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Implementing Interfaces to Transport Services

Abstract

The Transport Services system enables applications to use transport protocols flexibly for network communication and defines a protocol-independent Transport Services Application Programming Interface (API) that is based on an asynchronous, event-driven interaction pattern. This document serves as a guide to implementation on how to build such a system.

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Table of Contents

1. Introduction

The Transport Services architecture [I-D.ietf-taps-arch] defines a
system that allows applications to flexibly use transport networking
protocols. The API that such a system exposes to applications is
defined as the Transport Services API [I-D.ietf-taps-interface].
This API is designed to be generic across multiple transport
protocols and sets of protocols features.

This document serves as a guide to implementation on how to build a
system that provides a Transport Services API. It is the job of an
implementation of a Transport Services system to turn the requests
of an application into decisions on how to establish connections,
and how to transfer data over those connections once established.
The terminology used in this document is based on the Architecture
[I-D.ietf-taps-arch].

2. Implementing Connection Objects

The connection objects that are exposed to applications for
Transport Services are:

  *the Preconnection, the bundle of Properties that describes the
   application constraints on, and preferences for, the transport;

  *the Connection, the basic object that represents a flow of data
   as Messages in either direction between the Local and Remote
   Endpoints;

*and the Listener, a passive waiting object that delivers new
    Connections.

   Preconnection objects should be implemented as bundles of properties
   that an application can both read and write. A Preconnection object
   influences a Connection only at one point in time: when the
   Connection is created. Connection objects represent the interface
   between the application and the implementation to manage transport
   state, and conduct data transfer. During the process of
   establishment ([Section 4](#)), the Connection will not be bound to a
   specific transport protocol instance, since multiple candidate
   Protocol Stacks might be raced.

   Once a Preconnection has been used to create an outbound Connection
   or a Listener, the implementation should ensure that the copy of the
   properties held by the Connection or Listener is not affected when
   the application makes changes to a Preconnection object. This may
   involve the implementation performing a deep-copy, copying the
   object with all the objects that it references.

   Once the Connection is established, Transport Services
   implementation maps actions and events to the details of the chosen
   Protocol Stack. For example, the same Connection object may
   ultimately represent a single instance of one transport protocol
   (e.g., a TCP connection, a TLS session over TCP, a UDP flow with
   fully-specified Local and Remote Endpoints, a DTLS session, a SCTP
   stream, a QUIC stream, or an HTTP/2 stream). The properties held by
   a Connection or Listener is independent of other connections that
   are not part of the same Connection Group.

   Connection establishment is only a local operation for a Datagram
   transport (e.g., UDP(-Lite)), which serves to simplify the local
   send/receive functions and to filter the traffic for the specified
   addresses and ports [[RFC8085](#)].

   Once Initiate has been called, the Selection Properties and Endpoint
   information are immutable (i.e, an application is not able to later
   modify Selection Properties on the original Preconnection object).
   Listener objects are created with a Preconnection, at which point
   their configuration should be considered immutable by the
   implementation. The process of listening is described in [Section
   4.7](#).

3. Implementing Pre-Establishment

   During pre-establishment the application specifies one or more
   Endpoints to be used for communication as well as protocol
   preferences and constraints via Selection Properties and, if
   desired, also Connection Properties. Generally, Connection

Properties should be configured as early as possible, because they can serve as input to decisions that are made by the implementation (e.g., the Capacity Profile can guide usage of a protocol offering scavenger-type congestion control).

The implementation stores these properties as a part of the Preconnection object for use during connection establishment. For Selection Properties that are not provided by the application, the implementation must use the default values specified in the Transport Services API ([I-D.ietf-taps-interface]).

3.1. Configuration-time errors

The Transport Services system should have a list of supported protocols available, which each have transport features reflecting the capabilities of the protocol. Once an application specifies its Transport Properties, the transport system matches the required and prohibited properties against the transport features of the available protocols.

In the following cases, failure should be detected during pre-establishment:

  *A request by an application for Protocol Properties that cannot be satisfied by any of the available protocols. For example, if an application requires "Configure Reliability per Message", but no such feature is available in any protocol the host running the transport system on the host running the transport system this should result in an error, e.g., when SCTP is not supported by the operating system.

  *A request by an application for Protocol Properties that are in conflict with each other, i.e., the required and prohibited properties cannot be satisfied by the same protocol. For example, if an application prohibits "Reliable Data Transfer" but then requires "Configure Reliability per Message", this mismatch should result in an error.

To avoid allocating resources that are not finally needed, it is important that configuration-time errors fail as early as possible.

3.2. Role of system policy

The properties specified during pre-establishment have a close relationship to system policy. The implementation is responsible for combining and reconciling several different sources of preferences

when establishing Connections. These include, but are not limited
to:

1. Application preferences, i.e., preferences specified during the
   pre-establishment via Selection Properties.

2. Dynamic system policy, i.e., policy compiled from internally
   and externally acquired information about available network
   interfaces, supported transport protocols, and current/previous
   Connections. Examples of ways to externally retrieve policy-
   support information are through OS-specific statistics/
   measurement tools and tools that reside on middleboxes and
   routers.

3. Default implementation policy, i.e., predefined policy by OS or
   application.

In general, any protocol or path used for a connection must conform
to all three sources of constraints. A violation that occurs at any
of the policy layers should cause a protocol or path to be
considered ineligible for use. For an example of application
preferences leading to constraints, an application may prohibit the
use of metered network interfaces for a given Connection to avoid
user cost. Similarly, the system policy at a given time may prohibit
the use of such a metered network interface from the application's
process. Lastly, the implementation itself may default to
disallowing certain network interfaces unless explicitly requested
by the application and allowed by the system.

It is expected that the database of system policies and the method
of looking up these policies will vary across various platforms. An
implementation should attempt to look up the relevant policies for
the system in a dynamic way to make sure it is reflecting an
accurate version of the system policy, since the system's policy
regarding the application's traffic may change over time due to user
or administrative changes.

4. **Implementing Connection Establishment**

The process of establishing a network connection begins when an
application expresses intent to communicate with a Remote Endpoint
by calling Initiate. (At this point, any constraints or requirements
the application may have on the connection are available from pre-
establishment.) The process can be considered complete once there is
at least one Protocol Stack that has completed any required setup to
the point that it can transmit and receive the application's data.

Connection establishment is divided into two top-level steps:
Candidate Gathering, to identify the paths, protocols, and endpoints
to use, and Candidate Racing (see Section 4.2.2 of [I-D.ietf-taps-

arch]), in which the necessary protocol handshakes are conducted so
that the transport system can select which set to use.

This document structures the candidates for racing as a tree as
terminological convention. While a a tree structure is not the only
way in which racing can be implemented, it does ease the
illustration of how racing works.

The most simple example of this process might involve identifying
the single IP address to which the implementation wishes to connect,
using the system's current default interface or path, and starting a
TCP handshake to establish a stream to the specified IP address.
However, each step may also differ depending on the requirements of
the connection: if the endpoint is defined as a hostname and port,
then there may be multiple resolved addresses that are available;
there may also be multiple interfaces or paths available, other than
the default system interface; and some protocols may not need any
transport handshake to be considered "established" (such as UDP),
while other connections may utilize layered protocol handshakes,
such as TLS over TCP.

Whenever an implementation has multiple options for connection
establishment, it can view the set of all individual connection
establishment options as a single, aggregate connection
establishment. The aggregate set conceptually includes every valid
combination of endpoints, paths, and protocols. As an example,
consider an implementation that initiates a TCP connection to a
hostname + port endpoint, and has two valid interfaces available
(Wi-Fi and LTE). The hostname resolves to a single IPv4 address on
the Wi-Fi network, and resolves to the same IPv4 address on the LTE
network, as well as a single IPv6 address. The aggregate set of
connection establishment options can be viewed as follows:

```
Aggregate [Endpoint: www.example.com:80] [Interface: Any]   [Protocol: T
|-> [Endpoint: 192.0.2.1:80]       [Interface: Wi-Fi] [Protocol: TCP]
|-> [Endpoint: 192.0.2.1:80]       [Interface: LTE]   [Protocol: TCP]
|-> [Endpoint: 2001:DB8::1.80]     [Interface: LTE]   [Protocol: TCP]
```

Any one of these sub-entries on the aggregate connection attempt
would satisfy the original application intent. The concern of this
section is the algorithm defining which of these options to try,
when, and in what order.

During Candidate Gathering, an implementation first excludes all
protocols and paths that match a Prohibit or do not match all
Require properties. Then, the implementation will sort branches
according to Preferred properties, Avoided properties, and possibly
other criteria.

4.1. Structuring Candidates as a Tree

   As noted above, the considereration of multiple candidates in a
   gathering and racing process can be conceptually structured as a
   tree; this terminological convention is used throughout this
   document.

   Each leaf node of the tree represents a single, coherent connection
   attempt, with an endpoint, a path, and a set of protocols that can
   directly negotiate and send data on the network. Each node in the
   tree that is not a leaf represents a connection attempt that is
   either underspecified, or else includes multiple distinct options.
   For example, when connecting on an IP network, a connection attempt
   to a hostname and port is underspecified, because the connection
   attempt requires a resolved IP address as its Remote Endpoint. In
   this case, the node represented by the connection attempt to the
   hostname is a parent node, with child nodes for each IP address.
   Similarly, an implementation that is allowed to connect using
   multiple interfaces will have a parent node of the tree for the
   decision between the paths, with a branch for each interface.

