |                Sender's Random Number (32 bits)               |
    +---------------------------------------------------------------+
```

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 a reset (RST) signal, analogous to an unknown port in TCP, containing a MP_TCPRST option (Section 3.6) with an appropriate reason code. 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; 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 [RFC2104] is used, along with the SHA-256 hash algorithm [SHS] (potentially implemented as in [RFC6234]), 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; 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; 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.

```
                     1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---------------+---------------+-------+-----+-+---------------+
|     Kind      |  Length = 16  |Subtype|     |B|   Address ID  |
+---------------+---------------+-------+-----+-+---------------+
|                                                               |
|                Sender's Truncated HMAC (64 bits)              |
|                                                               |
+---------------------------------------------------------------+
|                Sender's Random Number (32 bits)               |
+---------------------------------------------------------------+
```

Figure 6: Join Connection (MP_JOIN) Option (for Responding SYN/ACK)

```
                     1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---------------+---------------+-------+-----------------------+
|     Kind      |  Length = 24  |Subtype|      (reserved)       |
+---------------+---------------+-------+-----------------------+
|                                                               |
|                                                               |
|                   Sender's HMAC (160 bits)                    |
|                                                               |
|                                                               |
+---------------------------------------------------------------+
```

Figure 7: Join Connection (MP_JOIN) Option (for Third ACK)

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

```
             Host A                                  Host B
   ------------------------                       ----------
   Address A1    Address A2                       Address B1
   ----------    ----------                       ----------
        |             |                               |
        |           SYN + MP_CAPABLE(Key-A)           |
        |-------------------------------------------->|
        |<--------------------------------------------|
        |         SYN/ACK + MP_CAPABLE(Key-B)         |
        |             |                               |
        |       ACK + MP_CAPABLE(Key-A, Key-B)        |
        |-------------------------------------------->|
        |             |                               |
        |             |   SYN + MP_JOIN(Token-B, R-A) |
        |             |------------------------------>|
        |             |<------------------------------|
        |             | SYN/ACK + MP_JOIN(HMAC-B, R-B) |
        |             |                               |
        |             |     ACK + MP_JOIN(HMAC-A)     |
        |             |------------------------------>|
        |             |<------------------------------|
        |             |             ACK               |

   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, with a MP_TCPRST option (Section 3.6) with an appropriate reason code.

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. In this, and all following cases of sending a RST in this section, the sender SHOULD send a MP_TCPRST option (Section 3.6) on this RST packet with the reason code for a "MPTCP specific error".

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 with a MP_TCPRST (Section 3.6) option with the reason code for "MPTCP specific error".

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 TCP RST with a MP_TCPRST (Section 3.6) option with the reason code for "MPTCP specific error".

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 a 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 3.10 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 set up, demultiplexing packets is done using the 5-tuple, as in traditional TCP. The 5-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 behavior in detail.

The data sequence mapping and the Data ACK are signaled in the Data Sequence Signal (DSS) option (Figure 9). Either or both can be signaled in one DSS, depending 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.

```
                     1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---------------+---------------+-------+----------------------+
|     Kind      |    Length     |Subtype| (reserved) |F|m|M|a|A|
+---------------+---------------+-------+----------------------+
|           Data ACK (4 or 8 octets, depending on flags)       |
+--------------------------------------------------------------+
|   Data sequence number (4 or 8 octets, depending on flags)   |
+--------------------------------------------------------------+
|              Subflow Sequence Number (4 octets)              |
+-------------------------------+------------------------------+
|  Data-Level Length (2 octets) |      Checksum (2 octets)     |
+-------------------------------+------------------------------+
```

Figure 9: Data Sequence Signal (DSS) Option

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

o A = Data ACK present

o a = Data ACK is 8 octets (if not set, Data ACK is 4 octets)

o M = Data Sequence Number (DSN), Subflow Sequence Number (SSN), Data-Level Length, and Checksum present

o 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 and 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. This RST SHOULD be accompanied with a MP_TCPRST option (Section 3.6) with the reason code for a "MPTCP specific error".

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 [RFC2018] 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 3.8, 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 [RFC0793], operating over the data covered by this mapping, along with a pseudo-header as shown in Figure 10.

```
                     1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+--------------------------------------------------------------+
|                                                              |
|                Data Sequence Number (8 octets)               |
|                                                              |
+--------------------------------------------------------------+
|              Subflow Sequence Number (4 octets)              |
+-------------------------------+------------------------------+
|  Data-Level Length (2 octets) |        Zeros (2 octets)      |
+-------------------------------+------------------------------+
```

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; 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 a 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 an implementation MAY 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 implementaton MUST be able to receive and process both 64-bit or 32-bit sequence number values, but it is not required that an implementation is able to send both.

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) [RFC1323]. 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-256 hash of the host’s key, as described in Section 3.1. This is required also in order for the receiver to know what the expected IDSN is, and thus determine if any initial connection-level packets are missing; this is particularly relevant if two subflows start transmitting simultaneously.

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 fall back to regular TCP by mapping the subflow-level data to the connection-level data for the remainder of the connection (see Section 3.8). 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 acknowledgment, 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 behavior 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 the 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 acknowledgment 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 on which the data was sent. 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, using an appropriate MP_TCPRST (Section 3.6) error code.

