

Network Working Group
Internet-Draft
Intended status: Informational
Expires: August 31, 2015

G. Fairhurst, Ed.
University of Aberdeen
B. Trammell, Ed.
M. Kuehlewind, Ed.
ETH Zurich
February 27, 2015

Services provided by IETF transport protocols and congestion control
mechanisms
draft-ietf-taps-transport-03

Abstract

This document describes services provided by existing IETF protocols and congestion control mechanisms. It is designed to help application and network stack programmers and to inform the work of the IETF TAPS Working Group.

Status of This Memo

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

Internet-Drafts are working documents of the Internet Engineering Task Force (IETF). Note that other groups may also distribute working documents as Internet-Drafts. The list of current Internet-Drafts is at <http://datatracker.ietf.org/drafts/current/>.

Internet-Drafts are draft documents valid for a maximum of six months and may be updated, replaced, or obsoleted by other documents at any time. It is inappropriate to use Internet-Drafts as reference material or to cite them other than as "work in progress."

This Internet-Draft will expire on August 31, 2015.

Copyright Notice

Copyright (c) 2015 IETF Trust and the persons identified as the document authors. All rights reserved.

This document is subject to BCP 78 and the IETF Trust's Legal Provisions Relating to IETF Documents (<http://trustee.ietf.org/license-info>) in effect on the date of publication of this document. Please review these documents carefully, as they describe your rights and restrictions with respect to this document. Code Components extracted from this document must include Simplified BSD License text as described in Section 4.e of

the Trust Legal Provisions and are provided without warranty as described in the Simplified BSD License.

1. Introduction

Most Internet applications make use of the Transport Services provided by TCP (a reliable, in-order stream protocol) or UDP (an unreliable datagram protocol). We use the term "Transport Service" to mean the end-to-end service provided to an application by the transport layer. That service can only be provided correctly if information about the intended usage is supplied from the application. The application may determine this information at design time, compile time, or run time, and may include guidance on whether a feature is required, a preference by the application, or something in between. Examples of features of Transport Services are reliable delivery, ordered delivery, content privacy to in-path devices, integrity protection, and minimal latency.

The IETF has defined a wide variety of transport protocols beyond TCP and UDP, including TCP, SCTP, DCCP, MP-TCP, and UDP-Lite. Transport services may be provided directly by these transport protocols, or layered on top of them using protocols such as WebSockets (which runs over TCP) or RTP (over TCP or UDP). Services built on top of UDP or UDP-Lite typically also need to specify additional mechanisms, including a congestion control mechanism (such as a windowed congestion control, TFRC or LEDBAT congestion control mechanism). This extends the set of available Transport Services beyond those provided to applications by TCP and UDP.

Transport protocols can also be differentiated by the features of the services they provide: for instance, SCTP offers a message-based service that does not suffer head-of-line blocking when used with multiple stream, because it can accept blocks of data out of order, UDP-Lite provides partial integrity protection, and LEDBAT can provide low-priority "scavenger" communication.

2. Terminology

The following terms are defined throughout this document, and in subsequent documents produced by TAPS describing the composition and decomposition of transport services.

[NOTE: The terminology below was presented at the TAPS WG meeting in Honolulu. While the factoring of the terminology seems uncontroversial, there may be some entities which still require names (e.g. information about the interface between the transport and lower layers which could lead to the availability or unavailability of

certain transport protocol features). Comments are welcome via the TAPS mailing list.]

Transport Service Feature: a specific end-to-end feature that a transport service provides to its clients. Examples include confidentiality, reliable delivery, ordered delivery, message-versus-stream orientation, etc.

Transport Service: a set of transport service features, without an association to any given framing protocol, which provides a complete service to an application.

Transport Protocol: an implementation that provides one or more different transport services using a specific framing and header format on the wire.

Transport Protocol Component: an implementation of a transport service feature within a protocol.

Transport Service Instance: an arrangement of transport protocols with a selected set of features and configuration parameters that implements a single transport service, e.g. a protocol stack (RTP over UDP).

Application: an entity that uses the transport layer for end-to-end delivery data across the network (this may also be an upper layer protocol or tunnel encapsulation).

3. Existing Transport Protocols

This section provides a list of known IETF transport protocol and transport protocol frameworks.

[EDITOR'S NOTE: Contributions to the subsections below are welcome]

3.1. Transport Control Protocol (TCP)

TCP is an IETF standards track transport protocol. [RFC0793] introduces TCP as follows: "The Transmission Control Protocol (TCP) is intended for use as a highly reliable host-to-host protocol between hosts in packet-switched computer communication networks, and in interconnected systems of such networks." Since its introduction, TCP has become the default connection-oriented, stream-based transport protocol in the Internet. It is widely implemented by endpoints and widely used by common applications.

3.1.1. Protocol Description

TCP is a connection-oriented protocol, providing a three way handshake to allow a client and server to set up a connection, and mechanisms for orderly completion and immediate teardown of a connection. TCP is defined by a family of RFCs [RFC4614].

