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Lossless and overhead free DCCP - UDP header conversion (U-DCCP)
draft-amend-tsvwg-dccp-udp-header-conversion-01

Abstract

The Datagram Congestion Control Protocol (DCCP) is a transport-layer protocol that provides upper layers with the ability to use non-reliable congestion-controlled flows. DCCP is not widely deployed in the Internet, and the reason for that can be defined as a typical example of a chicken-egg problem. Even if an application developer decided to use DCCP, the middle-boxes like firewalls and NATs would prevent DCCP end-to-end since they lack support for DCCP. Moreover, as long as the protocol penetration of DCCP does not increase, the middle-boxes will not handle DCCP properly. To overcome this challenge, NAT/NATP traversal and UDP encapsulation for DCCP is already defined. However, the former requires special middle-box support and the latter introduces overhead. The recent proposal of a multipath extension for DCCP further underlines the challenge of efficient middle-box passing as its main goal is to be applied over the Internet, traversing numerous uncontrolled middle-boxes. This document introduces a new solution which disguises DCCP during transmission as UDP without requiring middle-box modification or introducing any overhead.

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1. Introduction

The Datagram Congestion Control Protocol (DCCP) [RFC4340] is a transport-layer protocol that provides upper layers with the ability to use non-reliable congestion-controlled flows. The current specification for DCCP [RFC4340] specifies a direct native encapsulation in IPv4 or IPv6 packets.

DCCP support has been specified for devices that use Network Address Translation (NAT) or Network Address and Port Translation (NAPT)

[RFC5597]. However, there is a significant installed base of NAT/NAPT devices that do not support [RFC5597]. An UDP encapsulation for DCCP [RFC6773] circumvents such limitations and makes DCCP compatible with any UDP [RFC0768] compliant device that supports [RFC4787] but does not support [RFC5597]. For convenience, the standard encapsulation for DCCP [RFC4340] (including [RFC5596] and [RFC5597] as required) is referred to as DCCP-STD, whereas the UDP encapsulation for DCCP [RFC6773] is referred to as DCCP-UDP.

It can be stated that DCCP-STD and DCCP-UDP are techniques which increase the success rate of DCCP transmissions significantly. However, DCCP-STD fails on devices that block DCCP for any reasons. On the other hand, DCCP-UDP uses the well-accepted UDP to let devices assume they are handling the UDP protocol, but at the cost of a reduced goodput/throughput ratio.

To compensate for the inefficiency of DCCP-STD (device blocking) and DCCP-UDP (overhead), this document proposes a beneficial modification scheme relying on UDP (like DCCP-UDP), but with no overhead. This goal is reached by re-arranging DCCP's extended header to make it look like UDP, without losing critical information. This solution is referred to as U-DCCP.

U-DCCP is limited to DCCP's extended header, requiring X is set to 1. Otherwise U-DCCP relies on the NAT/NATP functionalities specified for UDP in [RFC4787], [RFC6888] and [RFC7857].

2. Terminology

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

3. U-DCCP

3.1. Overview

The basic approach of U-DCCP is to modify the extended header of a DCCP packet so that it appears like UDP [RFC0768]. In particular, this takes place without losing any header information, but requires a U-DCCP termination before the packet is delivered to the DCCP end system. This method does not change the 4-tuple of IP and port addressing, however it changes the protocol carried over IP from DCCP to UDP. As a consequence, the length of the packet remains unchanged and behaves like DCCP-STD. The solution is not a tunneling approach. It requires that the same port used by DCCP can be used by UDP.

The method is designed to support use when the IP addresses are modified by a device that implements NAT/NAPT. A NAT translates the IP addresses, which impacts the transport-layer checksum. A NAPT device may also translate the port values (usually the source port). In both cases, the outer transport header that includes these values would need to be updated by the NAT/NAPT.

U-DCCP supports IPv4 and IPv6.

The basic format of a U-DCCP packet is:

+-----+		
	IP Header (IPv4 or IPv6)	Variable length
+-----+		
	UDP like arranged DCCP ext. Header	8 bytes \
+-----+		
	Rest of rearranged DCCP ext. Header	8 bytes /
+-----+		
	Additional (type-specific) Fields	Variable length (could be 0)
+-----+		
	DCCP Options	Variable length (could be 0)
+-----+		
	Application Data Area	Variable length (could be 0)
+-----+		

Figure 1: Format of U-DCCP packet

The U-DCCP header is described in Section 3.4 after introducing the traditional DCCP header in Section 3.1 and its target appearance of a UDP header in Section 3.2. Section 3.3 discusses considerations for building the U-DCCP header upfront.

3.2. The DCCP Generic header

The DCCP Generic Header [RFC4340] takes two forms: one with long sequence numbers (48 bits) and the other with short sequence numbers (24 bits). The short one is not part of U-DCCP's modification.

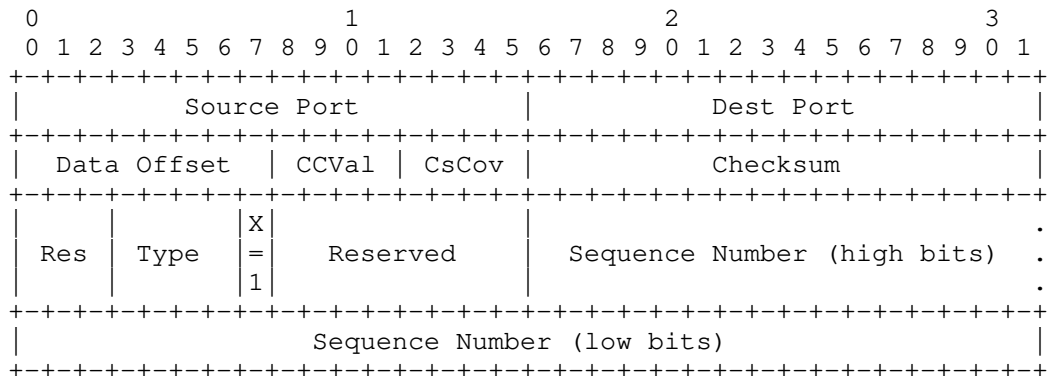


Figure 2: The extended DCCP Header with Long Sequence Numbers
[RFC4340]

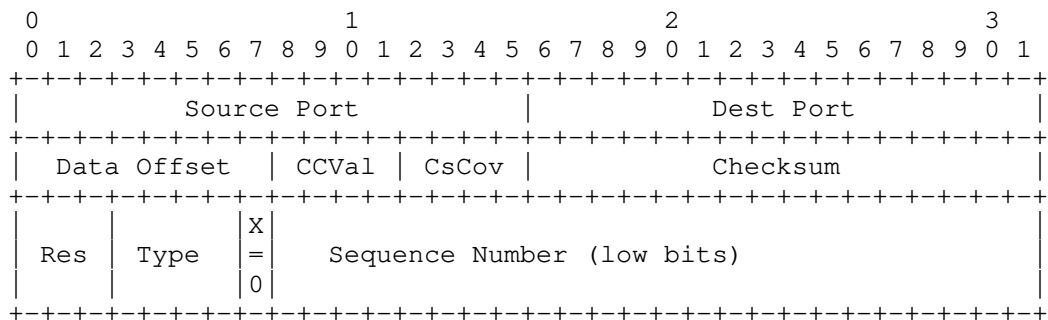


Figure 3: The short DCCP Header with Short Sequence Numbers [RFC4340]

All generic header fields have the meaning specified in [RFC4340], updated by [RFC5596].

3.3. UDP header

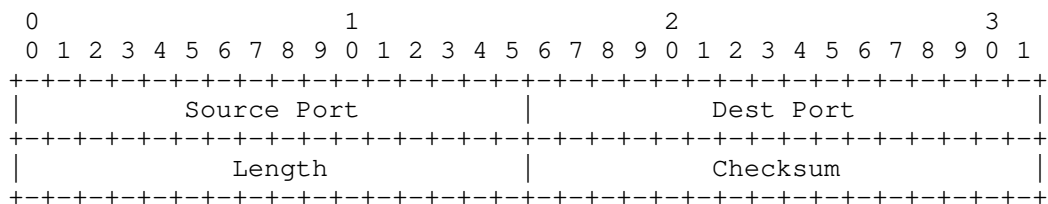


Figure 4: The UDP Header [RFC768]

All header fields have the meaning specified in [RFC0768].

3.4. U-DCCP conversion considerations

The U-DCCP header has the goal to merge the information of DCCP's extended header (Section 3.1) and imitates in the first 64 bits the UDP header (Section 3.2). Information required to restore a DCCP header from any conversion, which must not be lost, includes: source and destination port, Data Offset, CCVal, CsCov, Checksum, Type, X and the Sequence Number.

Compared with the UDP header, the DCCP extended header shows similarities in source and destination port and checksum. The length field of UDP (bits 33-48) is not part of the DCCP header and contains in case of DCCP the fields Data Offset, CCVal and CsCov.

For the goal of imitating UDP, the checksum must cover the whole datagram, which renders any limitation by CsCov useless. The checksum itself is required to re-calculate after conversion anyway.

If the conversion is limited to DCCP'S extended header only, X is always "1".

Thus, Data Offset, CCVal, Type and Sequence Number must be re-arranged in a way that the Length field of UDP can be applied.

3.5. U-DCCP header

The considerations of Section 3.3 leads to the following header, denoted as U-DCCP header.

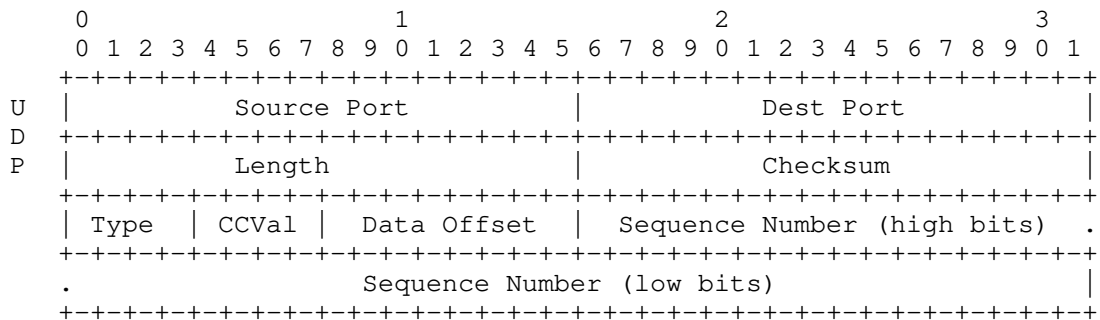


Figure 5: The U-DCCDP Header

The first 8 bytes of the U-DCCP header corresponds to [RFC0768] and the fields are interpreted as follows:

Source and Dest(ination) Ports: 16 bits each

These fields identify the UDP ports used by the source and destination (respectively) of the packet to listen for incoming UDP packets. The UDP port values identify the DCCP source and destination ports.