The Data ACK MAY be included in all segments; however, optimizations SHOULD be considered in more advanced implementations, where the Data ACK is present in segments only when the Data ACK value advances, and this behavior MUST be treated as valid. This behavior 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; 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 signaled by setting the ’F’ flag in the Data Sequence Signal option (Figure 9) to 1. A DATA_FIN occupies 1 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 sequence space 80-89, the DATA_FIN is DSN 90; 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 a 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 behavior 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 until 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 a RST with MP_TCPRST (Section 3.6) error code for "too much outstanding data".

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 the 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 put either in the in-order receive queue or in the out-of-order queue. In Multipath TCP, the same happens but at the 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 levels. 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 round-trip time (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 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 resend 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 behavior. Optimizations could be negotiated in future versions of this protocol. Note also that this property would also permit a sender to always send the same data, with the same data sequence number, on multiple subflows, if it so desired for reliability reasons.

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 retransmission by middleboxes that proactively 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 on which it has been sent. 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 a 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 misdiagnosed as underperforming. A RST for this purpose SHOULD be accompanied with an appropriate MP_TCPRST option (Section 3.6).

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 [RFC6356]; 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 maximize throughput, all available paths will be used simultaneously for data transfer, using coupled congestion control as described in [RFC6356]. 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 [RFC6897].

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 Explicit Congestion Notification (ECN) signal [RFC3168], 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.

```
                     1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---------------+---------------+-------+-----+-+
|     Kind      |     Length    |Subtype|     |B|
+---------------+---------------+-------+-----+-+
```

Figure 11: Change Subflow Priority (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 signaled B=1 to the other host.

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 MPTCP Architecture document [RFC6182].

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:

- o 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 multiaddressed, 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.

- o 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.

- o If subflow A2-B1 is successfully set up, 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) MPTCP option announces additional addresses (and optionally, ports) on which a host can be reached (Figure 12). 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, e.g. [RFC5961], before acting upon it.

Every address has an Address ID that 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 learned 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 it MAY choose to refresh advertisements periodically.

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

The 2 octets that specify the TCP port number to use are optional and their presence 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 3.10.

The Truncated HMAC present in this Option is the rightmost 64 bits of an HMAC, negotiated and calculated in the same way as for MP_JOIN as described in Section 3.2. For this specification of MPTCP, as there is only one hash algorithm option specified, this will be HMAC as defined in [RFC2104], using the SHA-256 hash algorithm [SHS], implemented as in [RFC6234]. In the same way as for MP_JOIN, 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 is the Address ID, IP Address, and Port which precede the HMAC in the ADD_ADDR option. If the port is not present in the ADD_ADDR option, the HMAC message will nevertheless include two octets of value zero. The rationale for the HMAC is to prevent unauthorized entities from injecting ADD_ADDR signals in an attempt to hijack a connection.

Note that additionally the presence of this HMAC prevents the address being changed in flight unless the key is known by an intermediary. If a host receives an ADD_ADDR option for which it cannot validate the HMAC, it SHOULD silently ignore the option.

A set of four flags are present after the subtype and before the Address ID. Only the rightmost bit - labelled 'E' - is assinged today. The other bits are currently unassigned and MUST be set to zero by a sender and MUST be ignored by the receiver.

The 'E' bit exists to provide reliability for this option. Because this option will often be sent on pure ACKs, there is no guarantee of reliability. Therefore, a receiver receiving a fresh ADD_ADDR option (where E=0), will send the same option back to the sender, but not including the HMAC, and with E=1. The lack of this echo can be used by the initial ADD_ADDR sender to retransmit the ADD_ADDR according to local policy.

```
                     1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---------------+---------------+-------+-------+---------------+
|     Kind      |     Length    |Subtype|(rsv)|E|  Address ID   |
+---------------+---------------+-------+-------+---------------+
|          Address (IPv4 - 4 octets / IPv6 - 16 octets)         |
+-------------------------------+-------------------------------+
|   Port (2 octets, optional)   |                               |
+-------------------------------+                               |
|        Truncated HMAC (8 octets, if length > 10 octets)       |
|                               +-------------------------------+
|                               |
+-------------------------------+
```

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 [RFC1918]. 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 minimize 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 misdirect, 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 MPTCP options is not necessary. In order to minimize 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 minimize 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 from 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 set up 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 [RFC5681], 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 [RFC1122] 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.

```
                     1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---------------+---------------+-------+-------+---------------+
|     Kind      |  Length = 3+n |Subtype|(resvd)|   Address ID  | ...
+---------------+---------------+-------+-------+---------------+
                           (followed by n-1 Address IDs, if required)
```

Figure 13: Remove Address (REMOVE_ADDR) Option

3.5. Fast Close

Regular TCP has the means of sending a reset (RST) signal to abruptly close a connection. With MPTCP, a regular 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 anymore. 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 [RFC0793]. 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.

```
                     1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---------------+---------------+-------+-----------------------+
|     Kind      |    Length     |Subtype|      (reserved)       |
+---------------+---------------+-------+-----------------------+
|                      Option Receiver's Key                    |
|                            (64 bits)                          |
|                                                               |
+---------------------------------------------------------------+
```

Figure 14: Fast Close (MP_FASTCLOSE) Option

If Host A wants to force the closure of an MPTCP connection, it has two different options:

- Option A (ACK) : 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.