TCP provides multiplexing to multiple sockets on each host using port numbers. An active TCP session is identified by its four-tuple of local and remote IP addresses and local port and remote port numbers. The destination port during connection setup has a different role as it is often used to indicate the requested service.

TCP partitions a continuous stream of bytes into segments, sized to fit in IP packets. ICMP-based PathMTU discovery [RFC1191][RFC1981] as well as Packetization Layer Path MTU Discovery (PMTUD) [RFC4821] are supported.

Each byte in the stream is identified by a sequence number. The sequence number is used to order segments on receipt, to identify segments in acknowledgments, and to detect unacknowledged segments for retransmission. This is the basis of TCP's reliable, ordered delivery of data in a stream. TCP Selective Acknowledgment [RFC2018] extends this mechanism by making it possible to identify missing segments more precisely, reducing spurious retransmission.

Receiver flow control is provided by a sliding window: limiting the amount of unacknowledged data that can be outstanding at a given time. The window scale option [RFC7323] allows a receiver to use windows greater than 64KB.

All TCP senders provide Congestion Control: This uses a separate window, where each time congestion is detected, this congestion window is reduced. A receiver detects congestion using one of three mechanisms: A retransmission timer, detection of loss (interpreted as a congestion signal), or Explicit Congestion Notification (ECN) [RFC3168] to provide early signaling (see [I-D.ietf-aqm-ecn-benefits])

A TCP protocol instance can be extended [RFC4614] and tuned. Some features are sender-side only, requiring no negotiation with the receiver; some are receiver-side only, some are explicitly negotiated during connection setup.

By default, TCP segment partitioning uses Nagle's algorithm [RFC0896] to buffer data at the sender into large segments, potentially incurring sender-side buffering delay; this algorithm can be disabled

by the sender to transmit more immediately, e.g. to enable smoother interactive sessions.

[EDITOR'S NOTE: add URGENT and PUSH flag (note [RFC6093] says SHOULD NOT use due to the range of TCP implementations that process TCP urgent indications differently.)]

A checksum provides an Integrity Check and is mandatory across the entire packet. The TCP checksum does not support partial corruption protection as in DCCP/UDP-Lite). This check protects from misdelivery of data corrupted data, but is relatively weak, and applications that require end to end integrity of data are recommended to include a stronger integrity check of their payload data.

A TCP service is unicast.

3.1.2. Interface description

A User/TCP Interface is defined in [RFC0793] providing six user commands: Open, Send, Receive, Close, Status. This interface does not describe configuration of TCP options or parameters beside use of the PUSH and URGENT flags.

In API implementations derived from the BSD Sockets API, TCP sockets are created using the "SOCK_STREAM" socket type.

The features used by a protocol instance may be set and tuned via this API.

(more on the API goes here)

3.1.3. Transport Protocol Components

The transport protocol components provided by TCP are:

- o unicast
- o connection setup with feature negotiation and application-to-port mapping
- o port multiplexing
- o reliable delivery
- o ordered delivery for each byte stream
- o error detection (checksum)

- o segmentation
- o stream-oriented delivery in a single stream
- o data bundling (Nagle's algorithm)
- o flow control
- o congestion control

[EDITOR'S NOTE: discussion of how to map this to features and TAPS: what does the higher layer need to decide? what can the transport layer decide based on global settings? what must the transport layer decide based on network characteristics?]

3.2. Multipath TCP (MP-TCP)

[EDITOR'S NOTE: a few sentences describing Multipath TCP [RFC6824] go here. Note that this adds transport-layer multihoming to the components TCP provides]

3.3. Stream Control Transmission Protocol (SCTP)

SCTP is a message oriented standards track transport protocol and the base protocol is specified in [RFC4960]. It supports multi-homing to handle path failures. An SCTP association has multiple unidirectional streams in each direction and provides in-sequence delivery of user messages only within each stream. This allows to minimize head of line blocking. SCTP is extensible and the currently defined extensions include mechanisms for dynamic re-configurations of streams [RFC6525] and IP-addresses [RFC5061]. Furthermore, the extension specified in [RFC3758] introduces the concept of partial reliability for user messages.

SCTP was originally developed for transporting telephony signalling messages and is deployed in telephony signalling networks, especially in mobile telephony networks. Additionally, it is used in the WebRTC framework for data channels and is therefore deployed in all WEB-browsers supporting WebRTC.

[EDITOR'S NOTE: Michael Tuexen and Karen Nielsen signed up as contributors for these sections.]

3.3.1. Protocol Description

SCTP is a connection oriented protocol using a four way handshake to establish an SCTP association and a three way message exchange to

gracefully shut it down. It uses the same port number concept as DCCP, TCP, UDP, and UDP-Lite do and only supports unicast.

SCTP uses the 32-bit CRC32c for protecting SCTP packets against bit errors. This is stronger than the 16-bit checksums used by TCP or UDP. However, a partial checksum coverage as provided by DCCP or UDP-Lite is not supported.