Length: 16 bits

This field is the length of the UDP datagram, including the UDP header and the payload (for U-DCCP, the payload comprises the payload of the original DCCP datagram and part of its header).

Checksum: 16 bits

This field is the Internet checksum of a network-layer pseudoheader and Length bytes of the UDP packet [RFC0768]. The UDP checksum MUST NOT be zero for a U-DCCP packet.

The remaining 8 bytes of the U-DCCP header contains:

Type, CCVal, Data Offset, Seq. Number: As specified in [RFC4340]

In case U-DCCP is applied, the IP layer must be instructed to carry an UDP datagram and its checksum must be re-calculated. For detailed information see Section 3.7.

3.6. Implementation

The process of applying U-DCCP is defined as follows:

DCCP generation -> U-DCCP conversion -> UDP transmission -> U-DCCP reception and restoration -> DCCP reception

The conversion can be integrated into DCCP endpoints directly or as an additional component on the way along the transmission route. Depending on the degree of integration, especially the process of checksum calculation and validation can be optimized. Section 3.7 and Section 3.8 provide a possible pseudo-code for the conversion without any optimized integration into the sender's network stack or into the receiver's network stack. The pseudo-code assumes explicit knowledge on which U-DCCP flows need conversion between the sender and the receiver.

3.7. Pseudo-code DCCP to U-DCCP conversion

A possible processing of an already generated DCCP datagram for U-DCCP conversion:

1. Receive DCCP datagram.

2. Check eligibility for conversion; otherwise bypass conversion.
3. Verify consistency, e.g. checksum; otherwise drop.
4. Shift Type and CCVal field to the ninth octet.
5. Shift Data Offset field to the tenth octet.
6. Place a length information at octet 5+6 corresponding to [RFC0768].
7. Modify the IP header's encapsulated protocol from DCCP to UDP.
8. Re-calculate IP header checksum.
9. Reset DCCP checksum field: octet 7+8 = 0.
10. Generate new checksum at octet 7+8 as described in [RFC0768].
11. Forward to destination based on the unmodified 4-tuple of IP-addresses and ports.

3.8. Pseudo-code U-DCCP to DCCP restoration

A possible processing of an already converted U-DCCP datagram for DCCP restoration:

1. Receive UDP datagram.
2. Check eligibility for restoration; otherwise bypass restoration
3. Validate UDP checksum; otherwise drop.
4. Restore Data Offset field according to [RFC4340].
5. Restore CCVal field according to [RFC4340].
6. Set CsCov field according to [RFC4340] to "0".
7. Restore Type field according to [RFC4340].
8. Set Reserved bits according to [RFC4340] to "0".
9. Set X according to [RFC4340] to "1".
10. Modify the IP header's encapsulated protocol from UDP to DCCP.
11. Re-calculate IP header checksum.

12. Reset DCCP checksum field: octet 7+8 = 0.
13. Generate new checksum at octet 7+8 as described in [RFC0768].
14. Forward to destination based on the unmodified 4-tuple of IP-addresses and ports.

3.9. U-DCCP negotiation (required???)

Tbd later if required. Otherwise assumes explicit knowledge about the U-DCCP conversion between sender and receiver.

4. Security Considerations

TBD.

5. IANA Considerations

6. Notes

This document is inspired by [RFC6773] and some text passages for the -00 version are copied unmodified.

7. Acknowledgments

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DCCP Extensions for Multipath Operation with Multiple Addresses
draft-amend-tsvwg-multipath-dccp-05

Abstract

DCCP communication is currently restricted to a single path per connection, yet multiple paths often exist between peers. The simultaneous use of these multiple paths for a DCCP session could improve resource usage within the network and, thus, improve user experience through higher throughput and improved resilience to network failures. Use cases for a Multipath DCCP (MP-DCCP) are mobile devices (handsets, vehicles) and residential home gateways simultaneously connected to distinct paths as, e.g., a cellular link and a WiFi link or to a mobile radio station and a fixed access network. Compared to existing multipath protocols such as MPTCP, MP-DCCP provides specific support for non-TCP user traffic as UDP or plain IP. More details on potential use cases are provided in [website], [slide] and [paper]. All these use cases profit from an Open Source Linux reference implementation provided under [website].

This document presents a set of extensions to traditional DCCP to support multipath operation. Multipath DCCP provides the ability to simultaneously use multiple paths between peers. The protocol offers the same type of service to applications as DCCP and it provides the components necessary to establish and use multiple DCCP flows across potentially disjoint paths.

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1. Introduction

Multipath DCCP (MP-DCCP) is a set of extensions to regular DCCP [RFC4340], i.e. the Datagram Congestion Control Protocol denoting a transport protocol that provides bidirectional unicast connections of congestion-controlled unreliable datagrams. A multipath extension to DCCP enables the transport of user data across multiple paths simultaneously. This is beneficial to applications that transfer fairly large amounts of data, due to the possibility to aggregate capacity of the multiple paths. In addition, it enables to tradeoff timeliness and reliability, which is important for low latency applications that do not require guaranteed delivery services such as Audio/Video streaming. DCCP multipath operation is suggested in the context of ongoing 3GPP work on 5G multi-access solutions [I-D.amend-tsvwg-multipath-framework-mpdccp] and for hybrid access networks [I-D.lhwxyz-hybrid-access-network-architecture][I-D.muley-net-work-based-bonding-hybrid-access]. It can be applied for load-balancing, seamless session handover, and aggregation purposes (referred to as ATSSS; Access steering, switching, and splitting in 3GPP terminology [TS23.501]).

This document presents the protocol changes required to add multipath capability to DCCP; specifically, those for signaling and setting up multiple paths ("subflows"), managing these subflows, re-assembly of data, and termination of sessions. DCCP, as stated in [RFC4340] does not provide reliable and ordered delivery. Consequently, multiple application subflows may be multiplexed over a single DCCP connection with no inherent performance penalty for flows that do not require in-ordered delivery. DCCP does not provide built-in support for those multiple application subflows.

In the following, use of the term subflow will refer to physical separate DCCP subflows transmitted via different paths, but not to application subflows. Application subflows are differing content-wise by source and destination port per application as, for example, enabled by Service Codes introduced to DCCP in [RFC5595], and those subflows can be multiplexed over a single DCCP connection. For sake of consistency we assume that only a single application is served by a DCCP connection here as shown in Figure 1 while use of that feature should not impact DCCP operation on each single path as noted in ([RFC5595], sect. 2.4).

1.1. Multipath DCCP in the Networking Stack

MP-DCCP operates at the transport layer and aims to be transparent to both higher and lower layers. It is a set of additional features on top of standard DCCP; Figure 1 illustrates this layering. MP-DCCP is designed to be used by applications in the same way as DCCP with no changes to the application itself.

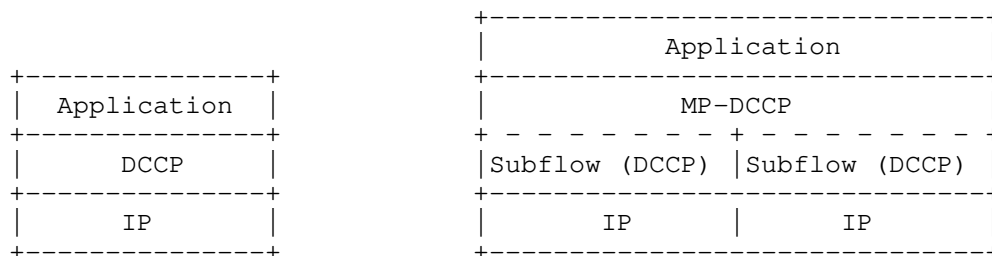


Figure 1: Comparison of Standard DCCP and MP-DCCP Protocol Stacks

1.2. Terminology

Throughout this document we make use of terms that are either specific for multipath transport or are defined in the context of MP-DCCP, similar to [RFC8684], as follows:

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

Subflow: A flow of DCCP segments operating over an individual path, which forms part of a larger MP-DCCP connection. A subflow is started and terminated similar to a regular (single-path) DCCP connection.

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

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

Host: An end host operating an MP-DCCP implementation, and either initiating or accepting an MP-DCCP connection. In addition to these terms, within framework of MP-DCCP the interpretation of, and effect on, regular single-path DCCP semantics is discussed in Section 3.

1.3. MP-DCCP Concept

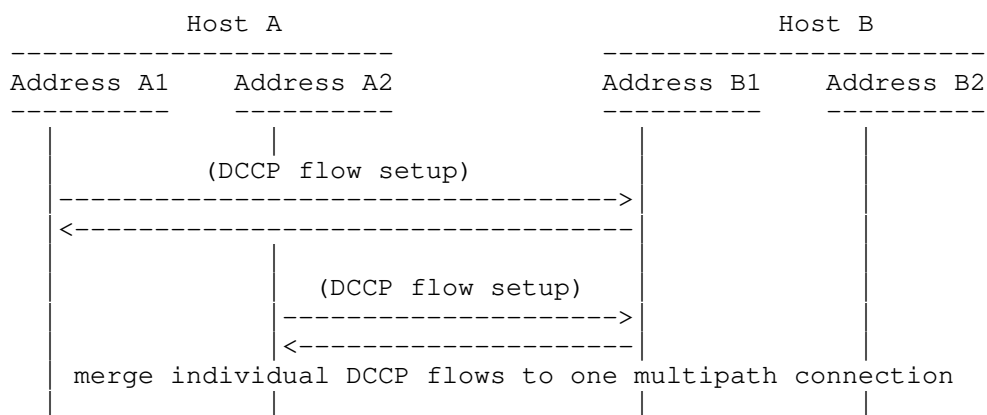


Figure 2: Example MP-DCCP Usage Scenario

1.4. Differences from Multipath TCP

Multipath DCCP is similar to Multipath TCP [RFC8684], in that it extends the related basic DCCP transport protocol [RFC4340] with multipath capabilities in the same way as Multipath TCP extends TCP [RFC0793]. However, because of the differences between the underlying TCP and DCCP protocols, the transport characteristics of MPTCP and MP-DCCP are different.