- Option R (RST) : Host A sends a RST containing the MP_FASTCLOSE option on all subflows, containing the key of Host B as declared in the initial connection handshake. Host A can tear the subflows and the connection down immediately.

If a host receives a packet with a valid MP_FASTCLOSE option, it shall process it as follows :

- o Upon receipt of an ACK with 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).

- o 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. Host A should reply with a TCP RST and tear down the connection.

- o If Host A does not receive a TCP RST in reply to its MP_FASTCLOSE after one retransmission timeout (RTO) (the RTO of the subflow where the MP_FASTCLOSE 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. If no TCP RST is received in response, Host A SHOULD send a TCP RST with the MP_FASTCLOSE option itself when it releases state in order to clear any remaining state at middleboxes.

- o Upon receipt of a RST with MP_FASTCLOSE, containing the valid key, Host B tears down all subflows. Host B can now close the whole MPTCP connection (it transitions directly to CLOSED state).

3.6. Subflow Reset

As discussed in Section 3.5 above, the MP_FASTCLOSE option provides a connection-level reset roughly analagous to a TCP RST. Regular TCP RST options remain used to at the subflow-level to indicate the receiving host has no knowledge of the MPTCP subflow or TCP connection to which the packet belongs.

However, in MPTCP, there may be many reasons for rejecting the opening of a subflow, but these semantics cannot be carried in a standard TCP RST. It would be beneficial for a host to the reasons why its subflow has been closed with a RST, and thus whether it should try to re-establish the subflow immediately, later, or never again. These semantics are carried in the MP_TCPRST option that can be included on a TCP RST packet.

```
                     1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---------------+---------------+-------+-----------------------+
|     Kind      |    Length     |Subtype|U|V|W|T|    Reason     |
+---------------+---------------+-------+-----------------------+
```

Figure 15: TCP RST Reason (MP_TCPRST) Option

The MP_TCPRST option contains a reason code that allows the sender of the option to provide more information about the reason for the termination of the subflow. Using 12 bits of option space, the first four bits are reserved for flags (only one of which is currently defined), and the remaining octet is used to express a reason code for this subflow termination, from which a receiver MAY infer information about the usability of this path.

The "T" flag is used by the sender to indicate whether the error condition that is reported is Transient (T bit set to 1) or Permanent (T bit set to 0). If the error condition is considered to be Transient by the sender of the RST segment, the recipient of this segment MAY try to reestablish a subflow for this connection over the failed path. The time at which a receiver may try to re-establish this is implementation-specific, but SHOULD take into account the properties of the failure defined by the following reason code. If the error condition is considered to be permanent, the receiver of the RST segment SHOULD NOT try to reestablish a subflow for this connection over this path. The "U", "V" and "W" flags are not defined by this specification and are reserved for future use.

The "Reason" code is an 8-bit field that indicates the reason for the termination of the subflow. The following codes are defined in this document:

- o Unspecified error (code 0x0). This is the default error implying the subflow is not longer available. The receiving host SHOULD take account of the 'T' bit in deciding whether to re-estbalish this subflow. The presence of this option shows that the RST was generated by a MPTCP-aware device.

- o MPTCP specific error (code 0x01). An error has been detected in the processing of MPTCP options. This is the usual reason code to return in the cases where a RST is being sent to close a subflow for reasons of an invalid response.

- o Lack of resources (code 0x02). This code indicates that the sending host does not have enough ressources to support the terminated subflow.

o Administratively prohibited (code 0x03). This code indicates that the requested subflow is prohibited by the policies of the sending host.

o Too much outstanding data (code 0x04). This code indicates that there is an excessive amount of data that need to be transmitted over the terminated subflow while having already been acknowledged over one or more other subflows. This may occur if a path has been unavailable for a short period and it is more efficient to reset and start again than it is to retransmit the queued data.

o Unacceptable performance (code 0x05). This code indicates that the performance of this subflow was too low compared to the other subflows of this Multipath TCP connection.

o Middlebox interference (code 0x06). Middlebox interference has been detected over this subflow making MPTCP signaling invalid. For example, this may be sent if the checksum does not validate.

3.7. MPTCP Experimental Option

In order to provide a structured identity and negotiation mechanism for private experimental MPTCP extensions, the MP_EXPERIMENTAL option has been reserved.

```
                     1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---------------+---------------+-------+-------+---------------+
|     Kind      |    Length     |Subtype|S|U|rsv|  Experiment   |
+---------------+---------------+-------+-------+---------------+
| Id. (16 bits) | Subtype-specific data  (variable length)  ...
+--------------------------------------------------------- ...
```

Figure 16: MPTCP Experimental (MP_EXPERIMENTAL) Option

Figure 16 shows the format of the experimental option. The Experiment identifier is a 16 bits integer that shall be assigned by using the same procedure as defined in [RFC6994]; a request to IANA is made in Section 8.4.

The two high order flags that are included in the MPTCP Experimental option have the following semantics:

o "S" flag (highest order bit) : This is the synchronising bit. When set to 1, it indicates that the host sending this option expects a reply from the remote host with an option having the same experiment identifier, but possibly containing other data.

- "U" flag (second highest order bit) : When set to 1, this flag indicates that the experimental option was received by the sending host but it was unable to parse it.