SCTP has been designed with extensibility in mind. Each SCTP packet starts with a single common header containing the port numbers, a verification tag and the CRC32c checksum. This common header is followed by a sequence of chunks. Each chunk consists of a type field, flags, a length field and a value. [RFC4960] defines how a receiver processes chunks with an unknown chunk type. The support of extensions can be negotiated during the SCTP handshake.

SCTP provides a message-oriented service. Multiple small user messages can be bundled into a single SCTP packet to improve the efficiency. User messages which would result in IP packets larger than the MTU will be fragmented at the sender side and reassembled at the receiver side. There is no protocol limit on the user message size. [RFC4821] defines a method to perform packetization layer path MTU discovery with probe packets using the padding chunks defined in [RFC4820].

[RFC4960] specifies a TCP friendly congestion control to protect the network against overload. SCTP also uses a sliding window flow control to protect receivers against overflow.

Each SCTP association has between 1 and 65536 uni-directional streams in each direction. The number of streams can be different in each direction. Every user-message is sent on a particular stream. User messages can be sent ordered or un-ordered upon request by the upper layer. Only all ordered messages sent on the same stream are delivered at the receiver in the same order as sent by the sender. For user messages not requiring fragmentation, this minimises head of line blocking. The base protocol defined in [RFC4960] doesn't allow interleaving of user-messages, which results in sending a large message on one stream can block the sending of user messages on other streams. [I-D.ietf-tsvwg-sctp-ndata] overcomes this limitation and also allows to specify a scheduler for the sender side streams selection. The stream re-configuration extension defined in [RFC6525] allows to reset streams during the lifetime of an association and to increase the number of streams, if the number of streams negotiated in the SCTP handshake is not sufficient.

According to [RFC4960], each user message sent is either delivered to the receiver or, in case of excessive retransmissions, the

association is terminated in a non-graceful way, similar to the TCP behaviour. In addition to this reliable transfer, the partial reliability extension defined in [RFC3758] allows the sender to abandon user messages. The application can specify the policy for abandoning user messages. Examples for these policies include:

- o Limiting the time a user message is dealt with by the sender.
- o Limiting the number of retransmissions for each fragment of a user message.
- o Abandoning messages of lower priority in case of a send buffer shortage.

SCTP supports multi-homing. Each SCTP end-point uses a list of IP-addresses and a single port number. These addresses can be any mixture of IPv4 and IPv6 addresses. These addresses are negotiated during the handshake and the address re-configuration extension specified in [RFC5061] can be used to change these addresses during the lifetime of an SCTP association. This allows for transport layer mobility. Multiple addresses are used for improved resilience. If a remote address becomes unreachable, the traffic is switched over to a reachable one, if one exists. Each SCTP end-point supervises continuously the reachability of all peer addresses using a heartbeat mechanism.

For securing user messages, the use of TLS over SCTP has been specified in [RFC3436]. However, this solution does not support all services provided by SCTP (for example un-ordered delivery or partial reliability), and therefore the use of DTLS over SCTP has been specified in [RFC6083] to overcome these limitations. When using DTLS over SCTP, the application can use almost all services provided by SCTP.

For legacy NAT traversal, [RFC6951] defines the UDP encapsulation of SCTP-packets. Alternatively, SCTP packets can be encapsulated in DTLS packets as specified in [I-D.ietf-tsvwg-sctp-dtls-encaps]. The latter encapsulation is used with in the WebRTC context.

Having a well defined API is also a feature provided by SCTP as described in the next subsection.

3.3.2. Interface Description

[RFC4960] defines an abstract API for the base protocol. An extension to the BSD Sockets API is defined in [RFC6458] and covers:

- o the base protocol defined in [RFC4960].

- o the SCTP Partial Reliability extension defined in [RFC3758].
- o the SCTP Authentication extension defined in [RFC4895].
- o the SCTP Dynamic Address Reconfiguration extension defined in [RFC5061].

For the following SCTP protocol extensions the BSD Sockets API extension is defined in the document specifying the protocol extensions:

- o the SCTP SACK-IMMEDIATELY extension defined in [RFC7053].
- o the SCTP Stream Reconfiguration extension defined in [RFC6525].
- o the UDP Encapsulation of SCTP packets extension defined in [RFC6951].
- o the additional PR-SCTP policies defined in [I-D.ietf-tsvwg-sctp-prpolicies].

Future documents describing SCTP protocol extensions are expected to describe the corresponding BSD Sockets API extension in a "Socket API Considerations" section.

The SCTP socket API supports two kinds of sockets:

- o one-to-one style sockets (by using the socket type "SOCK_STREAM").
- o one-to-many style socket (by using the socket type "SOCK_SEQPACKET").

One-to-one style sockets are similar to TCP sockets, there is a 1:1 relationship between the sockets and the SCTP associations (except for listening sockets). One-to-many style SCTP sockets are similar to unconnected UDP sockets as there is a 1:n relationship between the sockets and the SCTP associations.

The SCTP stack can provide information to the applications about state changes of the individual paths and the association whenever they occur. These events are delivered similar to user messages but are specifically marked as notifications.