Table 1 compares the protocol characteristics of TCP and DCCP, which are by nature inherited by their respective multipath extensions. A major difference lies in the delivery of payload, which is for TCP an exact copy of the generated byte-stream. DCCP behaves in a different way and does not guarantee to deliver any payload nor the order of delivery. Since this is mainly affecting the receiving endpoint of a TCP or DCCP communication, many similarities on the sender side can be identified. Both transport protocols share the 3-way initiation of a communication and both employ congestion control to adapt the sending rate to the path characteristics.

Feature	TCP	DCCP
Full-Duplex	yes	yes
Connection-Oriented	yes	yes
Header option space	40 bytes	< 1008 bytes or PMTU
Data transfer	reliable	unreliable
Packet-loss handling	re-transmission	report only
Ordered data delivery	yes	no
Sequence numbers	one per byte	one per PDU
Flow control	yes	no
Congestion control	yes	yes
ECN support	yes	yes
Selective ACK	yes	depends on congestion control
Fix message boundaries	no	yes
Path MTU discovery	yes	yes
Fragmentation	yes	no
SYN flood protection	yes	no
Half-open connections	yes	no

Table 1: TCP and DCCP protocol comparison

Consequently, the multipath features, shown in Table 2, are the same, supporting volatile paths having varying capacity and latency, session handover and path aggregation capabilities. All of them profit by the existence of congestion control.

Feature	MPTCP	MP-DCCP
Volatile paths	yes	yes
Session handover	yes	yes
Path aggregation	yes	yes
Robust session establishment	no	yes
Data re-assembly	yes	optional / modular
Expandability	limited by TCP header	flexible

Table 2: MPTCP and MP-DCCP protocol comparison

Therefore, the sender logic is not much different between MP-DCCP and MPTCP, even if the multipath session initiation differs. MP-DCCP inherits a robust session establishment feature, which guarantees communication establishment if at least one functional path is available. MPTCP relies on an initial path, which has to work; otherwise no communication can be established.

The receiver side for MP-DCCP has to deal with the unreliable transport character of DCCP and a possible re-assembly of the data stream while not advocating it. As many unreliable applications have built-in application support for reordering (such as adaptive audio and video buffers), those applications might not need support for re-assembly. However, for applications that benefit from partial or full support of reordering, MP-DCCP can provide flexible support for re-assembly, even if for DCCP the order of delivery is unreliable by nature. Such optional re-assembly mechanisms may account for the fact that packet loss may occur for any of the DCCP subflows. Another issue may occur as packet reordering may happen when the different DCCP subflows are routed across paths with different latencies. In theory, applications using DCCP are aware that packet reordering might happen, since DCCP has no mechanisms to prevent it.

The receiving process for MPTCP is on the other hand a rigid "just wait" approach, since TCP guarantees reliable delivery.

1.5. Requirements Language

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

2. Operation Overview

RFC 4340 states that some applications might want to share congestion control state among multiple DCCP flows between same source and destination addresses. This functionality could be provided by the Congestion Manager (CM) [RFC3124], a generic multiplexing facility. However, the CM would not fully support MP-DCCP without change; it does not gracefully handle multiple congestion control mechanisms, for example.

The operation of MP-DCCP for data transfer takes one input data stream from an application, and splits it into one or more subflows, with sufficient control information to allow received data to be re-assembled and delivered in order to the recipient application. The following subsections define this behavior in detail.

The Multipath Capability for MP-DCCP can be negotiated with a new DCCP feature, as described in Section 3. Once negotiated, all subsequent MP-DCCP operations are signalled with a variable length multipath-related option, as described in Section 3.1.

3. MP-DCCP Protocol

The DCCP protocol feature list ([RFC4340] section 6.4) will be enhanced by a new Multipath related feature with Feature number 10, as shown in Table 3.

Number	Meaning	Rule	Rec'n Value	Initial Req'd
0	Reserved			
1	Congestion Control ID (CCID)	SP	2	Y
2	Allow Short Seqnos	SP	0	Y
3	Sequence Window	NN	100	Y
4	ECN Incapable	SP	0	N
5	Ack Ratio	NN	2	N
6	Send Ack Vector	SP	0	N
7	Send NDP Count	SP	0	N
8	Minimum Checksum Coverage	SP	0	N
9	Check Data Checksum	SP	0	N
10	Multipath Capable	SP	0	N
11-127	Reserved			
128-255	CCID-specific features			

Table 3: Proposed Feature Set

The DCCP protocol options as defined in ([RFC4340] section 5.8) and ([RFC5634] section 2.2.1) will be enhanced by a new Multipath related variable-length option with option type 46, as shown in Table 4.

Type	Option Length	Meaning	DCCP-Data?
0	1	Padding	Y
1	1	Mandatory	N
2	1	Slow Receiver	Y
3-31	1	Reserved	
32	variable	Change L	N
33	variable	Confirm L	N
34	variable	Change R	N
35	variable	Confirm R	N
36	variable	Init Cookie	N
37	3-8	NDP Count	Y
38	variable	Ack Vector [Nonce 0]	N
39	variable	Ack Vector [Nonce 1]	N
40	variable	Data Dropped	N
41	6	Timestamp	Y
42	6/8/10	Timestamp Echo	Y
43	4/6	Elapsed Time	N
44	6	Data Checksum	Y
45	8	Quick-Start Response	?
46	variable	Multipath	Y
47-127	variable	Reserved	
128-255	variable	CCID-specific options	-

Table 4: Proposed Option Set

[Tbd/tbv] In addition to the multipath option, MP-DCCP requires particular considerations for:

- * The minimum PMTU of the individual paths must be announced to the application. Changes of individual path PMTUs must be re-announced to the application if they result in a value lower than the currently announced PMTU.
- * Overall sequencing for optional re-assembly procedure
- * Congestion control
- * Robust MP-DCCP session establishment (no dependency on an initial path setup)

3.1. Multipath Capable Feature

DCCP endpoints are multipath-disabled by default and multipath capability can be negotiated with the Multipath Capable Feature.

Multipath Capable has feature number 10 and is server-priority. It takes one-byte values. The first four bits are used to specify compatible versions of the MP-DCCP implementation. The following four bits are reserved for further use.

3.2. Multipath Option

```

+-----+-----+-----+-----+-----+
|00101110| Length | MP_OPT | Value(s) ...
+-----+-----+-----+-----+-----+
Type=46

```

Type	Option Length	MP_OPT	Meaning
46	var	0 =MP_CONFIRM	Confirm reception and processing of an MP_OPT option
46	11	1 =MP_JOIN	Join path to an existing MP-DCCP flow
46	3	2 =MP_FAST_CLOSE	Close MP-DCCP flow
46	var	3 =MP_KEY	Exchange key material for MP_HMAC
46	7	4 =MP_SEQ	Multipath Sequence Number
46	23	5 =MP_HMAC	HMA Code for authentication
46	12	6 =MP_RTT	Transmit RTT values
46	var	7 =MP_ADDADDR	Advertise additional Address
46	var	8 =MP_REMOVEADDR	Remove Address
46	4	9 =MP_PRIO	Change Subflow Priority

Table 5: MP_OPT Option Types

3.2.1. MP_CONFIRM

```

+-----+-----+-----+-----+
|00101110| Length |00000000| List of options ...
+-----+-----+-----+-----+
Type=46      MP_OPT=0

```

MP_CONFIRM can be used to send confirmation of received and processed options. Confirmed options are copied verbatim and appended as List of options. The length varies dependent on the amount of options.

[Tbd] Encoding "list of options"

3.2.2. MP_JOIN


```

+-----+-----+-----+-----+-----+-----+
|00101110|00001011|00000001| Path Token |
+-----+-----+-----+-----+-----+-----+
| Nonce |
+-----+-----+-----+-----+
Type=46 Length=11 MP_OPT=1

```

The MP_JOIN option is used to add a new path to an existing MP-DCCP flow. The Path Token is the SHA-1 HASH of the derived key (d-key), which was previously exchanged with the MP_KEY option. MP_HMAC MUST be set when using MP_JOIN to provide authentication (See MP_HMAC for details). Also MP_KEY MUST be set to provide key material for authentication purposes.

3.2.3. MP_FAST_CLOSE

```

+-----+-----+-----+
|00101110|00000011|00000010|
+-----+-----+-----+
Type=46 Length=3 MP_OPT=2

```

MP_FAST_CLOSE terminates the MP-DCCP flow and all corresponding subflows.

3.2.4. MP_KEY

```

+-----+-----+-----+-----+-----+-----+
|00101110| Length |00000011|Key Type| Key Data ...
+-----+-----+-----+-----+-----+-----+
Type=46 MP_OPT=3

```

The MP_KEY suboption is used to exchange key material between hosts. The Length varies between 5 and 8 Bytes. The Key Type field is used to specify the key type. Key types are shown in Table 6.

Key Type	Key Length	Meaning
0 =Plain Text	8	Plain Text Key
1 =ECDHE-C25519-SHA256	32	ECDHE with SHA256 and Curve25519
2 =ECDHE-C25519-SHA512	32	ECDHE with SHA512 and Curve25519
3-255		Reserved

Table 6: MP_KEY Key Types

Plain Text

Key Material is exchanged in plain text between hosts, and the key parts (key-a, key-b) are used by each host to generate the derived key (d-key) by concatenating the two parts with the local key in front (e.g. hostA d-key=(key-a+key-b), hostB d-key=(key-b+key-a)).

ECDHE-SHA256-C25519

Key Material is exchanged via ECDHE key exchange with SHA256 and Curve 25519 to generate the derived key (d-key).

ECDHE-SHA512-C25519

Key Material is exchanged via ECDHE key exchange with SHA512 and Curve 25519 to generate the derived key (d-key).