The two low order flags are currently reserved for further use. They MUST be set to zero when sending and ignored upon reception.

To use the Experimental MPTCP option with a given experiment identifier over a MPTCP connection, the sending host must first verify the ability of the remote host to support this particular Experimental option. For this, it first sends in any valid TCP segment, including a duplicate acknowledgement, an Experimental MPTCP option with the "S" flag set. Upon reception of this option, the receiving host will verify whether it supports it. If yes, it shall return a TCP segment that contains the experimental option with the same identifier and the "S" and the "U" flags both set to 1. This option may contain additional data depending on the semantics of the extension. If the receiving host does not recognise the experimental option that it has received, it shall return a TCP segment that contains the received experimental option with the "S" flag set to 0 and the "U" flag set to 1.

If a host receives an Experimental MPTCP option with the "U" flag set to 0 which it does not support, or which contains information that the host cannot parse, it shall return the exact option that it received with the "U" flag set to 1 to indicate the error to the remote host. If an invalid option is received with the "U" flag set to 0, it must be silently discarded.

Future documents specifying new experimental MPTCP options should specify the extract semantic of the Subtype-specific data and whether additional validation operations are to be followed at both sides. It should be noted that data can be included in an experimental option concurrently with the capability check (S/U).

3.8. 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 following: the loss of MPTCP 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 be either 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 MPTCP 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, behavior 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 MPTCP 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 that 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 acknowledgment, 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 a 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.

If a subflow breaks during operation, e.g. if it is re-routed and MPTCP options are no longer permitted, then once this is detected (by the subflow-level receive buffer filling up), the subflow SHOULD be treated as broken and closed with a RST, since no data can be delivered to the application layer, and no fallback signal can be reliably sent. This RST SHOULD include the MP_TCPRST option (Section 3.6) with an appropriate reason code.

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 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 a RST, containing a MP_TCPRST option (Section 3.6) with an appropriate reason code.

The case described above is a specialized 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, if checksums have been negotiated. 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 it 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.

Note that if checksum usage has not been negotiated, this fallback mechanism cannot be used unless there is some higher or lower layer signal to inform the MPTCP implementation that the payload has been tampered with.

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 a RST, featuring an MP_FAIL option

(Figure 17), which defines the data sequence number at the start of the segment (defined by the data sequence mapping) that 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.

```
                     1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---------------+---------------+-------+----------------------+
|     Kind      |   Length=12   |Subtype|      (reserved)      |
+---------------+---------------+-------+----------------------+
|                                                              |
|                 Data Sequence Number (8 octets)              |
|                                                              |
+--------------------------------------------------------------+
```

Figure 17: 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 retransmitted 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, referring 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 most recent segment that was known to be delivered intact (i.e. was successfully DATA_ACKed). 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. The MP_FAIL signal affects only one direction of traffic. It is not mandatory for the reciever of an MP_FAIL to also respond with an MP_FAIL, since the paths may only be damaged in one direction. However, implementations MAY choose to send a MP_FAIL in the reverse direction and entirely revert to a regular 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 a 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 emphasized 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.

3.9. 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 a RST -- are covered in Section 4. Where possible, we do not want to deviate from regular TCP behavior.

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

- o 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 behavior on an unknown port)

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

- o Remove request for unknown address ID: silently ignore

3.10. Heuristics

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

3.10.1. 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 [RFC2992]).

3.10.2. Delayed Subflow Start and Subflow Symmetry

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 that 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.

If the two communicating hosts immediately try to set up subflows from all available addresses to all available addresses on the other host, this could end up creating two subflows per path. This is an inefficient use of resources.

If the the same ports are used on all subflows, as recommended above, then standard TCP simultaneous open logic should take care of this situation and only one subflow will be established between the address pairs. However, this relies on the same ports being used at both end hosts. If a host does not support TCP simultaneous open, it is RECOMMENDED that some element of randomization is applied to the time waited before opening new subflows, so that only one subflow exists between a given address pair. If, however, hosts signal additional ports to use (for example, for leveraging ECMP on-path), this heuristic need not apply.

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

3.10.3. Failure Handling

Requirements for MPTCP’s handling of unexpected signals have been given in Section 3.9. There are other failure cases, however, where a hosts can choose appropriate behavior.

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 3.9 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. Behavior could also be learned for particularly badly performing subflows or subflows that regularly fail during use, in order to temporarily choose not to use these paths.

3.11. TCP Fast Open

TCP Fast Open, described in [RFC7413], has been introduced with the objective of gaining one RTT before transmitting data. This is considered a valuable gain as very short connections are very common, especially for HTTP request/response schemes. It achieves this by sending the SYN-segment together with data and allowing the server to reply immediately with data after the SYN/ACK. [RFC7413] secures this mechanism, by using a new TCP option that includes a cookie which is negotiated in a preceding connection.

When using TCP Fast Open in conjunction with MPTCP, there are two key points to take into account, detailed hereafter.

3.11.1. TFO cookie request with MPTCP

When a TFO client first connects to a server, it cannot immediately include data in the SYN for security reasons [RFC7413]. Instead, it requests a cookie that will be used in subsequent connections. This is done with the TCP cookie request/response options, of resp. 2 bytes and 6-18 bytes (depending on the chosen cookie length).