A couple of new functions have been introduced to support the use of multiple local and remote addresses. Additional SCTP-specific send and receive calls have been defined to allow dealing with the SCTP specific information without using ancillary data in the form of additional cmsgs, which are also defined. These functions provide

support for detecting partial delivery of user messages and notifications.

The SCTP socket API allows a fine-grained control of the protocol behaviour through an extensive set of socket options.

The SCTP kernel implementations of FreeBSD, Linux and Solaris follow mostly the specified extension to the BSD Sockets API for the base protocol and the corresponding supported protocol extensions.

3.3.3. Transport Protocol Components

The transport protocol components provided by SCTP are:

- o unicast
- o connection setup with feature negotiation and application-to-port mapping
- o port multiplexing
- o reliable or partially reliable delivery
- o ordered and unordered delivery within a stream
- o support for multiple prioritised streams
- o flow control (slow receiver function)
- o message-oriented delivery
- o congestion control
- o application PDU bundling
- o application PDU fragmentation and reassembly
- o integrity check
- o transport layer multihoming for resilience
- o transport layer mobility

[EDITOR'S NOTE: update this list.]

3.4. User Datagram Protocol (UDP)

The User Datagram Protocol (UDP) [RFC0768] [RFC2460] is an IETF standards track transport protocol. It provides a uni-directional, datagram protocol which preserves message boundaries. It provides none of the following transport features: error correction, congestion control, or flow control. It can be used to send broadcast datagrams (IPv4) or multicast datagrams (IPv4 and IPv6), in addition to unicast (and anycast) datagrams. IETF guidance on the use of UDP is provided in [RFC5405]. UDP is widely implemented and widely used by common applications, especially DNS.

[EDITOR'S NOTE: Kevin Fall signed up as a contributor for this section.]

3.4.1. Protocol Description

UDP is a connection-less protocol which maintains message boundaries, with no connection setup or feature negotiation. The protocol uses independent messages, ordinarily called datagrams. The lack of error control and flow control implies messages may be damaged, re-ordered, lost, or duplicated in transit. A receiving application unable to run sufficiently fast or frequently may miss messages. The lack of congestion handling implies UDP traffic may cause the loss of messages from other protocols (e.g., TCP) when sharing the same network paths. UDP traffic can also cause the loss of other UDP traffic in the same or other flows for the same reasons.

Messages with bit errors are ordinarily detected by an invalid end-to-end checksum and are discarded before being delivered to an application. There are some exceptions to this general rule, however. UDP-Lite (see [RFC3828], and below) provides the ability for portions of the message contents to be exempt from checksum coverage. It is also possible to create UDP datagrams with no checksum, and while this is generally discouraged [RFC1122] [RFC5405], certain special cases permit its use [RFC6935]. The checksum support considerations for omitting the checksum are defined in [RFC6936]. Note that due to the relatively weak form of checksum used by UDP, applications that require end to end integrity of data are recommended to include a stronger integrity check of their payload data.

On transmission, UDP encapsulates each datagram into an IP packet, which may in turn be fragmented by IP. Applications concerned with fragmentation or that have other requirements such as receiver flow control, congestion control, PathMTU discovery/PLPMTUD, support for ECN, etc need to be provided by protocols other than UDP [RFC5405].

3.4.2. Interface Description

[RFC0768] describes basic requirements for an API for UDP. Guidance on use of common APIs is provided in [RFC5405].

A UDP endpoint consists of a tuple of (IP address, port number). Demultiplexing using multiple abstract endpoints (sockets) on the same IP address are supported. The same socket may be used by a single server to interact with multiple clients (note: this behavior differs from TCP, which uses a pair of tuples to identify a connection). Multiple server instances (processes) binding the same socket can cooperate to service multiple clients- the socket implementation arranges to not duplicate the same received unicast message to multiple server processes.

Many operating systems also allow a UDP socket to be "connected", i.e., to bind a UDP socket to a specific (remote) UDP endpoint. Unlike TCP's connect primitive, for UDP, this is only a local operation that serves to simplify the local send/receive functions and to filter the traffic for the specified addresses and ports [RFC5405].

3.4.3. Transport Protocol Components

The transport protocol components provided by UDP are:

- o unidirectional
- o port multiplexing
- o 2-tuple endpoints
- o IPv4 broadcast, multicast and anycast
- o IPv6 multicast and anycast
- o IPv6 jumbograms
- o message-oriented delivery
- o error detection (checksum)
- o checksum optional

3.5. Lightweight User Datagram Protocol (UDP-Lite)

The Lightweight User Datagram Protocol (UDP-Lite) [RFC3828] is an IETF standards track transport protocol. UDP-Lite provides a bidirectional set of logical unicast or multicast message streams over a datagram protocol. IETF guidance on the use of UDP-Lite is provided in [RFC5405].

[EDITOR'S NOTE: Gorrry Fairhurst signed up as a contributor for this section.]

3.5.1. Protocol Description

UDP-Lite is a connection-less datagram protocol, with no connection setup or feature negotiation. The protocol use messages, rather than a byte-stream. Each stream of messages is independently managed, therefore retransmission does not hold back data sent using other logical streams.