3.2.5. MP_SEQ

```

+-----+-----+-----+-----+
| 00101110 | 00000111 | 00000100 | Multipath Sequence Number |
+-----+-----+-----+-----+
Type=46 Length=7 MP_OPT=4

```

The MP_SEQ option is used for end-to-end datagram-based sequence numbers of an MP-DCCP connection. The initial data sequence number (IDSN) SHOULD be set randomly. The MP_SEQ number space is different from path individual sequence number space.

3.2.6. MP_HMAC

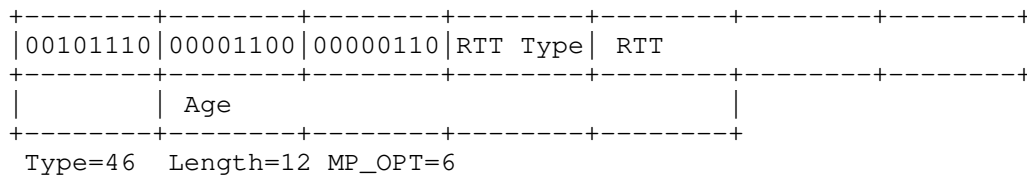
```

+-----+-----+-----+-----+
| 00101110 | 00001011 | 00000101 | HMAC-SHA1 (20 bytes) ... |
+-----+-----+-----+-----+
Type=46 Length=23 MP_OPT=5

```

The MP_HMAC option is used to provide authentication for the MP_JOIN option. The HMAC is built using the derived key (d-key) calculated previously from the handshake key material exchanged with the MP_KEY option. The Message for the HMAC is the header of the MP_JOIN for which authentication shall be performed. By including a nonce in these datagrams, possible replay-attacks are remedied.

3.2.7. MP_RTT



The MP_RTT option is used to transmit RTT values in milliseconds and MUST belong to the path over which this information is transmitted. Additionally, the age of the measurement is specified in milliseconds.

Raw RTT (=0)

Raw RTT value of the last Datagram Round-Trip. The Age parameter is set to the age of when the Ack for the datagram was received.

Min RTT (=1)

Min RTT value. The period for computing the Minimum can be specified by the Age parameter.

Max RTT (=2)

Max RTT value. The period for computing the Maximum can be specified by the Age parameter.

Smooth RTT (=3)

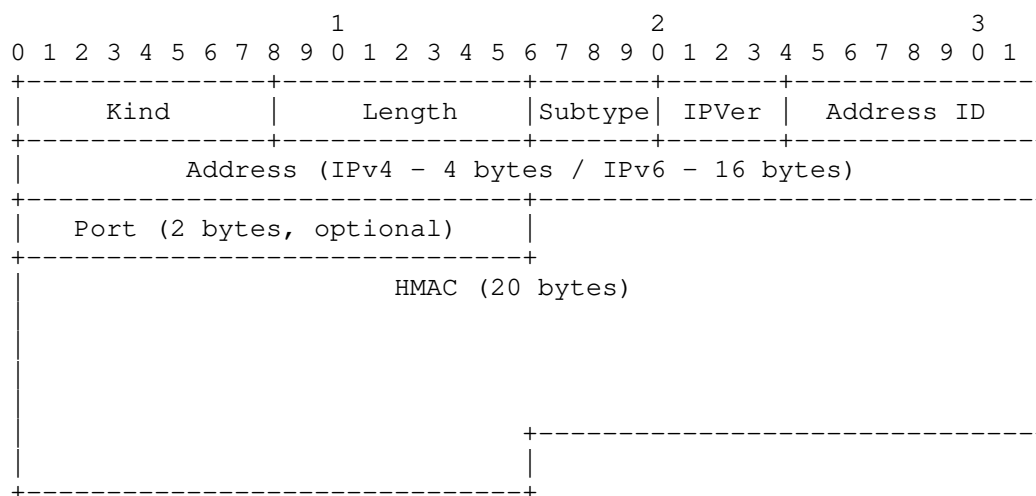
Averaged RTT value. The period for computing the smoothed RTT can be specified by the Age parameter.

Age (=4)

The Age parameter is a 4-byte value which is set to the age or timestamp when the Ack for the datagram was received in case of RTT type = 0 and may contain the periods for computing of derived RTT values depending on other RTT types, i.e., the Minimum (=1) and Maximum (=2) as well as the averaged smoothed RTT value (=3). [TBD/TBV]

3.2.8. MP_ADDADDR

The MP_ADDADDR option announces additional addresses (and, optionally, ports) on which a host can be reached. This option can be used at any time during an existing DCCP connection, when the sender wishes to enable multiple paths and/or when additional paths become available. Length is variable depending on IPv4 or IPv6 and whether port number is used and is in range between 28 and 42 bytes.



Every address has an Address ID that can be used for uniquely identifying the address within a connection for address removal. The Address ID is also used to identify MP_JOIN options (see Section 3.2.2) relating to the same address, even when address translators are in use. The Address ID MUST uniquely identify the address for the sender of the option (within the scope of the connection); the mechanism for allocating such IDs is implementation specific.

All Address IDs learned via either MP_JOIN or ADD_ADDR SHOULD be stored by the receiver in a data structure that gathers all the Address-ID-to-address mappings for a connection (identified by a token pair). In this way, there is a stored mapping between the Address ID, observed source address, and token pair for future processing of control information for a connection.

Ideally, ADD_ADDR and REMOVE_ADDR options would be sent reliably, and in order, to the other end. This would ensure that this address management does not unnecessarily cause an outage in the connection when remove/add addresses are processed in reverse order, and also to ensure that all possible paths are used. Note, however, that losing

reliability and ordering will not break the multipath connections, it will just reduce the opportunity to open new paths and to survive different patterns of path failures.

Therefore, implementing reliability signals for these DCCP options is not necessary. In order to minimize the impact of the loss of these options, however, it is RECOMMENDED that a sender should send these options on all available subflows. If these options need to be received in order, an implementation SHOULD only send one ADD_ADDR/REMOVE_ADDR option per RTT, to minimize the risk of misordering. A host that receives an ADD_ADDR but finds a connection set up to that IP address and port number is unsuccessful SHOULD NOT perform further connection attempts to this address/port combination for this connection. A sender that wants to trigger a new incoming connection attempt on a previously advertised address/port combination can therefore refresh ADD_ADDR information by sending the option again.

[TBD/TBV]

3.2.9. MP_REMOVEADDR

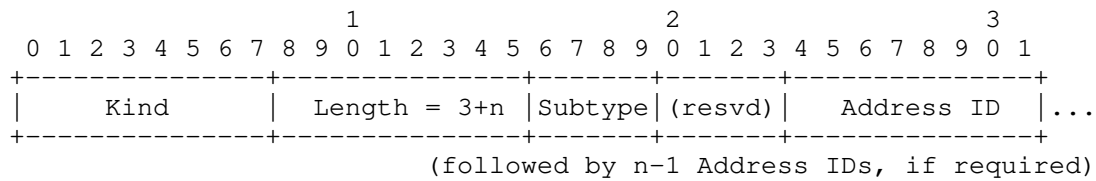
If, during the lifetime of an MP-DCCP connection, a previously announced address becomes invalid (e.g., if the interface disappears), the affected host SHOULD announce this so that the peer can remove subflows related to this address.

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

For security purposes, if a host receives a REMOVE_ADDR option, it must ensure the affected path(s) are no longer in use before it instigates closure. Typical DCCP validity tests on the subflow (e.g., packet type specific sequence and acknowledgement number check) MUST also be undertaken. An implementation can use indications of these test failures as part of intrusion detection or error logging.

The sending and receipt of this message SHOULD trigger the sending of DCCP-Close and DCCP-Reset by client and server, respectively on the affected subflow(s) (if possible), as a courtesy to cleaning up middlebox state, before cleaning up any local state.

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

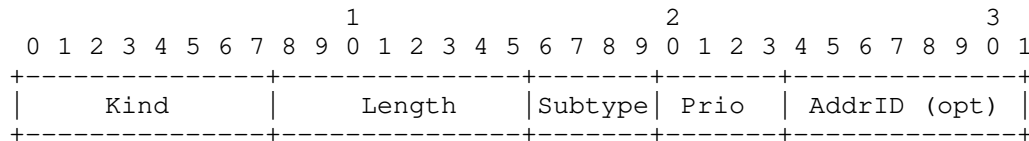


Minimum length of this option is 4 bytes (for one address to remove).

[TBD/TBV]

3.2.10. MP_PRIO

In the event that a single specific path out of the set of available paths shall be treated with higher priority compared to the others, a host may wish to signal such change in priority of subflows to the peer. Therefore, the MP_PRIO option, shown below, can be used to set a priority flag for the subflow on which it is sent.



Whether more than two values for priority (e.g., B for backup and P for prioritized path) are defined in case of more than two parallel paths is for further consideration.