   The example aggregate connection attempt above can be drawn as a
   tree by grouping the addresses resolved on the same interface into
   branches:

```
                            ||
             +=========================+
             |   www.example.com:80/Any  |
             +=========================+
                  //                      \\
 +=========================+       +=========================+
 | www.example.com:80/Wi-Fi |      |   www.example.com:80/LTE   |
 +=========================+       +=========================+
           ||                          //                   \\
    +==================+  +==================+  +====================
    | 192.0.2.1:80/Wi-Fi |  | 192.0.2.1:80/LTE  |  |   2001:DB8::1.80/LTE
    +==================+  +==================+  +====================
```

   The rest of this section will use a notation scheme to represent
   this tree. The parent (or trunk) node of the tree will be
   represented by a single integer, such as "1". Each child of that
   node will have an integer that identifies it, from 1 to the number
   of children. That child node will be uniquely identified by
   concatenating its integer to it's parents identifier with a dot in
   between, such as "1.1" and "1.2". Each node will be summarized by a
   tuple of three elements: endpoint, path, and protocol. The above
   example can now be written more succinctly as:

```
1 [www.example.com:80, Any, TCP]
   1.1 [www.example.com:80, Wi-Fi, TCP]
     1.1.1 [192.0.2.1:80, Wi-Fi, TCP]
   1.2 [www.example.com:80, LTE, TCP]
     1.2.1 [192.0.2.1:80, LTE, TCP]
     1.2.2 [2001:DB8::1.80, LTE, TCP]
```

   When an implementation views this aggregate set of connection
   attempts as a single connection establishment, it only will use one
   of the leaf nodes to transfer data. Thus, when a single leaf node
   becomes ready to use, then the entire connection attempt is ready to
   use by the application. Another way to represent this is that every
   leaf node updates the state of its parent node when it becomes
   ready, until the trunk node of the tree is ready, which then
   notifies the application that the connection as a whole is ready to
   use.

   A connection establishment tree may be degenerate, and only have a
   single leaf node, such as a connection attempt to an IP address over
   a single interface with a single protocol.

```
1 [192.0.2.1:80, Wi-Fi, TCP]
```

   A parent node may also only have one child (or leaf) node, such as a
   when a hostname resolves to only a single IP address.

```
1 [www.example.com:80, Wi-Fi, TCP]
   1.1 [192.0.2.1:80, Wi-Fi, TCP]
```

4.1.1. Branch Types

   There are three types of branching from a parent node into one or
   more child nodes. Any parent node of the tree must only use one type
   of branching.

4.1.1.1. Derived Endpoints

   If a connection originally targets a single endpoint, there may be
   multiple endpoints of different types that can be derived from the
   original. The connection library creates an ordered list of the
   derived endpoints according to application preference, system policy
   and expected performance.

   DNS hostname-to-address resolution is the most common method of
   endpoint derivation. When trying to connect to a hostname endpoint
   on a traditional IP network, the implementation should send DNS
   queries for both A (IPv4) and AAAA (IPv6) records if both are
   supported on the local interface. The algorithm for ordering and
   racing these addresses should follow the recommendations in Happy
   Eyeballs [RFC8305].

```
1 [www.example.com:80, Wi-Fi, TCP]
  1.1 [2001:DB8::1.80, Wi-Fi, TCP]
  1.2 [192.0.2.1:80, Wi-Fi, TCP]
  1.3 [2001:DB8::2.80, Wi-Fi, TCP]
  1.4 [2001:DB8::3.80, Wi-Fi, TCP]
```

   DNS-Based Service Discovery [RFC6763] can also provide an endpoint
   derivation step. When trying to connect to a named service, the
   client may discover one or more hostname and port pairs on the local
   network using multicast DNS [RFC6762]. These hostnames should each
   be treated as a branch that can be attempted independently from
   other hostnames. Each of these hostnames might resolve to one or
   more addresses, which would create multiple layers of branching.

```
1 [term-printer._ipp._tcp.meeting.ietf.org, Wi-Fi, TCP]
  1.1 [term-printer.meeting.ietf.org:631, Wi-Fi, TCP]
    1.1.1 [31.133.160.18.631, Wi-Fi, TCP]
```

4.1.1.2. Alternate Paths

   If a client has multiple network interfaces available to it, e.g., a
   mobile client with both Wi-Fi and Cellular connectivity, it can
   attempt a connection over any of the interfaces. This represents a
   branch point in the connection establishment. Similar to a derived
   endpoint, the interfaces should be ranked based on preference,
   system policy, and performance. Attempts should be started on one
   interface, and then on other interfaces successively after delays
   based on expected round-trip-time or other available metrics.

```
1 [192.0.2.1:80, Any, TCP]
  1.1 [192.0.2.1:80, Wi-Fi, TCP]
  1.2 [192.0.2.1:80, LTE, TCP]
```

   This same approach applies to any situation in which the client is
   aware of multiple links or views of the network. Multiple Paths,
   each with a coherent set of addresses, routes, DNS server, and more,
   may share a single interface. A path may also represent a virtual
   interface service such as a Virtual Private Network (VPN).

   The list of available paths should be constrained by any
   requirements or prohibitions the application sets, as well as system
   policy.

4.1.1.3. Protocol Options

   Differences in possible protocol compositions and options can also
   provide a branching point in connection establishment. This allows
   clients to be resilient to situations in which a certain protocol is
   not functioning on a server or network.

This approach is commonly used for connections with optional proxy
server configurations. A single connection might have several
options available: an HTTP-based proxy, a SOCKS-based proxy, or no
proxy. These options should be ranked and attempted in succession.

```
1 [www.example.com:80, Any, HTTP/TCP]
  1.1 [192.0.2.8:80, Any, HTTP/HTTP Proxy/TCP]
  1.2 [192.0.2.7:10234, Any, HTTP/SOCKS/TCP]
  1.3 [www.example.com:80, Any, HTTP/TCP]
    1.3.1 [192.0.2.1:80, Any, HTTP/TCP]
```

This approach also allows a client to attempt different sets of
application and transport protocols that, when available, could
provide preferable features. For example, the protocol options could
involve QUIC [I-D.ietf-quic-transport] over UDP on one branch, and
HTTP/2 [RFC7540] over TLS over TCP on the other:

```
1 [www.example.com:443, Any, Any HTTP]
  1.1 [www.example.com:443, Any, QUIC/UDP]
    1.1.1 [192.0.2.1:443, Any, QUIC/UDP]
  1.2 [www.example.com:443, Any, HTTP2/TLS/TCP]
    1.2.1 [192.0.2.1:443, Any, HTTP2/TLS/TCP]
```

Another example is racing SCTP with TCP:

```
1 [www.example.com:80, Any, Any Stream]
  1.1 [www.example.com:80, Any, SCTP]
    1.1.1 [192.0.2.1:80, Any, SCTP]
  1.2 [www.example.com:80, Any, TCP]
    1.2.1 [192.0.2.1:80, Any, TCP]
```

Implementations that support racing protocols and protocol options
should maintain a history of which protocols and protocol options
successfully established, on a per-network and per-endpoint basis
(see Section 9.2). This information can influence future racing
decisions to prioritize or prune branches.

4.1.2. Branching Order-of-Operations

Branch types must occur in a specific order relative to one another
to avoid creating leaf nodes with invalid or incompatible settings.
In the example above, it would be invalid to branch for derived
endpoints (the DNS results for www.example.com) before branching
between interface paths, since there are situations when the results
will be different across networks due to private names or different
supported IP versions. Implementations must be careful to branch in
an order that results in usable leaf nodes whenever there are
multiple branch types that could be used from a single node.

The order of operations for branching should be:

   1. Alternate Paths

   2. Protocol Options

   3. Derived Endpoints

   where a lower number indicates higher precedence and therefore
   higher placement in the tree. Branching between paths is the first
   in the list because results across multiple interfaces are likely
   not related to one another: endpoint resolution may return different
   results, especially when using locally resolved host and service
   names, and which protocols are supported and preferred may differ
   across interfaces. Thus, if multiple paths are attempted, the
   overall connection can be seen as a race between the available paths
   or interfaces.

   Protocol options are next checked in order. Whether or not a set of
   protocol, or protocol-specific options, can successfully connect is
   generally not dependent on which specific IP address is used.
   Furthermore, the protocol stacks being attempted may influence or
   altogether change the endpoints being used. Adding a proxy to a
   connection's branch will change the endpoint to the proxy's IP
   address or hostname. Choosing an alternate protocol may also modify
   the ports that should be selected.

   Branching for derived endpoints is the final step, and may have
   multiple layers of derivation or resolution, such as DNS service
   resolution and DNS hostname resolution.

   For example, if the application has indicated both a preference for
   WiFi over LTE and for a feature only available in SCTP, branches
   will be first sorted accord to path selection, with WiFi at the top.
   Then, branches with SCTP will be sorted to the top within their
   subtree according to the properties influencing protocol selection.
   However, if the implementation has current cache information that
   SCTP is not available on the path over WiFi, there is no SCTP node
   in the WiFi subtree. Here, the path over WiFi will be tried first,
   and, if connection establishment succeeds, TCP will be used. So the
   Selection Property of preferring WiFi takes precedence over the
   Property that led to a preference for SCTP.

1. [www.example.com:80, Any, Any Stream]
1.1 [192.0.2.1:80, Wi-Fi, Any Stream]
1.1.1 [192.0.2.1:80, Wi-Fi, TCP]
1.2 [192.0.3.1:80, LTE, Any Stream]
1.2.1 [192.0.3.1:80, LTE, SCTP]
1.2.2 [192.0.3.1:80, LTE, TCP]

4.1.3. Sorting Branches

Implementations should sort the branches of the tree of connection options in order of their preference rank, from most preferred to least preferred. Leaf nodes on branches with higher rankings represent connection attempts that will be raced first. Implementations should order the branches to reflect the preferences expressed by the application for its new connection, including Selection Properties, which are specified in [I-D.ietf-taps-interface].

In addition to the properties provided by the application, an implementation may include additional criteria such as cached performance estimates, see Section 9.2, or system policy, see Section 3.2, in the ranking. Two examples of how Selection and Connection Properties may be used to sort branches are provided below:

  * "Interface Instance or Type": If the application specifies an interface type to be preferred or avoided, implementations should accordingly rank the paths. If the application specifies an interface type to be required or prohibited, an implementation is expeceted to not include the non-conforming paths.