It provides multiplexing to multiple sockets on each host using port numbers. An active UDP-Lite session is identified by its four-tuple of local and remote IP addresses and local port and remote port numbers.

UDP-Lite fragments packets into IP packets, constrained by the maximum size of IP packet.

UDP-Lite changes the semantics of the UDP "payload length" field to that of a "checksum coverage length" field. Otherwise, UDP-Lite is semantically identical to UDP. Applications using UDP-Lite therefore can not make assumptions regarding the correctness of the data received in the insensitive part of the UDP-Lite payload.

As for UDP, mechanisms for receiver flow control, congestion control, PMTU or PLPMTU discovery, support for ECN, etc need to be provided by upper layer protocols [RFC5405].

Examples of use include a class of applications that can derive benefit from having partially-damaged payloads delivered, rather than discarded. One use is to support error tolerate payload corruption when used over paths that include error-prone links, another application is when header integrity checks are required, but payload integrity is provided by some other mechanism (e.g. [RFC6936]).

A UDP-Lite service may support IPv4 broadcast, multicast, anycast and unicast.

3.5.2. Interface Description

There is no current API specified in the RFC Series, but guidance on use of common APIs is provided in [RFC5405].

The interface of UDP-Lite differs from that of UDP by the addition of a single (socket) option that communicates a checksum coverage length value: at the sender, this specifies the intended checksum coverage, with the remaining unprotected part of the payload called the "error-insensitive part". The checksum coverage may also be made visible to the application via the UDP-Lite MIB module [RFC5097].

3.5.3. Transport Protocol Components

The transport protocol components provided by UDP-Lite are:

- o unicast
- o IPv4 broadcast, multicast and anycast
- o port multiplexing
- o non-reliable, non-ordered delivery
- o message-oriented delivery
- o partial integrity protection

3.6. Datagram Congestion Control Protocol (DCCP)

Datagram Congestion Control Protocol (DCCP) [RFC4340] is an IETF standards track bidirectional transport protocol that provides unicast connections of congestion-controlled unreliable messages.

[EDITOR'S NOTE: Gorrry Fairhurst signed up as a contributor for this section.]

The DCCP Problem Statement describes the goals that DCCP sought to address [RFC4336]. It is suitable for applications that transfer fairly large amounts of data and that can benefit from control over the trade off between timeliness and reliability [RFC4336].

It offers low overhead, and many characteristics common to UDP, but can avoid "Re-inventing the wheel" each time a new multimedia application emerges. Specifically it includes core functions (feature negotiation, path state management, RTT calculation, PMTUD, etc): This allows applications to use a compatible method defining how they send packets and where suitable to choose common algorithms

to manage their functions. Examples of suitable applications include interactive applications, streaming media or on-line games [RFC4336].

3.6.1. Protocol Description

DCCP is a connection-oriented datagram protocol, providing a three way handshake to allow a client and server to set up a connection, and mechanisms for orderly completion and immediate teardown of a connection. The protocol is defined by a family of RFCs.

It provides multiplexing to multiple sockets on each host using port numbers. An active DCCP session is identified by its four-tuple of local and remote IP addresses and local port and remote port numbers. At connection setup, DCCP also exchanges the the service code [RFC5595] mechanism to allow transport instantiations to indicate the service treatment that is expected from the network.

The protocol segments data into messages, typically sized to fit in IP packets, but which may be fragmented providing they are less than the A DCCP interface MAY allow applications to request fragmentation for packets larger than PMTU, but not larger than the maximum packet size allowed by the current congestion control mechanism (CCMPS) [RFC4340].

Each message is identified by a sequence number. The sequence number is used to identify segments in acknowledgments, to detect unacknowledged segments, to measure RTT, etc. The protocol may support ordered or unordered delivery of data, and does not itself provide retransmission. There is a Data Checksum option, which contains a strong CRC, lets endpoints detect application data corruption. It also supports reduced checksum coverage, a partial integrity mechanisms similar to UDP-Lite.

Receiver flow control is supported: limiting the amount of unacknowledged data that can be outstanding at a given time.

A DCCP protocol instance can be extended [RFC4340] and tuned. Some features are sender-side only, requiring no negotiation with the receiver; some are receiver-side only, some are explicitly negotiated during connection setup.

DCCP supports negotiation of the congestion control profile, to provide Plug and Play congestion control mechanisms. examples of specified profiles include [RFC4341] [RFC4342] [RFC5662]. All IETF-defined methods provide Congestion Control.

DCCP use a Connect packet to start a session, and permits half-connections that allow each client to choose features it wishes to

support. Simultaneous open [RFC5596], as in TCP, can enable interoperability in the presence of middleboxes. The Connect packet includes a Service Code field [RFC5595] designed to allow middle boxes and endpoints to identify the characteristics required by a session. A lightweight UDP-based encapsulation (DCCP-UDP) has been defined [RFC6773] that permits DCCP to be used over paths where it is not natively supported. Support in NAPT/NATs is defined in [RFC4340] and [RFC5595].