[TBD/TBV]

3.3. MP-DCCP Handshaking Procedure

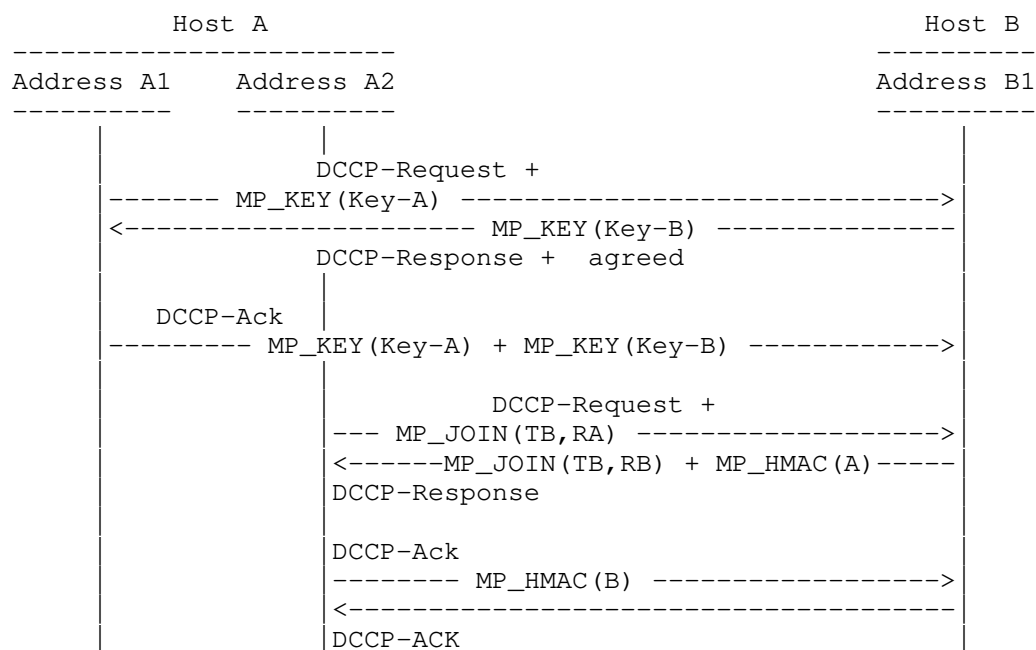


Figure 3: Example MP-DCCP Handshake

The basic initial handshake for the first flow is as follows:

- * Host A sends a DCCP-Request with the MP-Capable feature Change request and the MP_KEY option with Host-specific Key-A
- * Host B sends a DCCP-Response with Confirm feature for MP-Capable and the MP_Key option with Host-specific Key-B
- * Host A sends a DCCP-Ack with both Keys echoed to Host B.

The handshake for subsequent flows based on a successful initial handshake is as follows:

- * Host A sends a DCCP-Request with the MP-Capable feature Change request and the MP_JOIN option with Host B's Token TB, generated from the derived key by applying a SHA-1 hash and truncating to the first 32 bits. Additionally, an own random nonce RA is transmitted with the MP_JOIN.
- * Host B computes the HMAC of the DCCP-Request and sends a DCCP-Response with Confirm feature option for MP-Capable and the MP_JOIN option with the Token TB and a random nonce RB together with the computed MP_HMAC.

- * Host A sends a DCCP-Ack with the HMAC computed for the DCCP-Response.
- * Host B sends a DCCP-Ack confirm the HMAC and to conclude the handshaking.

4. Security Considerations

Similar to DCCP, MP-DCCP does not provide cryptographic security guarantees inherently. Thus, if applications need cryptographic security (integrity, authentication, confidentiality, access control, and anti-replay protection) the use of IPsec or some other kind of end-to-end security is recommended; Secure Real-time Transport Protocol (SRTP) [RFC3711] is one candidate protocol for authentication. Together with Encryption of Header Extensions in SRTP, as provided by [RFC6904], also integrity would be provided.

As described in [RFC4340], DCCP provides protection against hijacking and limits the potential impact of some denial-of-service attacks, but DCCP provides no inherent protection against attackers' snooping on data packets. Regarding the security of MP-DCCP no additional risks should be introduced compared to regular DCCP of today. Thereof derived are the following key security requirements to be fulfilled by MP-DCCP:

- * Provide a mechanism to confirm that parties involved in a subflow handshake are identical to those in the original connection setup.
- * Provide verification that the new address to be included in a MP connection is valid for a peer to receive traffic at before using it.
- * Provide replay protection, i.e., ensure that a request to add/remove a subflow is 'fresh'.

In order to achieve these goals, MP-DCCP includes a hash-based handshake algorithm documented in Sections Section 3.2.4 and Section 3.3. The security of the MP-DCCP connection depends on the use of keys that are shared once at the start of the first subflow and are never sent again over the network. To ease demultiplexing while not giving away any cryptographic material, future subflows use a truncated cryptographic hash of this key as the connection identification "token". The keys are concatenated and used as keys for creating Hash-based Message Authentication Codes (HMACs) used on subflow setup, in order to verify that the parties in the handshake are the same as in the original connection setup. It also provides verification that the peer can receive traffic at this new address. Replay attacks would still be possible when only keys are used;

therefore, the handshakes use single-use random numbers (nonces) at both ends -- this ensures that the HMAC will never be the same on two handshakes. Guidance on generating random numbers suitable for use as keys is given in [RFC4086]. During normal operation, regular DCCP protection mechanisms (such as header checksum to protect DCCP headers against corruption) will provide the same level of protection against attacks on individual DCCP subflows as exists for regular DCCP today.

5. Interactions with Middleboxes

Issues from interaction with on-path middleboxes such as NATs, firewalls, proxies, intrusion detection systems (IDSs), and others have to be considered for all extensions to standard protocols since otherwise unexpected reactions of middleboxes may hinder its deployment. DCCP already provides means to mitigate the potential impact of middleboxes, also in comparison to TCP (see [RFC4043], sect. 16). In case, however, both hosts are located behind a NAT or firewall entity, specific measures have to be applied such as the [RFC5596]-specified simultaneous-open technique that update the (traditionally asymmetric) connection-establishment procedures for DCCP. Further standardized technologies addressing NAT type middleboxes are covered by [RFC5597].

[RFC6773] specifies UDP Encapsulation for NAT Traversal of DCCP sessions, similar to other UDP encapsulations such as for SCTP [RFC6951]. The alternative U-DCCP approach proposed in [I-D.amend-tsvwg-dccp-udp-header-conversion] would reduce tunneling overhead. The handshaking procedure for DCCP-UDP header conversion or use of a DCCP-UDP negotiation procedure to signal support for DCCP-UDP header conversion would require encapsulation during the handshakes and use of two additional port numbers out of the UDP port number space, but would require zero overhead afterwards.

6. Implementation

The approach described above has been implemented in open source across different testbeds and a new scheduling algorithm has been extensively tested. Also demonstrations of a laboratory setup have been executed and have been published at [website].

7. Acknowledgments

1. Notes

This document is inspired by Multipath TCP [RFC6824]/[RFC8684] and some text passages for the -00 version of the draft are copied almost unmodified.

8. IANA Considerations

This document defines one new value to DCCP feature list and one new DCCP Option with ten corresponding Subtypes as follows. This document defines a new DCCP feature parameter for negotiating the support of multipath capability for DCCP sessions between hosts as described in Section 3. The following entry in Table 7 should be added to the "Feature Numbers Registry" according to [RFC4340], Section 19.4. under the "DCCP Protocol" heading.

Value	Feature Name	Specification
0x10	MP-DCCP capability feature	Section 3.1

Table 7: Addition to DCCP Feature list Entries

This document defines a new DCCP protocol option of type=46 as described in Section 3.2 together with 10 additional sub-options. The following entries in Table 8 should be added to the "DCCP Protocol options" and assigned as "MP-DCCP sub-options", respectively.

Value	Symbol	Name	Reference
TBD or Type=46	MP_OPT	DCCP Multipath option	Section 3.2
TBD or MP_OPT=0	MP_CONFIRM	Confirm reception/ processing of an MP_OPT option	Section 3.2.1
TBD or MP_OPT=1	MP_JOIN	Join path to existing MP-DCCP flow	Section 3.2.2
TBD or MP_OPT=2	MP_FAST_CLOSE	Close MP-DCCP flow	Section 3.2.3
TBD or MP_OPT=3	MP_KEY	Exchange key material for MP_HMAC	Section 3.2.4
TBD or MP_OPT=4	MP_SEQ	Multipath Sequence Number	Section 3.2.5
TBD or MP_OPT=5	MP_HMAC	Hash-based Message Auth. Code for MP- DCCP	Section 3.2.6
TBD or MP_OPT=6	MP_RTT	Transmit RTT values and calculation parameters	Section 3.2.7
TBD or MP_OPT=7	MP_ADDADDR	Advertise additional Address(es)/Port(s)	Section 3.2.8
TBD or MP_OPT=8	MP_REMOVEADDR	Remove Address(es)/ Port(s)	Section 3.2.9
TBD or MP_OPT=9	MP_PRIO	Change Subflow Priority	Section 3.2.10

Table 8: Addition to DCCP Protocol options and
corresponding sub-options

[Tbd], must include options for:

- * handshaking procedure to indicate MP support
- * handshaking procedure to indicate JOINING of an existing MP connection
- * signaling of new or changed addresses
- * setting handover or aggregation mode
- * setting reordering on/off

should include options carrying:

- * overall sequence number for restoring purposes
- * sender time measurements for restoring purposes
- * scheduler preferences
- * reordering preferences

9. Informative References

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A multipath framework for UDP traffic over heterogeneous access networks
draft-amend-tsvwg-multipath-framework-mpdccp-01

Abstract

More and more of today's devices are multi-homing capable, in particular 3GPP user equipment like smartphones. In the current standardization of the next upcoming mobile network generation 5G Rel.16, this is especially targeted in the study group Access Traffic Steering Switching Splitting [TR23.793]. ATSSS describes the flexible selection or combination of 3GPP untrusted access like Wi-Fi and cellular access, overcoming the single-access limitation of today's devices and services. Another multi-connectivity scenario is the Hybrid Access [I-D.lhwxyz-hybrid-access-network-architecture][I-D.muley-network-based-bonding-hybrid-access], providing multiple access for CPEs, which extends the traditional way of single access connectivity at home to dual-connectivity over 3GPP and fixed access. A missing piece in the ATSSS and Hybrid Access is the access and path measurement, which is required for efficient and beneficial traffic steering decisions. This becomes particularly important in heterogeneous access networks with a multitude of volatile access paths. While MP-TCP has been proposed to be used within ATSSS, there are drawbacks when being used to encapsulate unreliable traffic as it blindly retransmits each lost frame leading to excessive delay and potential head-of-line blocking. A decision for MP-TCP though leaves the increasing share of UDP in today's traffic mix (<<https://arxiv.org/abs/1801.05168>>) unconsidered. In this document, a multi-access framework is proposed leveraging the MP-DCCP network protocol, which enables flexible traffic steering, switching and splitting also for unreliable traffic. A benefit is the support for pluggable congestion control which enables our framework to be used either independent or complementary to MP-TCP.