  * "Capacity Profile": An implementation can use the Capacity Profile to prefer paths that match an application's expected traffic pattern. This match will use cached performance estimates, see Section 9.2:

    - Scavenger: Prefer paths with the highest expected available capacity, but minimising impact on other traffic, based on the observed maximum throughput;

    - Low Latency/Interactive: Prefer paths with the lowest expected Round Trip Time, based on observed round trip time estimates;

    - Low Latency/Non-Interactive: Prefer paths with a low expected Round Trip Time, but can tolerate delay variation;

    - Constant-Rate Streaming: Prefer paths that are expected to satisy the requested Stream Send or Stream Receive Bitrate, based on the observed maximum throughput;

    - Capacity-Seeking: Prefer adapting to paths to determine the highest available capacity, based on the observed maximum throughput.

Implementations process the Properties in the following order: Prohibit, Require, Prefer, Avoid. If Selection Properties contain any prohibited properties, the implementation should first purge

branches containing nodes with these properties. For required
properties, it should only keep branches that satisfy these
requirements. Finally, it should order the branches according to the
preferred properties, and finally use any avoided properties as a
tiebreaker. When ordering branches, an implementation can give more
weight to properties that the application has explicitly set, than
to the properties that are default.

The available protocols and paths on a specific system and in a
specific context can change; therefore, the result of sorting and
the outcome of racing may vary, even when using the same Selection
and Connection Properties. However, an implementation ought to
provide a consistent outcome to applications, e.g., by preferring
protocols and paths that are already used by existing Connections
that specified similar Properties.

4.2. Candidate Gathering

The step of gathering candidates involves identifying which paths,
protocols, and endpoints may be used for a given Connection. This
list is determined by the requirements, prohibitions, and
preferences of the application as specified in the Selection
Properties.

4.2.1. Gathering Endpoint Candidates

Both Local and Remote Endpoint Candidates must be discovered during
connection establishment. To support Interactive Connectivity
Establishment (ICE) [RFC8445], or similar protocols that involve
out-of-band indirect signalling to exchange candidates with the
Remote Endpoint, it is important to query the set of candidate Local
Endpoints, and provide the protocol stack with a set of candidate
Remote Endpoints, before the Local Endpoint attempts to establish
connections.

4.2.1.1. Local Endpoint candidates

The set of possible Local Endpoints is gathered. In the simple case,
this merely enumerates the local interfaces and protocols, and
allocates ephemeral source ports. For example, a system that has
WiFi and Ethernet and supports IPv4 and IPv6 might gather four
candidate Local Endpoints (IPv4 on Ethernet, IPv6 on Ethernet, IPv4
on WiFi, and IPv6 on WiFi) that can form the source for a transient.

If NAT traversal is required, the process of gathering Local
Endpoints becomes broadly equivalent to the ICE candidate gathering
phase (see Section 5.1.1. of [RFC8445]). The endpoint determines its
server reflexive Local Endpoints (i.e., the translated address of a
Local Endpoint, on the other side of a NAT, e.g via a STUN sever
[RFC5389]) and relayed Local Endpoints (e.g., via a TURN server

[RFC5766] or other relay), for each interface and network protocol. These are added to the set of candidate Local Endpoints for this connection.

Gathering Local Endpoints is primarily a local operation, although it might involve exchanges with a STUN server to derive server reflexive Local Endpoints, or with a TURN server or other relay to derive relayed Local Endpoints. However, it does not involve communication with the Remote Endpoint.

4.2.1.2. Remote Endpoint Candidates

The Remote Endpoint is typically a name that needs to be resolved into a set of possible addresses that can be used for communication. Resolving the Remote Endpoint is the process of recursively performing such name lookups, until fully resolved, to return the set of candidates for the Remote Endpoint of this connection.

How this resolution is done will depend on the type of the Remote Endpoint, and can also be specific to each Local Endpoint. A common case is when the Remote Endpoint is a DNS name, in which case it is resolved to give a set of IPv4 and IPv6 addresses representing that name. Some types of Remote Endpoint might require more complex resolution. Resolving the Remote Endpoint for a peer-to-peer connection might involve communication with a rendezvous server, which in turn contacts the peer to gain consent to communicate and retrieve its set of candidate Local Endpoints, which are returned and form the candidate remote addresses for contacting that peer.

Resolving the Remote Endpoint is not a local operation. It will involve a directory service, and can require communication with the Remote Endpoint to rendezvous and exchange peer addresses. This can expose some or all of the candidate Local Endpoints to the Remote Endpoint.

4.3. Candidate Racing

The primary goal of the Candidate Racing process is to successfully negotiate a protocol stack to an endpoint over an interface to connect a single leaf node of the tree with as little delay and as few unnecessary connections attempts as possible. Optimizing these two factors improves the user experience, while minimizing network load.

This section covers the dynamic aspect of connection establishment. The tree described above is a useful conceptual and architectural model. However, an implementation is unable to know the full tree before it is formed and many of the possible branches ultimately might not be used.

There are three different approaches to racing the attempts for different nodes of the connection establishment tree:

1. Simultaneous

2. Staggered

3. Failover

Each approach is appropriate in different use-cases and branch types. However, to avoid consuming unnecessary network resources, implementations should not use simultaneous racing as a default approach.

The timing algorithms for racing should remain independent across branches of the tree. Any timers or racing logic is isolated to a given parent node, and is not ordered precisely with regards to other children of other nodes.

4.3.1. Simultaneous

Simultaneous racing is when multiple alternate branches are started without waiting for any one branch to make progress before starting the next alternative. This means the attempts are effectively simultaneous. Simultaneous racing should be avoided by implementations, since it consumes extra network resources and establishes state that might not be used.

4.3.2. Staggered

Staggered racing can be used whenever a single node of the tree has multiple child nodes. Based on the order determined when building the tree, the first child node will be initiated immediately, followed by the next child node after some delay. Once that second child node is initiated, the third child node (if present) will begin after another delay, and so on until all child nodes have been initiated, or one of the child nodes successfully completes its negotiation.

Staggered racing attempts can proceed in parallel. Implementations should not terminate an earlier child connection attempt upon starting a secondary child.

If a child node fails to establish connectivity (as in Section 4.4.1) before the delay time has expired for the next child, the next child should be started immediately.

Staggered racing between IP addresses for a generic Connection should follow the Happy Eyeballs algorithm described in [RFC8305].

[RFC8421] provides guidance for racing when performing Interactive Connectivity Establishment (ICE).

Generally, the delay before starting a given child node ought to be based on the length of time the previously started child node is expected to take before it succeeds or makes progress in connection establishment. Algorithms like Happy Eyeballs choose a delay based on how long the transport connection handshake is expected to take. When performing staggered races in multiple branch types (such as racing between network interfaces, and then racing between IP addresses), a longer delay may be chosen for some branch types. For example, when racing between network interfaces, the delay should also take into account the amount of time it takes to prepare the network interface (such as radio association) and name resolution over that interface, in addition to the delay that would be added for a single transport connection handshake.

Since the staggered delay can be chosen based on dynamic information, such as predicted round-trip time, implementations should define upper and lower bounds for delay times. These bounds are implementation-specific, and may differ based on which branch type is being used.

4.3.3. Failover

If an implementation or application has a strong preference for one branch over another, the branching node may choose to wait until one child has failed before starting the next. Failure of a leaf node is determined by its protocol negotiation failing or timing out; failure of a parent branching node is determined by all of its children failing.

An example in which failover is recommended is a race between a protocol stack that uses a proxy and a protocol stack that bypasses the proxy. Failover is useful in case the proxy is down or misconfigured, but any more aggressive type of racing may end up unnecessarily avoiding a proxy that was preferred by policy.

4.4. Completing Establishment

The process of connection establishment completes when one leaf node of the tree has successfully completed negotiation with the Remote Endpoint, or else all nodes of the tree have failed to connect. The first leaf node to complete its connection is then used by the application to send and receive data.

Successes and failures of a given attempt should be reported up to parent nodes (towards the trunk of the tree). For example, in the following case, if 1.1.1 fails to connect, it reports the failure to 1.1. Since 1.1 has no other child nodes, it also has failed and

reports that failure to 1. Because 1.2 has not yet failed, 1 is not
considered to have failed. Since 1.2 has not yet started, it is
started and the process continues. Similarly, if 1.1.1 successfully
connects, then it marks 1.1 as connected, which propagates to the
trunk node 1. At this point, the connection as a whole is considered
to be successfully connected and ready to process application data.

```
1 [www.example.com:80, Any, TCP]
  1.1 [www.example.com:80, Wi-Fi, TCP]
    1.1.1 [192.0.2.1:80, Wi-Fi, TCP]
  1.2 [www.example.com:80, LTE, TCP]
...
```

If a leaf node has successfully completed its connection, all other
attempts should be made ineligible for use by the application for
the original request. New connection attempts that involve
transmitting data on the network ought not to be started after
another leaf node has already successfully completed, because the
connection as a whole has now been established. An implementation
may choose to let certain handshakes and negotiations complete in
order to gather metrics to influence future connections. Keeping
additional connections is generally not recommended since those
attempts were slower to connect and may exhibit less desirable
properties.

4.4.1. Determining Successful Establishment

Implementations may select the criteria by which a leaf node is
considered to be successfully connected differently on a per-
protocol basis. If the only protocol being used is a transport
protocol with a clear handshake, like TCP, then the obvious choice
is to declare that node "connected" when the last packet of the
three-way handshake has been received. If the only protocol being
used is an connectionless protocol, like UDP, the implementation may
consider the node fully "connected" the moment it determines a route
is present, before sending any packets on the network, see further
[Section 4.6](#).