Upper layer protocols specified on top of DCCP include: DTLS [RFC5595], RTP [RFC5672], ICE/SDP [RFC6773].

A DCCP service is unicast.

A common packet format has allowed tools to evolve that can read and interpret DCCP packets (e.g. Wireshark).

3.6.2. Interface Description

API characteristics include: - Datagram transmission. - Notification of the current maximum packet size. - Send and reception of zero-length payloads. - Set the Slow Receiver flow control at a receiver. - Detect a Slow receiver at the sender.

There is no current API specified in the RFC Series.

3.6.3. Transport Protocol Components

The transport protocol components provided by DCCP are:

- o unicast
- o connection setup with feature negotiation and application-to-port mapping
- o Service Codes
- o port multiplexing
- o non-reliable, ordered delivery
- o flow control (slow receiver function)
- o drop notification
- o timestamps
- o message-oriented delivery

- o partial integrity protection

3.7. Realtime Transport Protocol (RTP)

RTP provides an end-to-end network transport service, suitable for applications transmitting real-time data, such as audio, video or data, over multicast or unicast network services, including TCP, UDP, UDP-Lite, DCCP.

[EDITOR'S NOTE: Varun Singh signed up as contributor for this section.]

3.8. Transport Layer Security (TLS) and Datagram TLS (DTLS) as a pseudo transport

[NOTE: A few words on TLS [RFC5246] and DTLS [RFC6347] here, and how they get used by other protocols to meet security goals as an add-on interlayer above transport.]

3.8.1. Protocol Description

3.8.2. Interface Description

3.8.3. Transport Protocol Components

3.9. Hypertext Transport Protocol (HTTP) as a pseudotransport [RFC3205]

[EDITOR'S NOTE: No identified contributor for this section yet.]

3.9.1. Protocol Description

3.9.2. Interface Description

3.9.3. Transport Protocol Components

3.10. WebSockets

[RFC6455]

[EDITOR'S NOTE: No identified contributor for this section yet.]

3.10.1. Protocol Description

3.10.2. Interface Description

3.10.3. Transport Protocol Components

4. Transport Service Features

[EDITOR'S NOTE: this section will drawn from the candidate features provided by protocol components in the previous section - please discuss on taps@ietf.org list]

4.1. Complete Protocol Feature Matrix

[EDITOR'S NOTE: Dave Thaler has signed up as a contributor for this section. Michael Welzl also has a beginning of a matrix which could be useful here.]

[EDITOR'S NOTE: The below is a strawman proposal below by Gorrry Fairhurst for initial discussion]

The table below summarises protocol mechanisms that have been standardised. It does not make an assessment on whether specific implementations are fully compliant to these specifications.

Mechanism	UDP	UDP-L	DCCP	SCTP	TCP
Unicast	Yes	Yes	Yes	Yes	Yes
Mcast/IPv4Bcast	Yes(2)	Yes	No	No	No
Port Mux	Yes	Yes	Yes	Yes	Yes
Mode	Dgram	Dgram	Dgram	Dgram	Stream
Connected	No	No	Yes	Yes	Yes
Data bundling	No	No	No	Yes	Yes
Feature Nego	No	No	Yes	Yes	Yes
Options	No	No	Support	Support	Support
Data priority	*	*	*	Yes	No
Data bundling	No	No	No	Yes	Yes
Reliability	None	None	None	Select	Full
Ordered deliv	No	No	No	Stream	Yes
Corruption Tol.	No	Support	Support	No	No
Flow Control	No	No	Support	Yes	Yes
PMTU/PLPMTU	(1)	(1)	Yes	Yes	Yes
Cong Control	(1)	(1)	Yes	Yes	Yes
ECN Support	(1)	(1)	Yes	TBD	Yes
NAT support	Limited	Limited	Support	TBD	Support
Security	DTLS	DTLS	DTLS	DTLS	TLS, A0
UDP encaps	N/A	None	Yes	Yes	None
RTP support	Support	Support	Support	?	Support

Note (1): this feature requires support in an upper layer protocol.

Note (2): this feature requires support in an upper layer protocol when used with IPv6.

5. IANA Considerations

This document has no considerations for IANA.

6. Security Considerations

This document surveys existing transport protocols and protocols providing transport-like services. Confidentiality, integrity, and authenticity are among the features provided by those services. This document does not specify any new components or mechanisms for providing these features. Each RFC listed in this document discusses the security considerations of the specification it contains.

7. Contributors

[Editor's Note: turn this into a real contributors section with addresses once we figure out how to trick the toolchain into doing so]

- o Section 3.4 on UDP was contributed by Kevin Fall (kfall@kfall.com)
- o Section 3.3 on SCTP was contributed by Michael Tuexen (tuexen@fh-muenster.de)

8. Acknowledgments

This work is partially supported by the European Commission under grant agreement FP7-ICT-318627 mPlane; support does not imply endorsement.

9. References

9.1. Normative References

[RFC0791] Postel, J., "Internet Protocol", STD 5, RFC 791, September 1981.