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1. Introduction

Multi-connectivity access networks are evolving. Ongoing standardization at 3GPP for 5G mobile networks [TR23.793] or the so called Hybrid Access networks [I-D.lhwxyz-hybrid-access-network-architecture][I-D.muley-network-based-bonding-hybrid-access] already provides or will enable in the near future the possibility to use multi-connectivity for a very large number of mobile users. Multi-connectivity solutions come with many user benefits including superior resilience against network outages, higher capacities for user traffic and network cost optimizations. Since multi-connectivity architectures are almost mature, new network protocols are required to fully exploit multi-connectivity and maximise its potential. In the simplest case, multi-connectivity is used for load-balancing decisions in order to balance the user flows over multiple paths. However, this has no effect on resilience or capacity gain on those load balanced individual flows. More complex scenarios include the dynamic shifting of traffic flows seamlessly between multiple paths or even aggregating those paths for leveraging the available capacity of multiple individual paths. Like [TR23.793] this document refers to the three distribution schemes Steering (load balancing), Switching (seamless handover) and Splitting (capacity aggregation).

MP-TCP [RFC6824] is a protocol, which can be applied in the above mentioned use cases and supports load-balancing, traffic shifting among the multiple paths and also capacity aggregation. Further, it leverages the inherent congestion control from TCP which adapts the sending rate by observing congestion signals from the network. By design, MP-TCP is limited to TCP services as it blindly re-transmits lost packets. Consequently, when MP-TCP is used as a framework for ATSSS, it may re-transmit packets sent from unreliable services such as e.g. UDP unnecessary. This may lead to head-of-line blocking and increased latency, which is detrimental to real-time services. As future multi-connectivity systems must support latency sensitive traffic that might be transported over unreliable transport, it is not sufficient anymore to rely on supporting only TCP. The increasing share of UDP traffic, mainly impacted by the QUIC introduction, may significantly reduce the share from TCP. It might be expected that with HTTP/3 carried over QUIC [I-D.ietf-quic-http], the previous strong dominance of TCP will be challenged by UDP.

2. Requirements

A multiaccess framework shall meet the following requirements:

- o IP compatibility: A multiaccess framework shall be able to transport IP packets and not make any assumptions on which transport protocol is encapsulated.
- o Support for unreliable traffic: A multiaccess framework should provide support for transporting unreliable traffic, such as QUIC or UDP based flows. Therefore, unreliable transmission should be supported.
- o Support for flexible re-ordering: A multiaccess framework should support flexible re-ordering of user traffic, including no re-ordering at all. This requirement is important to support low latency traffic, where the re-creation of packet order may negatively impact delivery latency.
- o Support for flexible congestion control: A multiaccess framework should support flexible congestion control, including the disabling of the congestion control, if the inner traffic is known to be congestion controlled.
- o Support for flexible packet scheduling: A multiaccess framework should support different packet scheduling mechanisms, which should be configurable from the control plane. Examples are cheapest path first, or other more sophisticated schedulers.
- o Lightweight: A multiaccess framework should be lightweight in computational resources and limit the encapsulation overhead.

To use QUIC as part of a multiaccess framework, by for example providing multipath support for QUIC, it could be beneficial if unreliable transmission is supported as well as being able to influence or disable QUIC's congestion control. In addition, it would be beneficial if the encryption of QUIC can be disabled. This is because for ATSSS, it is foreseen that the underlying tunnel from the mobile over public WLANs is based on IPsec.

3. IP compatible multipath framework based on MP-DCCP

We propose a new multiaccess framework, which overcomes MP-TCP's restriction to TCP services and provides IP compatibility in Figure 1. The framework employs MP-DCCP [I-D.amend-tsvwg-multipath-dccp] in combination with an encapsulation scheme. For simplification, Figure 1 assumes a traffic direction from the left (sender) to the right (receiver) and requires application in each direction for bi-directional transmission. The framework in Figure 1 can replace or complement MP-TCP to reach IP compatibility.

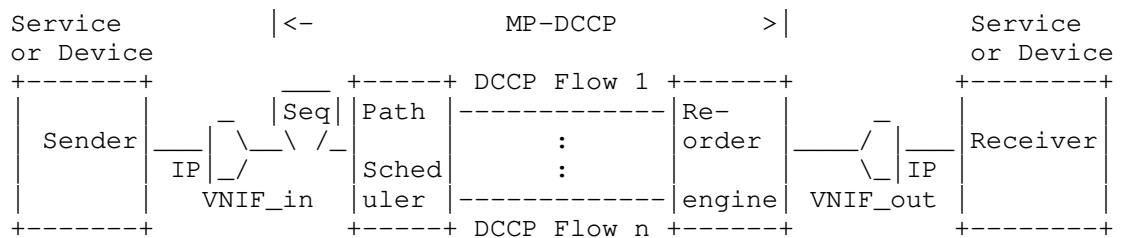


Figure 1: IP compatible multipath framework based on MP-DCCP

PDUs generated from the sender and travelling through the framework in Figure 1 pass the components in the following order:

1. Sender: Generates any PDU based on the IP protocol.
2. VNIF_in: IP based Virtual Network Interface as entry point to the multipath framework. A simple routing logic in front (between (1) and (2)) can act as gatekeeper and decides upon redirecting traffic through the VNIF or bypassing it. The VNIF adds an extra IP header to reach the multi-connectivity termination point.
3. Seq: Sequencing of the PDUs passed through (2) depending on the incoming order. Adds an incrementing number, which is later added to the DCCP encapsulation in (4).
4. Path Scheduler: Decision logic for scheduling sequenced PDUs over the individual connected DCCP flows for multipath transmission. The path scheduler can use the information from the DCCP flows (see (5)) inherent congestion control information like CWND, packet loss, RTT, Jitter, etc.. After selection of a DCCP flow, the PDU is encapsulated into the individual flow. Further information, at least the sequencing, is added on top as DCCP option.
5. DCCP Flow(s): Responsible to transmit the encapsulated PDUs to the MP-DCCP exit point.
6. Reorder engine: Depending on the sequencing information of (3), a re-assembly of the PDU stream can be applied. Different re-order algorithms should be supported in a configurable way, including no re-ordering.
7. VNIF_out: Releases PDUs that have passed the re-ordering engine and strips the DCCP specific overhead. Again, routing is responsible to deliver the PDUs to the receiver based on the destination information in the PDU.

8. Receiver: Receive the PDU as generated in (1).

The simple enclosing of the MP-DCCP with Virtual Network Interface (VNIF) provides the IP compatibility. However, a service or protocol classifier between sender and VNIF can reduce the scope to particular traffic, e.g. UDP, by simple routing decisions. The MP-DCCP takes over responsibility for the multi-path transfer of the traffic, which is directed through the VNIF_in. For possible re-assembly operations, the IP packets may be stamped with a continuously incremented sequence number. This is not mandatory, but assumed required in most seamless handover and capacity aggregation use cases. The path scheduler decides for each IP packet, which DCCP flow it should use for encapsulation, based on a configurable decision logic and supported by the congestion control information of the DCCP flows available for transmission. A DCCP flow selection for a PDU leads to its encapsulation into the respective DCCP flow and adding extra information required for the multipath transmission, e.g. the sequence number. Encapsulation also means, that a DCCP and IP header is added to the original PDU to reach the multi-connectivity end-point. When the encapsulated PDUs arrive at the multi-path termination point, they are re-ordered depending on the carried sequence number and a configurable logic. The re-ordering engine may also include a logic in which packets are just forwarded (no re-ordering). Re-ordering needs to be considered carefully since any active intervention changes the latency responsiveness. The multi-path termination is finally completed when the DCCP overhead is stripped and the PDU leaves VNIF_out. Further routing depends again on the IP layer of the original PDU.

4. Application and placement

The framework of Figure 1 is very flexible in applying multipath support in different architectures and allows MP-DCCP to be applied at any place between sender and receiver. Figure 2 to Figure 5 provide several architectural options for the deployment of the framework.

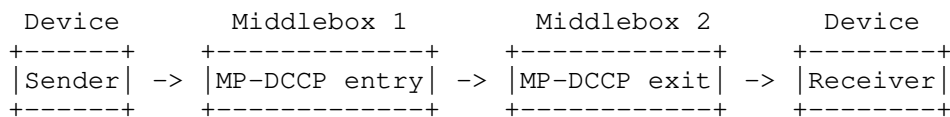


Figure 2: Sender and receiver independent MP-DCCP

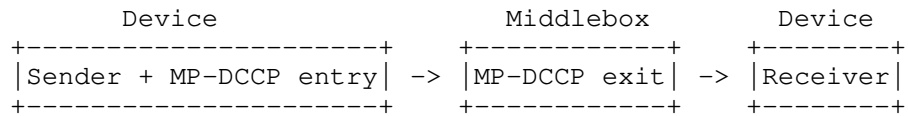


Figure 3: Sender integrated but receiver independent MP-DCCP

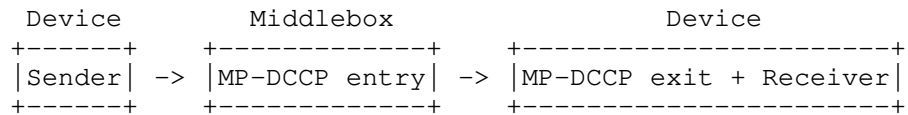


Figure 4: Sender independent and receiver integrated MP-DCCP

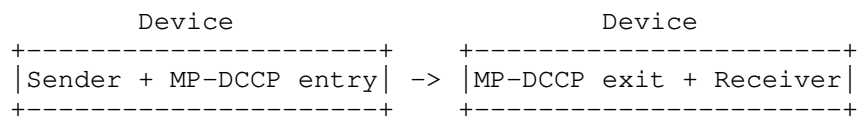


Figure 5: Sender and receiver integrated MP-DCCP

5. Conclusion

The specified IP compatible multipath framework based on MP-DCCP in this document comprises several benefits:

- o Pure routing
- o Inherent path estimation and measurement
- o Imposes no constraints on reliability or in-order delivery of application PDUs
- o Modular re-ordering
- o Modular scheduling
- o IP compatible
- o Based on the standardized DCCP.

Middle-box traversing, when the framework is applied in uncontrolled environments, is addressed in [RFC6733] and [I-D.amend-tsvwg-dccp-udp-header-conversion].