For protocol stacks with multiple handshakes, the decision becomes
more nuanced. If the protocol stack involves both TLS and TCP, an
implementation could determine that a leaf node is connected after
the TCP handshake is complete, or it can wait for the TLS handshake
to complete as well. The benefit of declaring completion when the
TCP handshake finishes, and thus stopping the race for other
branches of the tree, is reduced burden on the network and Remote
Endpoints from further connection attempts that are likely to be
abandoned. On the other hand, by waiting until the TLS handshake is
complete, an implementation avoids the scenario in which a TCP
handshake completes quickly, but TLS negotiation is either very slow

or fails altogether in particular network conditions or to a particular endpoint. To avoid the issue of TLS possibly failing, the implementation should not generate a Ready event for the Connection until TLS is established.

If all of the leaf nodes fail to connect during racing, i.e. none of the configurations that satisfy all requirements given in the Transport Properties actually work over the available paths, then the transport system should notify the application with an InitiateError event. An InitiateError event should also be generated in case the transport system finds no usable candidates to race.

4.5. Establishing multiplexed connections

Multiplexing several Connections over a single underlying transport connection requires that the Connections to be multiplexed belong to the same Connection Group (as is indicated by the application using the Clone call). When the underlying transport connection supports multi-streaming, the Transport Services System can map each Connection in the Connection Group to a different stream. Thus, when the Connections that are offered to an application by the Transport Services API are multiplexed, the Transport Services implementation can establish a new Connection by simply beginning to use a new stream of an already established transport Connection and there is no need for a connection establishment procedure. This, then, also means that there may not be any "establishment" message (like a TCP SYN), but the application can simply start sending or receiving. Therefore, when the Initiate action of a Transport Services API is called without Messages being handed over, it cannot be guaranteed that the Remote Endpoint will have any way to know about this, and hence a passive endpoint's ConnectionReceived event might not be called until data is received. Instead, calling the ConnectionReceived event could be delayed until the first Message arrives.

4.6. Handling connectionless protocols

While protocols that use an explicit handshake to validate a Connection to a peer can be used for racing multiple establishment attempts in parallel, connectionless protocols such as raw UDP do not offer a way to validate the presence of a peer or the usability of a Connection without application feedback. An implementation should consider such a protocol stack to be established as soon as the Transport Services system has selected a path on which to send data.

However, if a peer is not reachable over the network using the connectionless protocol, or data cannot be exchanged for any other reason, the application may want to attempt using another candidate

Protocol Stack. The implementation should maintain the list of other candidate Protocol Stacks that were eligible to use.

4.7. Implementing listeners

When an implementation is asked to Listen, it registers with the system to wait for incoming traffic to the Local Endpoint. If no Local Endpoint is specified, the implementation should use an ephemeral port.

If the Selection Properties do not require a single network interface or path, but allow the use of multiple paths, the Listener object should register for incoming traffic on all of the network interfaces or paths that conform to the Properties. The set of available paths can change over time, so the implementation should monitor network path changes, and change the registration of the Listener across all usable paths as appropriate. When using multiple paths, the Listener is generally expected to use the same port for listening on each.

If the Selection Properties allow multiple protocols to be used for listening, and the implementation supports it, the Listener object should support receiving inbound connections for each eligible protocol on each eligible path.

4.7.1. Implementing listeners for Connected Protocols

Connected protocols such as TCP and TLS-over-TCP have a strong mapping between the Local and Remote Endpoints (four-tuple) and their protocol connection state. These map into Connection objects. Whenever a new inbound handshake is being started, the Listener should generate a new Connection object and pass it to the application.

4.7.2. Implementing listeners for Connectionless Protocols

Connectionless protocols such as UDP and UDP-lite generally do not provide the same mechanisms that connected protocols do to offer Connection objects. Implementations should wait for incoming packets for connectionless protocols on a listening port and should perform four-tuple matching of packets to either existing Connection objects or the creation of new Connection objects. On platforms with facilities to create a "virtual connection" for connectionless protocols implementations should use these mechanisms to minimise the handling of datagrams intended for already created Connection objects.

4.7.3. Implementing listeners for Multiplexed Protocols

Protocols that provide multiplexing of streams into a single four-tuple can listen both for entirely new connections (a new HTTP/2 stream on a new TCP connection, for example) and for new sub-connections (a new HTTP/2 stream on an existing connection). If the abstraction of Connection presented to the application is mapped to the multiplexed stream, then the Listener should deliver new Connection objects in the same way for either case. The implementation should allow the application to introspect the Connection Group marked on the Connections to determine the grouping of the multiplexing.

5. Implementing Sending and Receiving Data

The most basic mapping for sending a Message is an abstraction of datagrams, in which the transport protocol naturally deals in discrete packets. Each Message here corresponds to a single datagram. Generally, these will be short enough that sending and receiving will always use a complete Message.

For protocols that expose byte-streams, the only delineation provided by the protocol is the end of the stream in a given direction. Each Message in this case corresponds to the entire stream of bytes in a direction. These Messages may be quite long, in which case they can be sent in multiple parts.

Protocols that provide the framing (such as length-value protocols, or protocols that use delimiters) may support Message sizes that do not fit within a single datagram. Each Message for framing protocols corresponds to a single frame, which may be sent either as a complete Message in the underlying protocol, or in multiple parts.

5.1. Sending Messages

The effect of the application sending a Message is determined by the top-level protocol in the established Protocol Stack. That is, if the top-level protocol provides an abstraction of framed messages over a connection, the receiving application will be able to obtain multiple Messages on that connection, even if the framing protocol is built on a byte-stream protocol like TCP.

5.1.1. Message Properties

   *Lifetime: this should be implemented by removing the Message from the queue of pending Messages after the Lifetime has expired. A queue of pending Messages within the transport system implementation that have yet to be handed to the Protocol Stack can always support this property, but once a Message has been sent into the send buffer of a protocol, only certain protocols

may support removing a message. For example, an implementation
cannot remove bytes from a TCP send buffer, while it can remove
data from a SCTP send buffer using the partial reliability
extension [RFC8303]. When there is no standing queue of Messages
within the system, and the Protocol Stack does not support the
removal of a Message from the stack's send buffer, this property
may be ignored.

*Priority: this represents the ability to prioritize a Message
 over other Messages. This can be implemented by the system re-
 ordering Messages that have yet to be handed to the Protocol
 Stack, or by giving relative priority hints to protocols that
 support priorities per Message. For example, an implementation of
 HTTP/2 could choose to send Messages of different Priority on
 streams of different priority.

*Ordered: when this is false, this disables the requirement of in-
 order-delivery for protocols that support configurable ordering.
 When the protocol stack does not support configurable ordering,
 this property may be ignored.

*Safely Replayable: when this is true, this means that the Message
 can be used by a transport mechanism that might transfer it
 multiple times - e.g., as a result of racing multiple transports
 or as part of TCP Fast Open. Also, protocols that do not protect
 against duplicated messages, such as UDP (when used directly,
 without a protocol layered atop), can only be used with Messages
 that are Safely Replayable. When a transport system is permitted
 to replay messages, replay protection could be provided by the
 application.

*Final: when this is true, this means that the sender will not
 send any further messages. The Connection need not be closed (in
 case the Protocol Stack supports half-close operation, like TCP).
 Any messages sent after a Final message will result in a
 SendError.

*Corruption Protection Length: when this is set to any value other
 than Full Coverage, it sets the minimum protection in protocols
 that allow limiting the checksum length (e.g. UDP-Lite). If the
 protocol stack does not support checksum length limitation, this
 property may be ignored.

*Reliable Data Transfer (Message): When true, the property
 specifies that the Message must be reliably transmitted. When
 false, and if unreliable transmission is supported by the
 underlying protocol, then the Message should be unreliably
 transmitted. If the underlying protocol does not support

unreliable transmission, the Message should be reliably
transmitted.

*Message Capacity Profile Override: When true, this expresses a
wish to override the Generic Connection Property Capacity Profile
for this Message. Depending on the value, this can, for example,
be implemented by changing the DSCP value of the associated
packet (note that the guidelines in Section 6 of [RFC7657] apply;
e.g., the DSCP value should not be changed for different packets
within a reliable transport protocol session or DCCP connection).

*No Fragmentation: When set, this property limits the message size
to the Maximum Message Size Before Fragmentation or Segmentation
(see Section 10.1.7 of [I-D.ietf-taps-interface]). Messages
larger than this size generate an error. Setting this avoids
transport-layer segmentation or network-layer fragmentation. When
used with transports running over IP version 4 the Don't Fragment
bit will be set to avoid on-path IP fragmentation ([RFC8304]).

5.1.2. Send Completion

The application should be notified whenever a Message or partial
Message has been consumed by the Protocol Stack, or has failed to
send. The time at which a Message is considered to have been
consumed by the Protocol Stack may vary depending on the protocol.
For example, for a basic datagram protocol like UDP, this may
correspond to the time when the packet is sent into the interface
driver. For a protocol that buffers data in queues, like TCP, this
may correspond to when the data has entered the send buffer. The
time at which a message failed to send is when Transport Services
implementation (including the Protocol Stack) has not successfully
sent the entire Message content or partial Message content on any
open candidate connection; this can depend on protocol-specific
timeouts.

5.1.3. Batching Sends

Since sending a Message may involve a context switch between the
application and the Transport Services system, sending patterns that
involve multiple small Messages can incur high overhead if each
needs to be enqueued separately. To avoid this, the application can
indicate a batch of Send actions through the API. When this is used,
the implementation can defer the processing of Messages until the
batch is complete.