TFO and MPTCP can be combined provided that the total length of their options does not exceed the maximum 40 bytes possible in TCP:

- o In the SYN: MPTCP uses a 4-bytes long MP_CAPABLE option. The MPTCP and TFO options sum up to 6 bytes. With typical TCP-options using up to 19 bytes in the SYN (24 bytes if options are padded at a word boundary), there is enough space to combine the MP_CAPABLE with the TFO Cookie Request.

- o In the SYN+ACK: MPTCP uses a 12-bytes long MP_CAPABLE option, but now TFO can be as long as 18 bytes. Since the maximum option length may be exceeded, it is up to the server to solve this by using a shorter cookie. As an example, if we consider that 19 bytes are used for classical TCP options, the maximum possible cookie length would be of 7 bytes. Note that the same limitation applies to subsequent connections, for the SYN packet (because the client then echoes back the cookie to the server). Finally, if the security impact of reducing the cookie size is not deemed acceptable, the server can reduce the amount of other TCP-options by omitting the TCP timestamps (as outlined in Appendix A).

3.11.2. Data sequence mapping under TFO

MPTCP uses, in the TCP establishment phase, a key exchange that is used to generate the Initial Data Sequence Numbers (IDSNs). In particular, the SYN with MP_CAPABLE occupies the first octet of the data sequence space. With TFO, one way to handle the data sent together with the SYN would be to consider an implicit DSS mapping that covers that SYN segment (since there is not enough space in the SYN to include a DSS option). The problem with that approach is that if a middlebox modifies the TFO data, this will not be noticed by MPTCP because of the absence of a DSS-checksum. For example, a TCP (but not MPTCP)-aware middlebox could insert bytes at the beginning of the stream and adapt the TCP checksum and sequence numbers accordingly. With an implicit mapping, this would give to client and server a different view on the DSS-mapping, with no way to detect this inconsistency as the DSS checksum is not present.

To solve this, the TFO data should not be considered part of the Data Sequence Number space: the SYN with MP_CAPABLE still occupies the first octet of data sequence space, but then the first non-TFO data byte occupies the second octet. This guarantees that, if the use of DSS-checksum is negotiated, all data in the data sequence number space is checksummed. We also note that this does not entail a loss of functionality, because TFO-data is always sent when only one path is active.

3.11.3. Connection establishment examples

The following shows a few examples of possible TFO+MPTCP establishment scenarios.

Before a client can send data together with the SYN, it must request a cookie to the server, as shown in Figure Figure 18. This is done by simply combining the TFO and MPTCP options.

```
client                                                         server
   |                                                              |
   |    S 0(0) <MP_CAPABLE>, <TFO cookie request>                 |
   | -----------------------------------------------------------> |
   |                                                              |
   |    S. 0(0) ack 1 <MP_CAPABLE>, <TFO cookie>                  |
   | <----------------------------------------------------------- |
   |                                                              |
   |    .  0(0) ack 1 <MP_CAPABLE>                                |
   | -----------------------------------------------------------> |
   |                                                              |
```

Figure 18: Cookie request

Once this is done, the received cookie can be used for TFO, as shown in Figure Figure 19. In this example, the client first sends 20 bytes in the SYN. The server immediately replies with 100 bytes following the SYN-ACK upon which the client replies with 20 more bytes. Note that the last segment in the figure has a TCP sequence number of 21, while the DSS subflow sequence number is 1 (because the TFO data is not part of the data sequence number space, as explained in Section Section 3.11.2.

```
client                                                             server
  |                                                                 |
  |    S  0(20) <MP_CAPABLE>, <TFO cookie>                          |
  | --------------------------------------------------------------> |
  |                                                                 |
  |    S. 0(0) ack 21 <MP_CAPABLE>                                  |
  | <-------------------------------------------------------------- |
  |                                                                 |
  |    .  1(100) ack 21 <DSS ack=1 seq=1 ssn=1 dlen=100>            |
  | <-------------------------------------------------------------- |
  |                                                                 |
  |    .  21(0) ack 1 <MP_CAPABLE>                                  |
  | --------------------------------------------------------------> |
  |                                                                 |
  |    .  21(20) ack 101 <DSS ack=101 seq=1 ssn=1 dlen=20>          |
  | --------------------------------------------------------------> |
  |                                                                 |
```

Figure 19: The server supports TFO

In Figure Figure 20, the server does not support TFO. The client detects that no state is created in the server (as no data is acked), and now sends the MP_CAPABLE in the third ack, in order for the server to build its MPTCP context at then end of the establishment. Now, the tfo data, retransmitted, becomes part of the data sequence mapping because it is effectively sent (in fact re-sent) after the establishment.

```
client                                                             server
  |                                                                 |
  |    S  0(20) <MP_CAPABLE>, <TFO cookie>                          |
  | --------------------------------------------------------------> |
  |                                                                 |
  |    S. 0(0) ack 1 <MP_CAPABLE>                                   |
  | <-------------------------------------------------------------- |
  |                                                                 |
  |    .  1(0) ack 1 <MP_CAPABLE>                                   |
  | --------------------------------------------------------------> |
  |                                                                 |
  |    .  1(20) ack 1 <DSS ack=1 seq=1 ssn=1 dlen=20>               |
  | --------------------------------------------------------------> |
  |                                                                 |
  |    .  0(0) ack 21 <DSS ack=21 seq=1 ssn=1 dlen=0>               |
  | <-------------------------------------------------------------- |
  |                                                                 |
```