9.2. Informative References

[RFC0768] Postel, J., "User Datagram Protocol", STD 6, RFC 768, August 1980.

[RFC0793] Postel, J., "Transmission Control Protocol", STD 7, RFC 793, September 1981.

- [RFC0896] Nagle, J., "Congestion control in IP/TCP internetworks", RFC 896, January 1984.
- [RFC1122] Braden, R., "Requirements for Internet Hosts - Communication Layers", STD 3, RFC 1122, October 1989.
- [RFC1191] Mogul, J. and S. Deering, "Path MTU discovery", RFC 1191, November 1990.
- [RFC1981] McCann, J., Deering, S., and J. Mogul, "Path MTU Discovery for IP version 6", RFC 1981, August 1996.
- [RFC2018] Mathis, M., Mahdavi, J., Floyd, S., and A. Romanow, "TCP Selective Acknowledgment Options", RFC 2018, October 1996.
- [RFC2460] Deering, S. and R. Hinden, "Internet Protocol, Version 6 (IPv6) Specification", RFC 2460, December 1998.
- [RFC3168] Ramakrishnan, K., Floyd, S., and D. Black, "The Addition of Explicit Congestion Notification (ECN) to IP", RFC 3168, September 2001.
- [RFC3205] Moore, K., "On the use of HTTP as a Substrate", BCP 56, RFC 3205, February 2002.
- [RFC3390] Allman, M., Floyd, S., and C. Partridge, "Increasing TCP's Initial Window", RFC 3390, October 2002.
- [RFC3436] Jungmaier, A., Rescorla, E., and M. Tuexen, "Transport Layer Security over Stream Control Transmission Protocol", RFC 3436, December 2002.
- [RFC3758] Stewart, R., Ramalho, M., Xie, Q., Tuexen, M., and P. Conrad, "Stream Control Transmission Protocol (SCTP) Partial Reliability Extension", RFC 3758, May 2004.
- [RFC3828] Larzon, L-A., Degermark, M., Pink, S., Jonsson, L-E., and G. Fairhurst, "The Lightweight User Datagram Protocol (UDP-Lite)", RFC 3828, July 2004.
- [RFC4336] Floyd, S., Handley, M., and E. Kohler, "Problem Statement for the Datagram Congestion Control Protocol (DCCP)", RFC 4336, March 2006.
- [RFC4340] Kohler, E., Handley, M., and S. Floyd, "Datagram Congestion Control Protocol (DCCP)", RFC 4340, March 2006.

- [RFC4341] Floyd, S. and E. Kohler, "Profile for Datagram Congestion Control Protocol (DCCP) Congestion Control ID 2: TCP-like Congestion Control", RFC 4341, March 2006.
- [RFC4342] Floyd, S., Kohler, E., and J. Padhye, "Profile for Datagram Congestion Control Protocol (DCCP) Congestion Control ID 3: TCP-Friendly Rate Control (TFRC)", RFC 4342, March 2006.
- [RFC4614] Duke, M., Braden, R., Eddy, W., and E. Blanton, "A Roadmap for Transmission Control Protocol (TCP) Specification Documents", RFC 4614, September 2006.
- [RFC4820] Tuexen, M., Stewart, R., and P. Lei, "Padding Chunk and Parameter for the Stream Control Transmission Protocol (SCTP)", RFC 4820, March 2007.
- [RFC4821] Mathis, M. and J. Heffner, "Packetization Layer Path MTU Discovery", RFC 4821, March 2007.
- [RFC4895] Tuexen, M., Stewart, R., Lei, P., and E. Rescorla, "Authenticated Chunks for the Stream Control Transmission Protocol (SCTP)", RFC 4895, August 2007.
- [RFC4960] Stewart, R., "Stream Control Transmission Protocol", RFC 4960, September 2007.
- [RFC5061] Stewart, R., Xie, Q., Tuexen, M., Maruyama, S., and M. Kozuka, "Stream Control Transmission Protocol (SCTP) Dynamic Address Reconfiguration", RFC 5061, September 2007.
- [RFC5097] Renker, G. and G. Fairhurst, "MIB for the UDP-Lite protocol", RFC 5097, January 2008.
- [RFC5246] Dierks, T. and E. Rescorla, "The Transport Layer Security (TLS) Protocol Version 1.2", RFC 5246, August 2008.
- [RFC5348] Floyd, S., Handley, M., Padhye, J., and J. Widmer, "TCP Friendly Rate Control (TFRC): Protocol Specification", RFC 5348, September 2008.
- [RFC5405] Eggert, L. and G. Fairhurst, "Unicast UDP Usage Guidelines for Application Designers", BCP 145, RFC 5405, November 2008.
- [RFC5595] Fairhurst, G., "The Datagram Congestion Control Protocol (DCCP) Service Codes", RFC 5595, September 2009.