5.2. Receiving Messages

Similar to sending, Receiving a Message is determined by the top-
level protocol in the established Protocol Stack. The main
difference with Receiving is that the size and boundaries of the

Message are not known beforehand. The application can communicate in
its Receive action the parameters for the Message, which can help
the Transport Services implementation know how much data to deliver
and when. For example, if the application only wants to receive a
complete Message, the implementation should wait until an entire
Message (datagram, stream, or frame) is read before delivering any
Message content to the application. This requires the implementation
to understand where messages end, either via a supplied deframer or
because the top-level protocol in the established Protocol Stack
preserves message boundaries. If the top-level protocol only
supports a byte-stream and no framers were supported, the
application can control the flow of received data by specifying the
minimum number of bytes of Message content it wants to receive at
one time.

If a Connection finishes before a requested Receive action can be
satisfied, the Transport Services API should deliver any partial
Message content outstanding, or if none is available, an indication
that there will be no more received Messages.

5.3. Handling of data for fast-open protocols

Several protocols allow sending higher-level protocol or application
data during their protocol establishment, such as TCP Fast Open
[RFC7413] and TLS 1.3 [RFC8446]. This approach is referred to as
sending Zero-RTT (0-RTT) data. This is a desirable feature, but
poses challenges to an implementation that uses racing during
connection establishment.

The amount of data that can be sent as 0-RTT data varies by protocol
and can be queried by the application using the Maximum Message Size
Concurrent with Connection Establishment Connection Property. An
implementation can set this property according to the protocols that
it will race based on the given Selection Properties when the
application requests to establish a connection.

If the application has 0-RTT data to send in any protocol
handshakes, it needs to provide this data before the handshakes have
begun. When racing, this means that the data should be provided
before the process of connection establishment has begun. If the
application wants to send 0-RTT data, it must indicate this to the
implementation by setting the Safely Replayable send parameter to
true when sending the data. In general, 0-RTT data may be replayed
(for example, if a TCP SYN contains data, and the SYN is
retransmitted, the data will be retransmitted as well but may be
considered as a new connection instead of a retransmission). Also,
when racing connections, different leaf nodes have the opportunity
to send the same data independently. If data is truly safely
replayable, this should be permissible.

Once the application has provided its 0-RTT data, a Transport Services implementation should keep a copy of this data and provide it to each new leaf node that is started and for which a 0-RTT protocol is being used.

It is also possible that protocol stacks within a particular leaf node use 0-RTT handshakes without any safely replayable application data. For example, TCP Fast Open could use a Client Hello from TLS as its 0-RTT data, shortening the cumulative handshake time.

0-RTT handshakes often rely on previous state, such as TCP Fast Open cookies, previously established TLS tickets, or out-of-band distributed pre-shared keys (PSKs). Implementations should be aware of security concerns around using these tokens across multiple addresses or paths when racing. In the case of TLS, any given ticket or PSK should only be used on one leaf node, since servers will likely reject duplicate tickets in order to prevent replays (see section-8.1 [RFC8446]). If implementations have multiple tickets available from a previous connection, each leaf node attempt can use a different ticket. In effect, each leaf node will send the same early application data, yet encoded (encrypted) differently on the wire.

6. **Implementing Message Framers**

Message Framers are functions that define simple transformations between application Message data and raw transport protocol data. A Framer can encapsulate or encode outbound Messages, and decapsulate or decode inbound data into Messages.

While many protocols can be represented as Message Framers, for the purposes of the Transport Services API, these are ways for applications or application frameworks to define their own Message parsing to be included within a Connection's Protocol Stack. As an example, TLS is exposed as a protocol natively supported by the Transport Services API, even though it could also serve the purpose of framing data over TCP.

Most Message Framers fall into one of two categories:

  *Header-prefixed record formats, such as a basic Type-Length-Value
   (TLV) structure

  *Delimiter-separated formats, such as HTTP/1.1.

Common Message Framers can be provided by a Transport Services implementation, but an implementation ought to allow custom Message Framers to be defined by the application or some other piece of software. This section describes one possible API for defining Message Framers as an example.

6.1. Defining Message Framers

A Message Framer is primarily defined by the code that handles
events for a framer implementation, specifically how it handles
inbound and outbound data parsing. The function that implements
custom framing logic will be referred to as the "framer
implementation", which may be provided by a Transport Services
implementation or the application itself. The Message Framer refers
to the object or function within the main Connection implementation
that delivers events to the custom framer implementation whenever
data is ready to be parsed or framed.

When a Connection establishment attempt begins, an event can be
delivered to notify the framer implementation that a new Connection
is being created. Similarly, a stop event can be delivered when a
Connection is being torn down. The framer implementation can use the
Connection object to look up specific properties of the Connection
or the network being used that may influence how to frame Messages.

```
MessageFramer -> Start(Connection)
MessageFramer -> Stop(Connection)
```

When a Message Framer generates a Start event, the framer
implementation has the opportunity to start writing some data prior
to the Connection delivering its Ready event. This allows the
implementation to communicate control data to the Remote Endpoint
that can be used to parse Messages.

```
MessageFramer.MakeConnectionReady(Connection)
```

Similarly, when a Message Framer generates a Stop event, the framer
implementation has the opportunity to write some final data or clear
up its local state before the Closed event is delivered to the
Application. The framer implementation can indicate that it has
finished with this.

```
MessageFramer.MakeConnectionClosed(Connection)
```

At any time if the implementation encounters a fatal error, it can
also cause the Connection to fail and provide an error.

```
MessageFramer.FailConnection(Connection, Error)
```

Should the framer implementation deem the candidate selected during
racing unsuitable, it can signal this to the Transport Services API
by failing the Connection prior to marking it as ready. If there are
no other candidates available, the Connection will fail. Otherwise,
the Connection will select a different candidate and the Message
Framer will generate a new Start event.

Before an implementation marks a Message Framer as ready, it can
also dynamically add a protocol or framer above it in the stack.
This allows protocols that need to add TLS conditionally, like
STARTTLS [RFC3207], to modify the Protocol Stack based on a
handshake result.

```
otherFramer := NewMessageFramer()
MessageFramer.PrependFramer(Connection, otherFramer)
```

A Message Framer might also choose to go into a passthrough mode
once an initial exchange or handshake has been completed, such as
the STARTTLS case mentioned above. This can also be useful for proxy
protocols like SOCKS [RFC1928] or HTTP CONNECT [RFC7230]. In such
cases, a Message Framer implementation can intercept sending and
receiving of messages at first, but then indicate that no more
processing is needed.

```
MessageFramer.StartPassthrough()
```

6.2. Sender-side Message Framing

Message Framers generate an event whenever a Connection sends a new
Message.

```
MessageFramer -> NewSentMessage<Connection, MessageData, MessageContext,
```

Upon receiving this event, a framer implementation is responsible
for performing any necessary transformations and sending the
resulting data back to the Message Framer, which will in turn send
it to the next protocol. Implementations SHOULD ensure that there is
a way to pass the original data through without copying to improve
performance.

```
MessageFramer.Send(Connection, Data)
```

To provide an example, a simple protocol that adds a length as a
header would receive the NewSentMessage event, create a data
representation of the length of the Message data, and then send a
block of data that is the concatenation of the length header and the
original Message data.

6.3. Receiver-side Message Framing

In order to parse a received flow of data into Messages, the Message
Framer notifies the framer implementation whenever new data is
available to parse.

```
MessageFramer -> HandleReceivedData<Connection>
```

Upon receiving this event, the framer implementation can inspect the inbound data. The data is parsed from a particular cursor representing the unprocessed data. The application requests a specific amount of data it needs to have available in order to parse. If the data is not available, the parse fails.

MessageFramer.Parse(Connection, MinimumIncompleteLength, MaximumLength)

The framer implementation can directly advance the receive cursor once it has parsed data to effectively discard data (for example, discard a header once the content has been parsed).

To deliver a Message to the application, the framer implementation can either directly deliver data that it has allocated, or deliver a range of data directly from the underlying transport and simultaneously advance the receive cursor.

MessageFramer.AdvanceReceiveCursor(Connection, Length)
MessageFramer.DeliverAndAdvanceReceiveCursor(Connection, MessageContext, MessageFramer.Deliver(Connection, MessageContext, Data, IsEndOfMessage)

Note that MessageFramer.DeliverAndAdvanceReceiveCursor allows the framer implementation to earmark bytes as part of a Message even before they are received by the transport. This allows the delivery of very large Messages without requiring the implementation to directly inspect all of the bytes.

To provide an example, a simple protocol that parses a length as a header value would receive the HandleReceivedData event, and call Parse with a minimum and maximum set to the length of the header field. Once the parse succeeded, it would call AdvanceReceiveCursor with the length of the header field, and then call DeliverAndAdvanceReceiveCursor with the length of the body that was parsed from the header, marking the new Message as complete.

7. Implementing Connection Management

Once a Connection is established, the Transport Services API allows applications to interact with the Connection by modifying or inspecting Connection Properties. A Connection can also generate events in the form of Soft Errors.

The set of Connection Properties that are supported for setting and getting on a Connection are described in [I-D.ietf-taps-interface]. For any properties that are generic, and thus could apply to all protocols being used by a Connection, the Transport Services implementation should store the properties in storage common to all protocols, and notify all protocol instances in the Protocol Stack whenever the properties have been modified by the application. For protocol-specfic properties, such as the User Timeout that applies

to TCP, the Transport Services implementation only needs to update
the relevant protocol instance.

If an error is encountered in setting a property (for example, if
the application tries to set a TCP-specific property on a Connection
that is not using TCP), the action should fail gracefully. The
application may be informed of the error, but the Connection itself
should not be terminated.

The Transport Services API should allow protocol instances in the
Protocol Stack to pass up arbitrary generic or protocol-specific
errors that can be delivered to the application as Soft Errors.
These allow the application to be informed of ICMP errors, and other
similar events.

7.1. Pooled Connection

For applications that do not need in-order delivery of Messages, the
Transport Services implementation may distribute Messages of a
single Connection across several underlying transport connections or
multiple streams of multi-streaming connections between endpoints,
as long as all of these satisfy the Selection Properties. The
Transport Services implementation will then hide this connection
management and only expose a single Connection object, which we here
call a "Pooled Connection". This is in contrast to Connection
Groups, which explicitly expose combined treatment of Connections,
giving the application control over multiplexing, for example.