Figure 20: The server does not support TFO

It is also possible that the server acknowledges only part of the TFO data, as illustrated in Figure Figure 21. The client will simply retransmit the missing data together with a DSS-mapping.

```
client                                                           server
  |                                                                 |
  |  S  0(1000) <MP_CAPABLE>, <TFO cookie>                          |
  | -------------------------------------------------------------> |
  |                                                                 |
  |  S. 0(0) ack 501 <MP_CAPABLE>                                   |
  | <------------------------------------------------------------- |
  |                                                                 |
  |    .  501(0) ack 1 <MP_CAPABLE>                                 |
  | -------------------------------------------------------------> |
  |                                                                 |
  |   .  501(500) ack 1 <DSS ack=1 seq=1 ssn=1 dlen=500>            |
  | -------------------------------------------------------------> |
  |                                                                 |
```

Figure 21: Partial data acknowledgement

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 1 octet of sequence space. There is an explicit mapping of data sequence space to subflow sequence space, which is signaled 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 acknowledgment 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 [RFC6897].

5. Security Considerations

As identified in [RFC6181], the addition of multipath capability to TCP will bring with it a number of new classes of threat. In order to prevent these, [RFC6182] 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 Sections 3.1 and 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 while 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 (HMACs) 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; 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 [RFC4086] 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 document should therefore protect against all forms of flooding and hijacking attacks discussed in [RFC6181].

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 individual 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 3.10) 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 analysis ([RFC6181]), 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:

o 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.

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

o 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 [TCPLO]), or perhaps building on the current approach with a second stage of MPTCP-option-based security.

o revisiting the working group's decision to exclusively use TCP options for MPTCP signaling, 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:

o available flags in MP_CAPABLE (Figure 4);

- available subtypes in the MPTCP option (Figure 3);
- 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 behavior. 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 22 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.

```
       Host A                              Host B
         |              Middlebox M          |
         |                   |               |
         |  SYN(MP_CAPABLE)  |        SYN    |
         |-------------------|-------------->|
         |                SYN/ACK            |
         |<----------------------------------|
    a) MP_CAPABLE option stripped on outgoing path

      Host A                                Host B
         |            SYN(MP_CAPABLE)        |
         |---------------------------------->|
         |             Middlebox M           |
         |                 |                 |
         |    SYN/ACK      |SYN/ACK(MP_CAPABLE)|
         |<----------------|-----------------|
    b) MP_CAPABLE option stripped on return path
```

Figure 22: Connection Setup with Middleboxes that Strip Options from Packets

We now examine data flow with MPTCP, assuming the flow is correctly set up, 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, behavior 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 3.8. 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 behavior, 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:

o NATs [RFC3022] (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 [RFC1918] 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) [RFC3135] might proactively 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 3.8).

- Traffic Normalizers [norm] 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 [RFC2979] 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 multiaddressed 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 analyze 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 reassemble for analysis.

- Application-level middleboxes such as content-aware firewalls may alter the payload within a subflow, such as rewriting 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 fall back 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.

7. Acknowledgments

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8. IANA Considerations

This document updates [RFC6824] and as such IANA is requested to update the TCP option space registry to point to this document for Multipath TCP, as follows:

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

Table 1: TCP Option Kind Numbers

8.1. MPTCP Option Subtypes

The 4-bit MPTCP subtype sub-registry ("MPTCP Option Subtypes" under the "Transmission Control Protocol (TCP) Parameters" registry) was defined in [RFC6824]. This document defines one additional subtype (ADD_ADDR) and updates the references to this document for all sub-types except ADD_ADDR, which is deprecated. The updates are listed in the following table.

Value	Symbol	Name	Reference
0x0	MP_CAPABLE	Multipath Capable	This document, Section 3.1
0x1	MP_JOIN	Join Connection	This document, Section 3.2
0x2	DSS	Data Sequence Signal (Data ACK and data sequence mapping)	This document, Section 3.3
0x3	ADD_ADDR	Add Address	This document, Section 3.4.1
0x4	REMOVE_ADDR	Remove Address	This document, Section 3.4.2
0x5	MP_PRIO	Change Subflow Priority	This document, Section 3.3.8
0x6	MP_FAIL	Fallback	This document, Section 3.8
0x7	MP_FASTCLOSE	Fast Close	This document, Section 3.5
0x8	MP_TCPRST	Subflow Reset	This document, Section 3.6
0xf	MP_EXPERIMENTAL	MPTCP Experimental Option	This document, Section 3.7

Table 2: MPTCP Option Subtypes

Values 0x9 through 0xe are currently unassigned.

8.2. MPTCP Handshake Algorithms

IANA has created another sub-registry, "MPTCP Handshake Algorithms" under the "Transmission Control Protocol (TCP) Parameters" registry, based on the flags in MP_CAPABLE (Section 3.1). IANA is requested to update the references of this table to this document, as follows:

Flag Bit	Meaning	Reference
A	Checksum required	This document, Section 3.1
B	Extensibility	This document, Section 3.1
C	Do not attempt to connect to source address	This document, Section 3.1
D-G	Unassigned	
H	HMAC-SHA256	This document, Section 3.2

Table 3: MPTCP Handshake Algorithms

Note that the meanings of bits D 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 [RFC5226]. Assignments consist of the value of the flags, a symbolic name for the algorithm, and a reference to its specification.