- [RFC5596] Fairhurst, G., "Datagram Congestion Control Protocol (DCCP) Simultaneous-Open Technique to Facilitate NAT/Middlebox Traversal", RFC 5596, September 2009.
- [RFC5662] Shepler, S., Eisler, M., and D. Noveck, "Network File System (NFS) Version 4 Minor Version 1 External Data Representation Standard (XDR) Description", RFC 5662, January 2010.
- [RFC5672] Crocker, D., "RFC 4871 DomainKeys Identified Mail (DKIM) Signatures -- Update", RFC 5672, August 2009.
- [RFC6773] Phelan, T., Fairhurst, G., and C. Perkins, "DCCP-UDP: A Datagram Congestion Control Protocol UDP Encapsulation for NAT Traversal", RFC 6773, November 2012.
- [RFC5925] Touch, J., Mankin, A., and R. Bonica, "The TCP Authentication Option", RFC 5925, June 2010.
- [RFC5681] Allman, M., Paxson, V., and E. Blanton, "TCP Congestion Control", RFC 5681, September 2009.
- [RFC6083] Tuexen, M., Seggelmann, R., and E. Rescorla, "Datagram Transport Layer Security (DTLS) for Stream Control Transmission Protocol (SCTP)", RFC 6083, January 2011.
- [RFC6093] Gont, F. and A. Yourtchenko, "On the Implementation of the TCP Urgent Mechanism", RFC 6093, January 2011.
- [RFC6525] Stewart, R., Tuexen, M., and P. Lei, "Stream Control Transmission Protocol (SCTP) Stream Reconfiguration", RFC 6525, February 2012.
- [RFC6298] Paxson, V., Allman, M., Chu, J., and M. Sargent, "Computing TCP's Retransmission Timer", RFC 6298, June 2011.
- [RFC6935] Eubanks, M., Chimento, P., and M. Westerlund, "IPv6 and UDP Checksums for Tunneled Packets", RFC 6935, April 2013.
- [RFC6936] Fairhurst, G. and M. Westerlund, "Applicability Statement for the Use of IPv6 UDP Datagrams with Zero Checksums", RFC 6936, April 2013.
- [RFC6455] Fette, I. and A. Melnikov, "The WebSocket Protocol", RFC 6455, December 2011.

- [RFC6347] Rescorla, E. and N. Modadugu, "Datagram Transport Layer Security Version 1.2", RFC 6347, January 2012.
- [RFC6458] Stewart, R., Tuexen, M., Poon, K., Lei, P., and V. Yasevich, "Sockets API Extensions for the Stream Control Transmission Protocol (SCTP)", RFC 6458, December 2011.
- [RFC6691] Borman, D., "TCP Options and Maximum Segment Size (MSS)", RFC 6691, July 2012.
- [RFC6824] Ford, A., Raiciu, C., Handley, M., and O. Bonaventure, "TCP Extensions for Multipath Operation with Multiple Addresses", RFC 6824, January 2013.
- [RFC6951] Tuexen, M. and R. Stewart, "UDP Encapsulation of Stream Control Transmission Protocol (SCTP) Packets for End-Host to End-Host Communication", RFC 6951, May 2013.
- [RFC7053] Tuexen, M., Ruengeler, I., and R. Stewart, "SACK-IMMEDIATELY Extension for the Stream Control Transmission Protocol", RFC 7053, November 2013.
- [RFC7323] Borman, D., Braden, B., Jacobson, V., and R. Scheffenegger, "TCP Extensions for High Performance", RFC 7323, September 2014.
- [I-D.ietf-aqm-ecn-benefits] Welzl, M. and G. Fairhurst, "The Benefits and Pitfalls of using Explicit Congestion Notification (ECN)", draft-ietf-aqm-ecn-benefits-00 (work in progress), October 2014.
- [I-D.ietf-tsvwg-sctp-dtls-encaps] Tuexen, M., Stewart, R., Jesup, R., and S. Loreto, "DTLS Encapsulation of SCTP Packets", draft-ietf-tsvwg-sctp-dtls-encaps-09 (work in progress), January 2015.
- [I-D.ietf-tsvwg-sctp-prpolicies] Tuexen, M., Seggelmann, R., Stewart, R., and S. Loreto, "Additional Policies for the Partial Reliability Extension of the Stream Control Transmission Protocol", draft-ietf-tsvwg-sctp-prpolicies-07 (work in progress), February 2015.
- [I-D.ietf-tsvwg-sctp-ndata] Stewart, R., Tuexen, M., Loreto, S., and R. Seggelmann, "Stream Schedulers and a New Data Chunk for the Stream Control Transmission Protocol", draft-ietf-tsvwg-sctp-ndata-02 (work in progress), January 2015.

Authors' Addresses

Godred Fairhurst (editor)
University of Aberdeen
School of Engineering, Fraser Noble Building
Aberdeen AB24 3UE

Email: gorry@erg.abdn.ac.uk

Brian Trammell (editor)
ETH Zurich
Gloriastrasse 35
8092 Zurich
Switzerland

Email: ietf@trammell.ch

Mirja Kuehlewind (editor)
ETH Zurich
Gloriastrasse 35
8092 Zurich
Switzerland

Email: mirja.kuehlewind@tik.ee.ethz.ch