Pooled Connections can be useful when the application using the
Transport Services system implements a protocol such as HTTP, which
employs request/response pairs and does not require in-order
delivery of responses. This enables implementations of Transport
Services systems to realize transparent connection coalescing,
connection migration, and to perform per-message endpoint and path
selection by choosing among multiple underlying connections.

7.2. Handling Path Changes

When a path change occurs, e.g., when the IP address of an interface
changes or a new interface becomes available, the Transport Services
implementation is responsible for notifying the Protocol Instance of
the change. The path change may interrupt connectivity on a path for
an active connection or provide an opportunity for a transport that
supports multipath or migration to adapt to the new paths. Note
that, in the model of the Transport Services API, migration is
considered a part of multipath connectivity; it is just a limiting
policy on multipath usage. If the multipath Selection Property is
set to Disabled, migration is disallowed.

For protocols that do not support multipath or migration, the Protocol Instances should be informed of the path change, but should not be forcibly disconnected if the previously used path becomes unavailable. There are many common user scenarios that can lead to a path becoming temporarily unavailable, and then recovering before the transport protocol reaches a timeout error. These are particularly common using mobile devices. Examples include: an Ethernet cable becoming unplugged and then plugged back in; a device losing a Wi-Fi signal while a user is in an elevator, and reattaching when the user leaves the elevator; and a user losing the radio signal while riding a train through a tunnel. If the device is able to rejoin a network with the same IP address, a stateful transport connection can generally resume. Thus, while it is useful for a Protocol Instance to be aware of a temporary loss of connectivity, the Transport Services implementation should not aggressively close connections in these scenarios.

If the Protocol Stack includes a transport protocol that supports multipath connectivity, the Transport Services implementation should also inform the Protocol Instance of potentially new paths that become permissible based on the multipath Selection Property and the multipath-policy Connection Property choices made by the application. A protocol can then establish new subflows over new paths while an active path is still available or, if migration is supported, also after a break has been detected, and should attempt to tear down subflows over paths that are no longer used. The Connection Property multipath-policy of the Transport Services API allows an application to indicate when and how different paths should be used. However, detailed handling of these policies is still implementation-specific. For example, if the multipath Selection Property is set to active, the decision about when to create a new path or to announce a new path or set of paths to the Remote Endpoint, e.g., in the form of additional IP addresses, is implementation-specific. If the Protocol Stack includes a transport protocol that does not support multipath, but does support migrating between paths, the update to the set of available paths can trigger the connection to be migrated.

In case of Pooled Connections [Section 7.1](), the Transport Services implementation may add connections over new paths to the pool if permissible based on the multipath policy and Selection Properties. In case a previously used path becomes unavailable, the transport system may disconnect all connections that require this path, but should not disconnect the pooled connection object exposed to the application. The strategy to do so is implementation-specific, but should be consistent with the behavior of multipath transports.

8. Implementing Connection Termination

With TCP, when an application closes a connection, this means that
it has no more data to send (but expects all data that has been
handed over to be reliably delivered). However, with TCP only,
"close" does not mean that the application will stop receiving data.
This is related to TCP's ability to support half-closed connections.

SCTP is an example of a protocol that does not support such half-
closed connections. Hence, with SCTP, the meaning of "close" is
stricter: an application has no more data to send (but expects all
data that has been handed over to be reliably delivered), and will
also not receive any more data.

Implementing a protocol independent transport system means that the
exposed semantics must be the strictest subset of the semantics of
all supported protocols. Hence, as is common with all reliable
transport protocols, after a Close action, the application can
expect to have its reliability requirements honored regarding the
data provided to the Transport Services API, but it cannot expect to
be able to read any more data after calling Close.

Abort differs from Close only in that no guarantees are given
regarding any data that the application sent to the Transport
Services API before calling Abort.

As explained in Section 4.5, when a new stream is multiplexed on an
already existing connection of a Transport Protocol Instance, there
is no need for a connection establishment procedure. Because the
Connections that are offered by a Transport Services implementation
can be implemented as streams that are multiplexed on a transport
protocol's connection, it can therefore not be guaranteed an
Initiate action from one endpoint provokes a ConnectionReceived
event at its peer.

For Close (provoking a Finished event) and Abort (provoking a
ConnectionError event), the same logic applies: while it is
desirable to be informed when a peer closes or aborts a Connection,
whether this is possible depends on the underlying protocol, and no
guarantees can be given. With SCTP, the transport system can use the
stream reset procedure to cause a Finish event upon a Close action
from the peer [NEAT-flow-mapping].

9. Cached State

Beyond a single Connection's lifetime, it is useful for an
implementation to keep state and history. This cached state can help
improve future Connection establishment due to re-using results and
credentials, and favoring paths and protocols that performed well in
the past.

Cached state may be associated with different endpoints for the same Connection, depending on the protocol generating the cached content. For example, session tickets for TLS are associated with specific endpoints, and thus should be cached based on a Connection's hostname endpoint (if applicable). However, performance characteristics of a path are more likely tied to the IP address and subnet being used.

9.1. Protocol state caches

Some protocols will have long-term state to be cached in association with endpoints. This state often has some time after which it is expired, so the implementation should allow each protocol to specify an expiration for cached content.

Examples of cached protocol state include:

  *The DNS protocol can cache resolution answers (A and AAAA queries, for example), associated with a Time To Live (TTL) to be used for future hostname resolutions without requiring asking the DNS resolver again.

  *TLS caches session state and tickets based on a hostname, which can be used for resuming sessions with a server.

  *TCP can cache cookies for use in TCP Fast Open.

Cached protocol state is primarily used during Connection establishment for a single Protocol Stack, but may be used to influence an implementation's preference between several candidate Protocol Stacks. For example, if two IP address endpoints are otherwise equally preferred, an implementation may choose to attempt a connection to an address for which it has a TCP Fast Open cookie.

Applications can use the Transport Services API to request that a Connection Group maintain a separate cache for protocol state. Connections in the group will not use cached state from connections outside the group, and connections outside the group will not use state cached from connections inside the group. This may be necessary, for example, if application-layer identifiers rotate and clients wish to avoid linkability via trackable TLS tickets or TFO cookies.

9.2. Performance caches

In addition to protocol state, Protocol Instances should provide data into a performance-oriented cache to help guide future protocol and path selection. Some performance information can be gathered

generically across several protocols to allow predictive comparisons
between protocols on given paths:

   *Observed Round Trip Time

   *Connection Establishment latency

   *Connection Establishment success rate

These items can be cached on a per-address and per-subnet
granularity, and averaged between different values. The information
should be cached on a per-network basis, since it is expected that
different network attachments will have different performance
characteristics. Besides Protocol Instances, other system entities
may also provide data into performance-oriented caches. This could
for instance be signal strength information reported by radio modems
like Wi-Fi and mobile broadband or information about the battery-
level of the device. Furthermore, the system may cache the observed
maximum throughput on a path as an estimate of the available
bandwidth.

An implementation should use this information, when possible, to
influence preference between candidate paths, endpoints, and
protocol options. Eligible options that historically had
significantly better performance than others should be selected
first when gathering candidates (see Section 4.2) to ensure better
performance for the application.

The reasonable lifetime for cached performance values will vary
depending on the nature of the value. Certain information, like the
connection establishment success rate to a Remote Endpoint using a
given protocol stack, can be stored for a long period of time (hours
or longer), since it is expected that the capabilities of the Remote
Endpoint are not changing very quickly. On the other hand, the Round
Trip Time observed by TCP over a particular network path may vary
over a relatively short time interval. For such values, the
implementation should remove them from the cache more quickly, or
treat older values with less confidence/weight.

[I-D.ietf-tcpm-2140bis] provides guidance about sharing of TCP
Control Block information between connections on initialization.

10. Specific Transport Protocol Considerations

Each protocol that is supported by a Transport Services
implementation should have a well-defined API mapping. API mappings
for a protocol are important for Connections in which a given
protocol is the "top" of the Protocol Stack. For example, the
mapping of the Send function for TCP applies to Connections in which
the application directly sends over TCP.

Each protocol has a notion of Connectedness. Possible values for Connectedness are:

  *Connectionless. Connectionless protocols do not establish
   explicit state between endpoints, and do not perform a handshake
   during Connection establishment.

  *Connected. Connected protocols establish state between endpoints,
   and perform a handshake during Connection establishment. The
   handshake may be 0-RTT to send data or resume a session, but
   bidirectional traffic is required to confirm connectedness.

  *Multiplexing Connected. Multiplexing Connected protocols share
   properties with Connected protocols, but also explictly support
   opening multiple application-level flows. This means that they
   can support cloning new Connection objects without a new explicit
   handshake.

Protocols also define a notion of Data Unit. Possible values for
Data Unit are:

  *Byte-stream. Byte-stream protocols do not define any Message
   boundaries of their own apart from the end of a stream in each
   direction.

  *Datagram. Datagram protocols define Message boundaries at the
   same level of transmission, such that only complete (not partial)
   Messages are supported.

  *Message. Message protocols support Message boundaries that can be
   sent and received either as complete or partial Messages. Maximum
   Message lengths can be defined, and Messages can be partially
   reliable.

Below, terms in capitals with a dot (e.g., "CONNECT.SCTP") refer to
the primitives with the same name in section 4 of [RFC8303]. For
further implementation details, the description of these primitives
in [RFC8303] points to section 3 of [RFC8303] and section 3 of
[RFC8304], which refers back to the relevant specifications for each
protocol. This back-tracking method applies to all elements of
[RFC8923] (see appendix D of [I-D.ietf-taps-interface]): they are
listed in appendix A of [RFC8923] with an implementation hint in the
same style, pointing back to section 4 of [RFC8303].