6. Security Considerations

[Tbd]

7. Acknowledgments

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New Spin bit enabled measurements with one or two more bits
draft-cfb-tsvwg-spinbit-new-measurements-00

Abstract

This document introduces additional measurements by using the same spin bit signal as defined in [I-D.trammell-tsvwg-spin] and [I-D.trammell-ippm-spin]. The spin bit signal alone is not enough to evaluate correctly in every network condition the RTT of a flow. In order to solve this problem, it is theorized the possibility of introducing an additional validation signal called delay bit, similar to what is done by the Valid Edge Counter (VEC), but using just one bit instead of two. An alternative with two bits is also introduced with a so called loss bit. More in general a loss signal is defined to measure packet loss and two alternatives are presented with one bit and two bits.

Requirements Language

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

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1. Introduction

Both [I-D.trammell-tsvwg-spin] and [I-D.trammell-ippm-spin] define an explicit per-flow transport-layer signal for hybrid measurement of end-to-end RTT. This signal consists of three bits: a spin bit, which oscillates once per end-to-end RTT, and a two-bit Valid Edge Counter (VEC), which compensates for loss and reordering of the spin bit to increase fidelity of the signal in less than ideal network conditions.

In this document it is introduced the delay bit, that is a single bit signal that can be used together with the spin bit by passive observers to measure the RTT of a network flow, avoiding the spin bit ambiguities that arise as soon as network conditions deteriorate. Unlike the spin bit, which is actually set in every packet transmitted on the network, the delay bit is set only once per round trip.

This document defines a hybrid measurement RFC 7799 [RFC7799] path signal to be embedded into a transport layer protocol, explicitly intended for exposing end-to-end RTT to measurement devices on path.

The document introduces a mechanism applicable to any transport-layer protocol, then explains how to bind the signal to a variety of IETF transport protocols, and in particular to QUIC and TCP.

The application of the Spin bit to QUIC is described in [I-D.ietf-quic-spin-exp] which adds the spin bit only (without the VEC) to QUIC for experimentation purposes.

Note that both the spin bit and the delay bit are inspired by RFC 8321 [RFC8321]. This is also mentioned in [I-D.trammell-quic-spin].

2. Spin bit and Delay bit mechanism

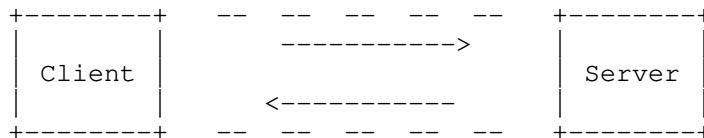
The main idea is to have a single packet, with a second marked bit (the delay bit), that bounces between client and server during the entire connection life. This single packet is called Delay Sample.

A simple observer placed in an intermediate point, tracking the delay sample and the relative timestamp in every spin bit period, can measure the end-to-end round trip delay of the connection. In the same way as seen with the spin bit and the VEC, it is possible to carry out other types of measurements. The next paragraphs give an overview of the observer capabilities.

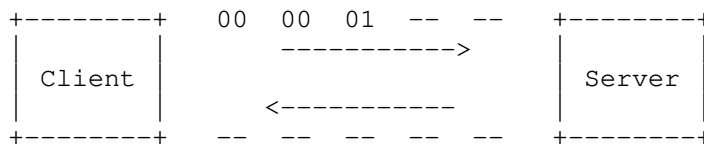
In order to describe the delay sample working mechanism in detail, we have to distinguish two different phases which take part in the delay

bit lifetime: initialization and reflection. The initialization is the generation of the delay sample, while the reflection realizes the bounce behavior of this single packet between the two endpoints.

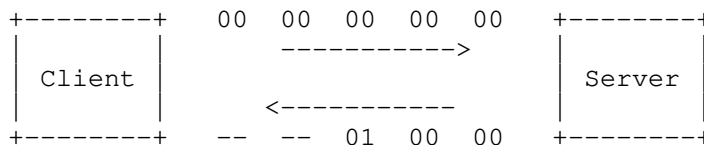
The next figure describes the Delay bit mechanism: the first bit is the spin bit and the second one is the delay bit.



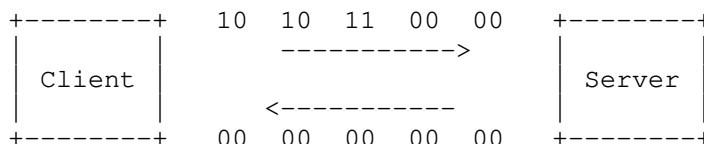
(a) No traffic at beginning.



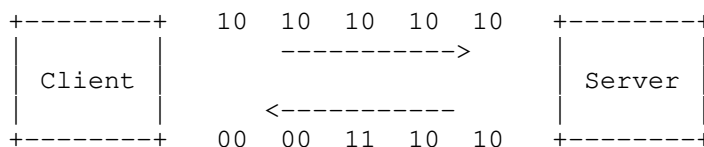
(b) The Client starts sending data and sets the first packet as Delay Sample.



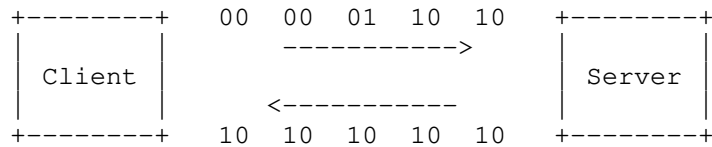
(c) The Server starts sending data and reflects the Delay Sample.



(d) The Client inverts the spin bit and reflects the Delay Sample.



(e) The Server reflects the Delay Sample.



(f) The client reverts the spin bit and reflects the Delay Sample.

Figure 1: Spin bit and Delay bit

2.1. Delay Sample generation

During this first phase, endpoints play different roles. First of all a single delay sample must be bouncing per round trip period (and so per spin bit period). According to that statement and in order to simplify the general algorithm, the delay sample generation is in charge of just one of the two endpoints:

- o the Client, when connection starts and spin bit is set to 0, initializes the delay bit of the first packet to 1, so it becomes the delay sample for that marking period. Only this packet is marked with the delay bit set to 1 for this round trip period; the other ones will carry only the spin bit;
- o the server never initializes the delay bit to 1; its only task is to reflect the incoming delay bit into the next outgoing packet only if certain conditions occur.

Theoretically, in absence of network impairments, the delay sample should bounce between client and server continuously, for the entire duration of the connection. Actually, that is highly unlikely mainly for two different reasons:

- 1) the packet carrying the delay bit might be lost during its journey on the network which is unreliable by definition;
- 2) one of the two endpoints could stop or delay sending data because the application is limiting the amount of traffic transmitted;

To deal with these problems, the algorithm provides a procedure to regenerate the delay sample and to inform a possible observer that a problem has occurred, and then the measurement has to be restarted.

2.1.1. The recovery process

In order to relieve the server from tasks that go beyond the mere reflection of the sample, even in this case the recovery process belongs to the client. A fundamental assumption is that a delay sample is strictly related to its spin bit period. Considering this rule, the client verifies that every spin bit period ends with its delay sample. If that does not happen and a marking period terminates without a delay sample, the client waits a further empty period; then, in the following period, it reinitializes the mechanism by setting the delay bit of the first outgoing packet to 1, making it the new delay sample. The empty period is needed to inform the intermediate points that there was an issue and a new delay measurement session is starting.

2.2. Delay Sample reflection

The reflection is the process that enables the bouncing of the delay sample between client and server. The behavior of the two endpoints is slightly different. With the exception of the client that, as previously exposed, generates a new delay sample, by default the delay bit is set to 0.

Server side reflection: when a packet with the delay bit set to 1 arrives, the server marks the first packet in the opposite direction as the delay sample, if it has the same spin bit value. While if it has the opposite spin bit value this sample is considered lost.

Client side reflection: when a packet with delay bit set to 1 arrives, the client marks the first packet in the opposite direction as the delay sample, if it has the opposite spin bit value. While if it has the same spin bit value this sample is considered lost.

In both cases, if the outgoing marked packet is transmitted with a delay greater than a predetermined threshold after the reception of the incoming delay sample (lms by default), reflection is aborted and this sample is considered lost.

It is noteworthy that differently from what happens with the VEC for which the reflection always concerns the edge of the period, in this case reflection takes place for the packet that is carrying the delay bit regardless of its position within the period. For this reason it is necessary to introduce that condition of validation in order to identify and discard those samples that, due to reordering, might move to a contiguous period. Furthermore, by introducing a threshold for the retransmission delay of the sample, it is possible to eliminate all those measurements which, due to lack of traffic on the endpoints, would be overestimated and not true. Thus, the maximum

estimation error, without considering any other delays due to flow control, would amount to twice the threshold (e.g. 2ms) per measurement, in the worst case.

3. Using the Spin bit and Delay bit for Hybrid RTT Measurement

Unlike what happens with the spin bit for which it is necessary to validate or at least heuristically evaluate the goodness of an edge, the delay sample can be used by an intermediate observer as a simple demarcator between a period and the following one eliminating the ambiguities on the calculation of the RTT found with the analysis of the spin-bit only. The measurement types, that can be done from the observation of the delay sample, are exactly the same achievable with the spin bit only (with or without the VEC).

3.1. End-to-end RTT measurement

The delay sample generation process ensures that only one packet marked with the delay bit set to 1 runs back and forth on the wire between two endpoints per round trip time. Therefore, in order to determine the end-to-end RTT measurement of a QUIC flow, an on-path passive observer can simply compute the time difference between two delay samples observed in a single direction. Note that a measurement, to be valid, must take into account the difference in time between the timestamps of two consecutive delay samples belonging to adjacent spin-bit periods. For this reason, an observer, in addition to intercepting and analyzing the packets containing the delay bit set to 1, must maintain awareness of each spin period in such a way as to be able to assign each delay sample to its period and, at the same time, identifying those periods that do not contain it.

3.2. Half-RTT measurement

An on-path passive observer that is sniffing traffic in both directions -- from client to server and from server to client -- can also use the delay sample to measure "upstream" and "downstream" RTT components. Also known as the half-RTT measurement, it represents the components of the end-to-end RTT concerning the paths between the client and the observer (upstream), and the observer and the server (downstream). It does this by measuring the delay between a delay sample observed in the downstream direction and the one observed in the upstream direction, and vice versa. Also in this case, it should verify that the two delay samples belong to two adjacent periods, for the upstream component, or to the same period for the downstream component.