8.3. MP_TCPRST Reason Codes

IANA is requested to create a further sub-registry, "MP_TCPRST Reason Codes" under the "Transmission Control Protocol (TCP) Parameters" registry, based on the reason code in MP_TCPRST (Section 3.6). The contents of this sub-registry are to to this document, as follows:

Code	Meaning	Reference
0x00	Unspecified TCP error	This document, Section 3.6
0x01	MPTCP specific error	This document, Section 3.6
0x02	Lack of resources	This document, Section 3.6
0x03	Administratively prohibited	This document, Section 3.6
0x04	Too much outstanding data	This document, Section 3.6
0x05	Unacceptable performance	This document, Section 3.6
0x06	Middlebox interference	This document, Section 3.6

Table 4: MPTCP MP_TCPRST Reason Codes

8.4. Experimental option registry

Section 3.7 has defined the MP_EXPERIMENTAL option for private, experimental MPTCP options, and the same considerations as for [RFC6994] apply. IANA should create a "Multipath TCP Experimental Option Identifiers (MPTCP ExIDs)" sub-registry. This registry contains the 16 bits ExIDs and a reference (description, document pointer, or assignee name and e-mail contact) for each entry. MPTCP ExIDs are assigned on a First Come, First Served (FCFS) basis [RFC5226].

IANA will advise applicants of duplicate entries to select an alternate value, as per typical FCFS processing.

IANA will record known duplicate uses to assist the community in both debugging assigned uses as well as correcting unauthorized duplicate uses.

IANA should impose no requirement on making a registration other than indicating the desired codepoint and providing a point of contact. A short description or acronym for the use is desired but should not be required.

9. References

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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 document.

SYN packets typically include Maximum Segment Size (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.

Note that due to the use of a 64-bit data-level sequence space, it is feasible that MPTCP will not require the timestamp option for protection against wrapped sequence numbers (PAWS [RFC1323]), since the data-level sequence space has far less chance of wrapping. Confirmation of the validity of this optimisation is for further study.

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 bidirectional 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 16 and 30 bytes, depending on whether IPv4 or IPv6 is used, and whether or not the port number is present. 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.

RFC 793 [RFC0793] specifies several state variables. Whenever possible, we reuse the same terminology as RFC 793 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 A 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 RFC 1323, 16 bits otherwise): 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 that 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 that 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 that is sent in the DSS option on all subflows.

RCV.WND (32 bits with RFC 1323, 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 acknowledgment 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 that 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 RFC 1323, 16 bits otherwise): This is the subflow-level receive window that is updated with the window field from the segments received on this subflow.

Appendix C. Finite State Machine

The diagram in Figure 23 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.

```
                              +---------+
                              | M_ESTAB |
                              +---------+
                     M_CLOSE    |     |    rcv DATA_FIN
                      -------   |     |    -------
 +---------+       snd DATA_FIN /       \ snd DATA_ACK[DFIN] +---------+
 |  M_FIN  |<-----------------           ------------------->| M_CLOSE |
 | WAIT-1  |---------------------------                      |   WAIT  |
 +---------+               rcv DATA_FIN \                    +---------+
   | rcv DATA_ACK[DFIN]         ------- |                   M_CLOSE |
   | --------------        snd DATA_ACK |                   ------- |
   | CLOSE all subflows                 |              snd DATA_FIN |
   V                                    V                           V
 +-----------+              +-----------+                +-----------+
 |M_FINWAIT-2|              | M_CLOSING |                | M_LAST-ACK|
 +-----------+              +-----------+                +-----------+
   |              rcv DATA_ACK[DFIN] |          rcv DATA_ACK[DFIN] |
   | rcv DATA_FIN     -------------- |              -------------- |
   |  -------     CLOSE all subflows |          CLOSE all subflows |
   | snd DATA_ACK[DFIN]             V            delete MPTCP PCB  V
   \                          +-----------+                  +---------+
     ------------------------>|M_TIME WAIT|----------------->| M_CLOSED|
                              +-----------+                  +---------+
                                         All subflows in CLOSED
                                              ------------
                                         delete MPTCP PCB
```

Figure 23: Finite State Machine for Connection Closure

Authors' Addresses

Alan Ford
Pexip

EMail: alan.ford@gmail.com

Costin Raiciu
University Politehnica of Bucharest
Splaiul Independentei 313
Bucharest
Romania

EMail: costin.raiciu@cs.pub.ro

Mark Handley
University College London
Gower Street
London WC1E 6BT
UK

EMail: m.handley@cs.ucl.ac.uk

Olivier Bonaventure
Universite catholique de Louvain
Pl. Ste Barbe, 2
Louvain-la-Neuve 1348
Belgium

EMail: olivier.bonaventure@uclouvain.be

Christoph Paasch
Apple, Inc.
Cupertino
US

EMail: cpaasch@apple.com

Multipath TCP K. Nguyen
Internet-Draft K. Ishizu
Intended status: Standards Track M. Kibria
Expires: May 3, 2017 F. Kojima
National Institute of Information and Communications Technology
October 30, 2016

An Improvement of MPTCP Initialization
draft-kien-mptcp-init-00

Abstract

This draft describes a new method of connection initialization for Multipath TCP (MPTCP). In the current implementation, the MPTCP's first subflow needs to be successfully initialized before an additional flow takes its turn. This yields to a degradation of MPTCP benefit in many use cases (e.g., transferring short flows). To overcome the problem, we propose to duplicate the first SYN packet and send the duplicating ones via multiple interfaces.