This document defines the API mappings for protocols defined in
[RFC8923]. Other protocol mappings can be provided as separate
documents, following the mapping template Appendix A.

10.1. TCP

Connectedness: Connected

Data Unit: Byte-stream

API mappings for TCP are as follows:

Connection Object: TCP connections between two hosts map directly to Connection objects.

Initiate: CONNECT.TCP. Calling Initiate on a TCP Connection causes it to reserve a local port, and send a SYN to the Remote Endpoint.

InitiateWithSend: CONNECT.TCP with parameter user message. Early safely replayable data is sent on a TCP Connection in the SYN, as TCP Fast Open data.

Ready: A TCP Connection is ready once the three-way handshake is complete.

InitiateError: Failure of CONNECT.TCP. TCP can throw various errors during connection setup. Specifically, it is important to handle a RST being sent by the peer during the handshake.

ConnectionError: Once established, TCP throws errors whenever the connection is disconnected, such as due to receiving a RST from the peer.

Listen: LISTEN.TCP. Calling Listen for TCP binds a local port and prepares it to receive inbound SYN packets from peers.

ConnectionReceived: TCP Listeners will deliver new connections once they have replied to an inbound SYN with a SYN-ACK.

Clone: Calling Clone on a TCP Connection creates a new Connection with equivalent parameters. These Connections, and Connections generated via later calls to Clone on an Establied Connection, form a Connection Group. To realize entanglement for these Connections, with the exception of Connection Priority, changing a Connection Property on one of them must affect the Connection Properties of the others too. No guarantees of honoring the Connection Property Connection Priority are given, and thus it is safe for an implementation of a transport system to ignore this property. When it is reasonable to assume that Connections traverse the same path (e.g., when they share the same encapsulation), support for it can also experimentally be implemented using a congestion control coupling mechanism (see for example [TCP-COUPLING] or [RFC3124]).

Send:
>   SEND.TCP. TCP does not on its own preserve Message
>   boundaries. Calling Send on a TCP connection lays out the bytes
>   on the TCP send stream without any other delineation. Any Message
>   marked as Final will cause TCP to send a FIN once the Message has
>   been completely written, by calling CLOSE.TCP immediately upon
>   successful termination of SEND.TCP. Note that transmitting a
>   Message marked as Final should not cause the Closed event to be
>   delivered to the application, as it will still be possible to
>   receive data until the peer closes or aborts the TCP connection.

Receive: With RECEIVE.TCP, TCP delivers a stream of bytes without
>   any Message delineation. All data delivered in the Received or
>   ReceivedPartial event will be part of a single stream-wide
>   Message that is marked Final (unless a Message Framer is used).
>   EndOfMessage will be delivered when the TCP Connection has
>   received a FIN (CLOSE-EVENT.TCP) from the peer. Note that
>   reception of a FIN should not cause the Closed event to be
>   delivered to the application, as it will still be possible for
>   the application to send data.

Close: Calling Close on a TCP Connection indicates that the
>   Connection should be gracefully closed (CLOSE.TCP) by sending a
>   FIN to the peer. It will then still be possible to receive data
>   until the peer closes or aborts the TCP connection. The Closed
>   event will be issued upon reception of a FIN.

Abort: Calling Abort on a TCP Connection indicates that the
>   Connection should be immediately closed by sending a RST to the
>   peer (ABORT.TCP).

10.2. MPTCP

Connectedness: Connected

Data Unit: Byte-stream

the Transport Services API mappings for MPTCP are identical to TCP.
MPTCP adds support for multipath properties, such as "Multipath
Transport" and "Policy for using Multipath Transports".

10.3. UDP

Connectedness: Connectionless

Data Unit: Datagram

API mappings for UDP are as follows:

Connection Object:

UDP connections represent a pair of specific IP addresses and ports on two hosts.

Initiate: CONNECT.UDP. Calling Initiate on a UDP Connection causes it to reserve a local port, but does not generate any traffic.

InitiateWithSend: Early data on a UDP Connection does not have any special meaning. The data is sent whenever the Connection is Ready.

Ready: A UDP Connection is ready once the system has reserved a local port and has a path to send to the Remote Endpoint.

InitiateError: UDP Connections can only generate errors on initiation due to port conflicts on the local system.

ConnectionError: Once in use, UDP throws "soft errors" (ERROR.UDP(-Lite)) upon receiving ICMP notifications indicating failures in the network.

Listen: LISTEN.UDP. Calling Listen for UDP binds a local port and prepares it to receive inbound UDP datagrams from peers.

ConnectionReceived: UDP Listeners will deliver new connections once they have received traffic from a new Remote Endpoint.

Clone: Calling Clone on a UDP Connection creates a new Connection with equivalent parameters. The two Connections are otherwise independent.

Send: SEND.UDP(-Lite). Calling Send on a UDP connection sends the data as the payload of a complete UDP datagram. Marking Messages as Final does not change anything in the datagram's contents. Upon sending a UDP datagram, some relevant fields and flags in the IP header can be controlled: DSCP (SET_DSCP.UDP(-Lite)), DF in IPv4 (SET_DF.UDP(-Lite)) and ECN flag (SET_ECN.UDP(-Lite)).

Receive: RECEIVE.UDP(-Lite). UDP only delivers complete Messages to Received, each of which represents a single datagram received in a UDP packet. Upon receiving a UDP datagram, the ECN flag from the IP header can be obtained (GET_ECN.UDP(-Lite)).

Close: Calling Close on a UDP Connection (ABORT.UDP(-Lite)) releases the local port reservation.

Abort: Calling Abort on a UDP Connection (ABORT.UDP(-Lite)) is identical to calling Close.

10.4. UDP-Lite

Connectedness: Connectionless

Data Unit: Datagram

The Transport Services API mappings for UDP-Lite are identical to
UDP. Properties that require checksum coverage are not supported by
UDP-Lite, such as "Corruption Protection Length", "Full Checksum
Coverage on Sending", "Required Minimum Corruption Protection
Coverage for Receiving", and "Full Checksum Coverage on Receiving".

10.5. UDP Multicast Receive

Connectedness: Connectionless

Data Unit: Datagram

API mappings for Receiving Multicast UDP are as follows:

Connection Object: Established UDP Multicast Receive connections
   represent a pair of specific IP addresses and ports. The
   "unidirectional receive" transport property is required, and the
   Local Endpoint must be configured with a group IP address and a
   port.

Initiate: Calling Initiate on a UDP Multicast Receive Connection
   causes an immediate InitiateError. This is an unsupported
   operation.

InitiateWithSend: Calling InitiateWithSend on a UDP Multicast
   Receive Connection causes an immediate InitiateError. This is an
   unsupported operation.

Ready: A UDP Multicast Receive Connection is ready once the system
   has received traffic for the appropriate group and port.

InitiateError: UDP Multicast Receive Connections generate an
   InitiateError if Initiate is called.

ConnectionError: Once in use, UDP throws "soft errors" (ERROR.UDP(-
   Lite)) upon receiving ICMP notifications indicating failures in
   the network.

Listen: LISTEN.UDP. Calling Listen for UDP Multicast Receive binds
   a local port, prepares it to receive inbound UDP datagrams from
   peers, and issues a multicast host join. If a Remote Endpoint
   with an address is supplied, the join is Source-specific
   Multicast, and the path selection is based on the route to the
   Remote Endpoint. If a Remote Endpoint is not supplied, the join

is Any-source Multicast, and the path selection is based on the
outbound route to the group supplied in the Local Endpoint.

There are cases where it is required to open multiple connections
for the same address(es). For example, one Connection might be
opened for a multicast group to for a multicast control bus, and
another application later opens a separate Connection to the same
group to send signals to and/or receive signals from the common bus.
In such cases, the Transport Services system needs to explicitly
enable re-use of the same set of addresses (equivalent to setting
SO_REUSEADDR in the socket API).

ConnectionReceived: UDP Multicast Receive Listeners will deliver
new connections once they have received traffic from a new Remote
Endpoint.

Clone: Calling Clone on a UDP Multicast Receive Connection creates
a new Connection with equivalent parameters. The two Connections
are otherwise independent.

Send: SEND.UDP(-Lite). Calling Send on a UDP Multicast Receive
connection causes an immediate SendError. This is an unsupported
operation.

Receive: RECEIVE.UDP(-Lite). The Receive operation in a UDP
Multicast Receive connection only delivers complete Messages to
Received, each of which represents a single datagram received in
a UDP packet. Upon receiving a UDP datagram, the ECN flag from
the IP header can be obtained (GET_ECN.UDP(-Lite)).

Close: Calling Close on a UDP Multicast Receive Connection
(ABORT.UDP(-Lite)) releases the local port reservation and leaves
the group.

Abort: Calling Abort on a UDP Multicast Receive Connection
(ABORT.UDP(-Lite)) is identical to calling Close.

10.6. SCTP

Connectedness: Connected

Data Unit: Message

API mappings for SCTP are as follows:

Connection Object: Connection objects can be mapped to an SCTP
association or a stream in an SCTP association. Mapping
Connection objects to SCTP streams is called "stream mapping" and
has additional requirements as follows. The following explanation
assumes a client-server communication model.

Stream mapping requires an association to already be in place between the client and the server, and it requires the server to understand that a new incoming stream should be represented as a new Connection Object by the Transport Services system. A new SCTP stream is created by sending an SCTP message with a new stream id. Thus, to implement stream mapping, the Transport Services API MUST provide a newly created Connection Object to the application upon the reception of such a message. The necessary semantics to implement a Transport Services system Close and Abort primitives are provided by the stream reconfiguration (reset) procedure described in [RFC6525]. This also allows to re-use a stream id after resetting ("closing") the stream. To implement this functionality, SCTP stream reconfiguration [RFC6525] MUST be supported by both the client and the server side.