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Table of Contents

1. Introduction

MPTCP is an evolvable and efficient tool for link aggregation, e.g., on multi-homing hosts in mobile wireless networks. The flexibility of adding and deleting subflows introduce MPTCP benefits in aggregation and soft handover in many use cases. In the current implementation, MPTCP shows several drawbacks including operations in a scenario with imbalanced paths, especially handling short flows. Since a large amount of Internet TCP traffic is short flows, MPTCP should be improved to be more suitable with the traffic pattern. In the such scenario, the problem of selection of initialization path has big impacts on the MPTCP performance. It has been proven by the theoretical analysis [analysis] and real measurements [measurement]. However, there are limited works on solving the problem.

In fact, MPTCP can not choose the initial path itself. MPTCP relies on routing information to determine the destination for the initialization. The routing is static in most cases. The host is normally configured to route all the traffic through a default gateway. As a result, the first SYN of initialization has to be sent to the gateway associated network regardless of its quality. On the other hand, the routing information is available on a host for supporting beneficial operations of MPTCP. To solve the mentioned problem, we propose to duplicate the first SYN packet. The available routing information is leveraged in sending the duplications through several networks. The first received SYN/ACK is determined the best network (i.e., the one with the smallest RTT) to initialize the MPTCP connection.

2. Problem Description

This section describes the limitation of the current implementation of MPTCP. Consider an example scenario of MPTCP communication between two host (host A and host B). MPTCP on host A with multiple addresses (i.e., two addresses A1, A2) communicates with host B via network A1, A2, respectively. In this scenario, the gateway associated with address A1 is the default. Obviously, Host A send the first SYN with MP_CAPABLE to Address B1 for MPTCP initialization. After the successful initialization, the additional subflow will be added to ongoing MPTCP transmissions following one of two methods. The later subflow is initialized a new SYNC+MP_JOIN from A2 to B1 if there is no NAT between them. In case of under NAT, the SYN+MP_JOIN will be added after sending MP_ADDADDR.

For long flows, the standard mechanism works well, even the quality of services provided by the network A1 and network A2 are different. However, if the network A1 has longer Round Trip Time (RTT) than the one of network A2. The MPTCP performance is degraded, especially in the case of short flows. Besides, the similar scenario will become popular since the different network technologies are emerging especially for the next generation of mobile networks. Therefore, it is necessary and important to solve the problem.

3. Proposal

Our proposal for solving the previous problem relies on the idea of packet duplication, specifically SYN duplication. The first SYN in initialization is duplicated. The newly created SYN packets are then sent through the multiple gateways. The proposal only requires a modifications in sending/receiving procedures of MPTCP.

We describe an beneficial use case of the proposal, which is similar to the scenario mentioned in Section 3. Note that, the network A2 has shorter RTT than the one of network A1. Initially, the key is generated on host A for the first SYN. The first SYN is constructed just like as in the standard. The SYN is also included MP_CAPABLE option. Additionally, the second SYN is newly constructed with the same content. The only difference is on the layer 3 source addresses. More specifically, the source-destination pair is (A1, B1) on the first, and (A2, B1) on the second SYN, respectively. Concurrently, the two SYNs are departed from host A to host B. This task is feasible when the routing information is available for the departures.

At the host B, the early arriving SYN (i.e., the one from A2 to B1) is received. The host B then sends an acknowledgment (SYN/ACK with MP_CAPABLE) to A2. We can observe that without modification of the default gateway information, MPTCP has a good path selection via Network A2 for initialization. Further consideration is that, the later acknowledgment (SYN/ACK to B1) is used for an additional subflow (i.e., similar to MP_JOIN). Following the further operation, the whole period of MPTCP initialization is shortened comparing to the one in current implementation. Another obvious benefit of SYN duplications is enhancing the resilience of SYN transmission.

4. Acknowledgements

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5. Security Considerations

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Authors' Addresses

Kien Nguyen
National Institute of Information and Communications Technology
YRP Center No.1 Building 7F, 3-4 Hikarinooka, Yokosuka
Kanagawa 239-0847
Japan

Email: kienng@nict.go.jp

Kentaro Ishizu
National Institute of Information and Communications Technology
YRP Center No.1 Building 7F, 3-4 Hikarinooka, Yokosuka
Kanagawa 239-0847
Japan

Email: ishidu@nict.go.jp

Mirza Golam Kibria
National Institute of Information and Communications Technology
YRP Center No.1 Building 7F, 3-4 Hikarinooka, Yokosuka
Kanagawa 239-0847
Japan

Email: mirza.kibria@nict.go.jp

Fumihide Kojima
National Institute of Information and Communications Technology
YRP Center No.1 Building 7F, 3-4 Hikarinooka, Yokosuka
Kanagawa 239-0847
Japan

Email: f-kojima@nict.go.jp