To avoid head-of-line blocking, stream mapping SHOULD only be implemented when both sides support message interleaving [RFC8260]. This allows a sender to schedule transmissions between multiple streams without risking that transmission of a large message on one stream might block transmissions on other streams for a long time.

To avoid conflicts between stream ids, the following procedure is recommended: the first Connection, for which the SCTP association has been created, MUST always use stream id zero. All additional Connections are assigned to unused stream ids in growing order. To avoid a conflict when both endpoints map new Connections simultaneously, the peer which initiated association MUST use even stream ids whereas the remote side MUST map its Connections to odd stream ids. Both sides maintain a status map of the assigned stream ids. Generally, new streams SHOULD consume the lowest available (even or odd, depending on the side) stream id; this rule is relevant when lower ids become available because Connection objects associated with the streams are closed.

SCTP stream mapping as described here has been implemented in a research prototype; a desription of this implementation is given in [NEAT-flow-mapping].

Initiate: If this is the only Connection object that is assigned to the SCTP Association or stream mapping is not used, CONNECT.SCTP is called. Else, unless the Selection Property activeReadBeforeSend is Preferred or Required, a new stream is used: if there are enough streams available, Initiate is a local operation that assigns a new stream id to the Connection object. The number of streams is negotiated as a parameter of the prior CONNECT.SCTP call, and it represents a trade-off between local resource usage and the number of Connection objects that can be mapped without requiring a reconfiguration signal. When running out of streams, ADD_STREAM.SCTP must be called.

InitiateWithSend:
                   If this is the only Connection object that is
     assigned to the SCTP association or stream mapping is not used,
     CONNECT.SCTP is called with the "user message" parameter. Else, a
     new stream is used (see Initiate for how to handle running out of
     streams), and this just sends the first message on a new stream.

Ready: Initiate or InitiateWithSend returns without an error, i.e.
     SCTP's four-way handshake has completed. If an association with
     the peer already exists, stream mapping is used and enough
     streams are available, a Connection Object instantly becomes
     Ready after calling Initiate or InitiateWithSend.

InitiateError: Failure of CONNECT.SCTP.

ConnectionError: TIMEOUT.SCTP or ABORT-EVENT.SCTP.

Listen: LISTEN.SCTP. If an association with the peer already exists
     and stream mapping is used, Listen just expects to receive a new
     message with a new stream id (chosen in accordance with the
     stream id assignment procedure described above).

ConnectionReceived: LISTEN.SCTP returns without an error (a result
     of successful CONNECT.SCTP from the peer), or, in case of stream
     mapping, the first message has arrived on a new stream (in this
     case, Receive is also invoked).

Clone: Calling Clone on an SCTP association creates a new
     Connection object and assigns it a new stream id in accordance
     with the stream id assignment procedure described above. If there
     are not enough streams available, ADD_STREAM.SCTP must be called.

Priority (Connection): When this value is changed, or a Message
     with Message Property Priority is sent, and there are multiple
     Connection objects assigned to the same SCTP association,
     CONFIGURE_STREAM_SCHEDULER.SCTP is called to adjust the
     priorities of streams in the SCTP association.

Send: SEND.SCTP. Message Properties such as Lifetime and Ordered
     map to parameters of this primitive.

Receive: RECEIVE.SCTP. The "partial flag" of RECEIVE.SCTP invokes a
     ReceivedPartial event.

Close: If this is the only Connection object that is assigned to the
SCTP association, CLOSE.SCTP is called, and the Closed event will be
delivered to the application upon the ensuing CLOSE-EVENT.SCTP.
Else, the Connection object is one out of several Connection objects
that are assigned to the same SCTP assocation, and RESET_STREAM.SCTP
must be called, which informs the peer that the stream will no

longer be used for mapping and can be used by future Initiate,
InitiateWithSend or Listen calls. At the peer, the event
RESET_STREAM-EVENT.SCTP will fire, which the peer must answer by
issuing RESET_STREAM.SCTP too. The resulting local RESET_STREAM-
EVENT.SCTP informs the Transport Services system that the stream id
can now be re-used by the next Initiate, InitiateWithSend or Listen
calls, and invokes a Closed event towards the application.

Abort: If this is the only Connection object that is assigned to the
SCTP association, ABORT.SCTP is called. Else, the Connection object
is one out of several Connection objects that are assigned to the
same SCTP assocation, and shutdown proceeds as described under
Close.

11. IANA Considerations

RFC-EDITOR: Please remove this section before publication.

This document has no actions for IANA.

12. Security Considerations

[I-D.ietf-taps-arch] outlines general security consideration and
requirements for any system that implements the Transport Services
archtecture. [I-D.ietf-taps-interface] provides further discussion
on security and privacy implications of the Transport Services API.
This document provides additional guidance on implementation
specifics for the Transport Services API and as such the security
considerations in both of these documents apply. The next two
subsections discuss further considerations that are specific to
mechanisms specified in this document.

12.1. Considerations for Candidate Gathering

Implementations should avoid downgrade attacks that allow network
interference to cause the implementation to select less secure, or
entirely insecure, combinations of paths and protocols.

12.2. Considerations for Candidate Racing

See Section 5.3 for security considerations around racing with 0-RTT
data.

An attacker that knows a particular device is racing several options
during connection establishment may be able to block packets for the
first connection attempt, thus inducing the device to fall back to a
secondary attempt. This is a problem if the secondary attempts have
worse security properties that enable further attacks.
Implementations should ensure that all options have equivalent
security properties to avoid incentivizing attacks.

Since results from the network can determine how a connection attempt tree is built, such as when DNS returns a list of resolved endpoints, it is possible for the network to cause an implementation to consume significant on-device resources. Implementations should limit the maximum amount of state allowed for any given node, including the number of child nodes, especially when the state is based on results from the network.

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Appendix A. API Mapping Template

Any protocol mapping for the Transport Services API should follow a
common template.

Connectedness: (Connectionless/Connected/Multiplexing Connected)

Data Unit: (Byte-stream/Datagram/Message)

Connection Object:

Initiate:

InitiateWithSend:

Ready:

InitiateError:

ConnectionError:

Listen:

ConnectionReceived:

Clone:

Send:

Receive:

Close:

Abort:

Appendix B. Additional Properties

This appendix discusses implementation considerations for additional
parameters and properties that could be used to enhance transport
protocol and/or path selection, or the transmission of messages
given a Protocol Stack that implements them. These are not part of
the interface, and may be removed from the final document, but are
presented here to support discussion within the TAPS working group
as to whether they should be added to a future revision of the base
specification.

B.1. Properties Affecting Sorting of Branches

In addition to the Protocol and Path Selection Properties discussed
in Section 4.1.3, the following properties under discussion can
influence branch sorting:

  *Bounds on Send or Receive Rate: If the application indicates a
   bound on the expected Send or Receive bitrate, an implementation
   may prefer a path that can likely provide the desired bandwidth,
   based on cached maximum throughput, see Section 9.2. The
   application may know the Send or Receive Bitrate from metadata in
   adaptive HTTP streaming, such as MPEG-DASH.

  *Cost Preferences: If the application indicates a preference to
   avoid expensive paths, and some paths are associated with a
   monetary cost, an implementation should decrease the ranking of
   such paths. If the application indicates that it prohibits using
   expensive paths, paths that are associated with a cost should be
   purged from the decision tree.

Appendix C. Reasons for errors

The Transport Services API [I-D.ietf-taps-interface] allows for the
several generic error types to specify a more detailed reason as to

why an error occurred. This appendix lists some of the possible
reasons.

  *InvalidConfiguration: The transport properties and endpoints
   provided by the application are either contradictory or
   incomplete. Examples include the lack of a Remote Endpoint on an
   active open or using a multicast group address while not
   requesting a unidirectional receive.

  *NoCandidates: The configuration is valid, but none of the
   available transport protocols can satisfy the transport
   properties provided by the application.

  *ResolutionFailed: The remote or local specifier provided by the
   application can not be resolved.

  *EstablishmentFailed: The Transport Services system was unable to
   establish a transport-layer connection to the Remote Endpoint
   specified by the application.

  *PolicyProhibited: The system policy prevents the transport system
   from performing the action requested by the application.

  *NotCloneable: The protocol stack is not capable of being cloned.

  *MessageTooLarge: The message size is too big for the transport
   system to handle.

  *ProtocolFailed: The underlying protocol stack failed.

  *InvalidMessageProperties: The message properties are either
   contradictory to the transport properties or they can not be
   satisfied by the transport system.

  *DeframingFailed: The data that was received by the underlying
   protocol stack could not be deframed.

  *ConnectionAborted: The connection was aborted by the peer.

  *Timeout: Delivery of a message was not possible after a timeout.

Appendix D. Existing Implementations

This appendix gives an overview of existing implementations, at the time of writing, of transport systems that are (to some degree) in line with this document.

* *Apple's Network.framework:

    - Network.framework is a transport-level API built for C, Objective-C, and Swift. It a connect-by-name API that supports transport security protocols. It provides userspace implementations of TCP, UDP, TLS, DTLS, proxy protocols, and allows extension via custom framers.

    - Documentation: https://developer.apple.com/documentation/network

* *NEAT and NEATPy:

    - NEAT is the output of the European H2020 research project "NEAT"; it is a user-space library for protocol-independent communication on top of TCP, UDP and SCTP, with many more features such as a policy manager.

    - Code: https://github.com/NEAT-project/neat

    - NEAT project: https://www.neat-project.org

    - NEATPy is a Python shim over NEAT which updates the NEAT API to be in line with version 6 of the Transport Services API draft.

    - Code: https://github.com/theagilepadawan/NEATPy

* *PyTAPS:

    - A TAPS implementation based on Python asyncio, offering protocol-independent communication to applications on top of TCP, UDP and TLS, with support for multicast.

    - Code: https://github.com/fg-inet/python-asyncio-taps

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