3.3. Intra-domain RTT measurement

Taking advantage of the half-RTT measurements it is also possible to calculate the intra-domain RTT which is the portion of the entire RTT used by a QUIC flow to traverse the network of a provider (or part of it). To achieve this result two observers, able to watch traffic in both directions, must be employed simultaneously at ingress and egress of the network to be measured. At this point, to determine the delay between the two observers, it is enough to subtract the two computed upstream (or downstream) RTT components.

The spin bit is an alternate marking generated signal and the only difference than RFC 8321 [RFC8321] is the size of the alternation that will change with the flight size each RTT. So it can be useful to segment the RTT and deduce the contribution to the RTT of the portion of the network between two on-path observers and it can be easily performed by calculating the delay between two or more measurement points on a single direction by applying RFC 8321 [RFC8321].

4. Observer's algorithm and Waiting Interval

Given below is a formal summary of the functioning of the observer every time a delay sample is detected. A packet containing the delay bit set to 1:

- o if it has the same spin bit value of the current period and no delay sample was detected in the previous period, then it can be used as a left edge (i.e. to start measuring an RTT sample), but not as a right edge (i.e. to complete an RTT measurement since the last edge). If the observation point is symmetric (i.e. it can see both upstream and downstream packets in the flow) and in the current period a delay sample was detected in the opposite direction (i.e. in the upstream direction), the packet can also be used to compute the downstream RTT component.
- o if it has the same spin bit value of the current period and a delay sample was detected in the previous period, then it can be used at the same time as a left or right edge, and to compute RTT component in both directions.

Like stated previously, every time an empty period is detected, the observer must restart the measurement process and consider the next delay sample that will come as the beginning of a new measure, then as a left edge. As a result, being able to assign the delay sample to the corresponding spin period becomes a crucial factor for the proper functioning of the entire algorithm.

Considering that the division into periods is realized by exploiting the spin bit square wave, it is easy to understand that the presence of spurious spin edges -- caused by packet reordering -- would inevitably lead the observer to overestimate the amount of periods actually present in the transmission. This results in a greater number of empty periods detected and the consequent decrease of the actual RTT samples achievable. Therefore, in order to maximize the performance of the whole algorithm, the observer must implement a mechanism to filter out spurious spin edges.

To face this problem the waiting interval has to be introduced. Basically, every time a spin bit edge is detected, the observer sets a time interval during which it rejects every potential spurious edges observed on the wire. While, at the end of the interval it starts again to accept changes in the spin bit value. This guarantees a proper protection against the spurious edges in relation to the size of the interval itself. For instance, an interval of 5ms is able to filter out edges that have been reordered by a maximum of 5ms. Clearly, the mechanism does its job for intervals smaller than the RTT of the observed connection (if RTT is smaller than the waiting interval the observer can't measure the RTT).

5. Adding a Loss signal for Packet loss measurement

It is possible to introduce a mechanism to evaluate also the packet loss together with the delay measurement. This can be achieved by introducing the loss signal, a single or two bits signal whose purpose is to mark a variable number of packets (from live traffic) which are exchanged two times between the endpoints realizing a two round-trip reflection. The overall exchange comprises:

- o The client first selects, generates and consequent transmits to the server a first train of packets, by marking the loss bit to 1;
- o The server, upon reception from the client of each one of the packets included in the first train, reflects to the client a respective second train of packets of the same size as the first train received, by marking the loss bit to 1;
- o The client, upon reception from the server of each one of the packets included in the second train, reflects to the server a respective third train of packets of the same size as the second train received, by marking the loss bit to 1;
- o The server, upon reception from the client of each one of the packets included in the third train, finally reflects to the client a respective fourth train of packets of the same size as the third train received, by marking the loss bit to 1.

Packets belonging to the first round (first and second train) represent the Generation Phase while those belonging to the second round (third and fourth train) represent the Reflection Phase.

A passive on-path observer, placed on whatever direction, can trivially count and compare the number of marked packets seen during the two mentioned phases (i.e. the first and third or the second and the fourth trains of packets, depending on which direction is observed) and estimate the loss rate experienced by the connection. This process is repeated continuously to obtain more measurements as long as the endpoints exchange traffic. These measurements can be called Round Trip (RT) losses

The general algorithm shown above gives an idea of its underlying principles but is not enough to make the whole process working properly.

Firstly, there is the issue that packet rates in the two directions may be different. Therefore, the right number of packets to be marked has to be chosen in order to avoid their congestion on the slowest traffic direction. As a consequence, this number is inevitably equal to the amount of packets transited, indeed, on the slowest direction. This problem can be easily addressed by a method wherein the two endpoints of a communication exchange marked packets interleaved with unmarked packets. From an implementation point of view, this result can be achieved by introducing a single token system that adjusts the number of outgoing marked packets. Basically, the token is enabled every time a packet arrives and disabled when a marked packet is transmitted. Since the creation of the initial train of marked packets is carried out by the client, the management and use of this single token is also assigned to it, which in fact "calculates" the correct number of packets to be marked each time.

Secondly, a mechanism to individually identify each train of packets must be provided to enable the observer to distinguish between trains belonging to different phases (Generation and Reflection). About this point, different approaches are used depending on the number of bits of the loss signal and it will be discussed in the next sections.

5.1. Round Trip Packet Loss measurement

Since the measurements are performed on a portion of the traffic exchanged between client and server, the observer calculates the end-to-end Round Trip Packet Loss that, statistically, will be equal to the loss rate experienced by the connection along the entire network

path. So this measurement can be simply referred as the Round Trip Packet Loss (RTPL).

In addition, this methodology allows the Half-RTPL measurement and the Intra-domain RTPL measurement, in the same way as described in the previous sections for RTT measurement.

6. RTT dependent Packet Loss using one bit loss signal

The single bit loss signal is implemented using just one bit: marked packets have this bit set to 1, whereas unmarked ones have it set to 0. This solution requires a working spin-bit signal used to separate different trains of packets. In particular, a "pause" of at least one empty spin-bit period is introduced between each phase of the algorithm. An on-path observer can determine in this way if a phase (and therefore a train of packets) is ended and a new one is starting.

The client is in charge of almost the entire complexity of the algorithm. Its task can be summarized in 4 different points:

1. The client starts generating marked packets for two consecutive spin-bit periods; it maintains a generation token that is enabled every time a packet arrives and disabled when another one is forwarded. When this token is disabled, the generation process is paused (i.e. outgoing packets are transmitted unmarked) and resumes as soon as its value returns true, and that happens as soon as a packet is received. In addition, at the end of the first spin-bit period spent in generation, the reflection counter is unlocked to start counting incoming marked packets which will be later reflected;
2. When the generation is completed, the client waits to see in input an empty spin-bit period so as to be sure that everyone has seen at least that empty period. This one will be used by the observer as a divider between generated and reflected packets. During this phase, all the outgoing packets are forwarded with the loss bit set to 0. The reflection counter is still incremented every time a marked packet arrives;
3. The client starts reflecting marked packets until the reflection counter is zeroed; the generation token is also used (in the same way) during this phase to avoid congestion on the slowest traffic direction. In addition, at the end of the first spin-period spent in reflection, the reflection counter is locked to avoid incoming reflected packets incrementing it;

4. When the reflection is completed, the client waits to see in input an empty spin-bit period so as to be sure that everyone has seen at least that empty period. This one will be used by the observer as a divider between reflected and newly generated packets. During this phase, all the outgoing packets are forwarded with the loss bit set to 0. The whole process restarts going back to the first point.

As previously anticipated, the server simply reflects each incoming marked packet sent by the client. It maintains a simple counter that is incremented every time a marked packet arrives and decremented when a marked one is sent in the opposite direction.

This one bit loss signal methodology replies and exposes the RTT of the connection on the wire in any case, when the spin bit and the delay bit are used and when these are disabled.

6.1. Observer's logic for one bit loss signal

The on-path observer, placed in any direction, counts marked packets and separates different trains detecting empty spin-bit periods between them (one or more). Then, it simply computes the difference between a Generation train and a Reflection train to produce a statistical measurement of the Round Trip Packet Loss (RTPL) and of the connection end-to-end loss rate.

Here is an example. Packets are represented by two digits (first one is the spin bit, second one is the loss bit):

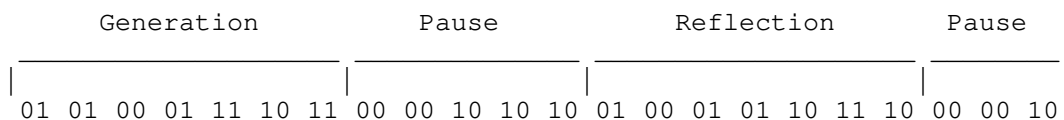


Figure 2: one bit loss signal example

Note that 5 marked packets have been generated of which 4 reflected.

7. RTT independent Packet Loss using two bits loss signal

An RTT independent version of this algorithm requires two bits and does not rely on the spin-bit signal to enable pause detection. That is because packets generated and reflected by the client are marked using two different marking values thus removing the need of introducing a pause between them. Furthermore, instead of generating marked packets for the duration of two spin-bit periods (as seen in

the one bit loss signal), a fixed duration for the generation phase can be used (e.g., 100ms).

In this way, no information related to the RTT of the connection is exposed on wire.

Using a two bits loss signal, four possible values can be used inside each packet (i.e. 0 to 3). During the Generation phase, marked packets have the loss value set to 1 whereas unmarked ones to 0. On the contrary, during the Reflection phase, marked packets have the loss value set to 2 whereas unmarked ones to 3. By doing so, even unmarked packets have their own alternate marking methodology that can be used by intermediate points to compute the one-way loss rate between them (RFC 8321 [RFC8321]).

Even in this case, the client is in charge of almost the entire complexity of the algorithm. Its task can be summarized in 2 different points:

1. The client generates marked packets (i.e. with loss bits set to 1) for 100ms; it also maintains a generation token that is enabled every time a packet arrives and disabled when another one is forwarded. When this token is disabled, the generation process is paused (i.e. outgoing packets are transmitted unmarked with the loss bits set to 0) and resumes as soon as its value returns true.
2. When the generation is completed, the client starts reflecting marked packets (i.e. with loss bits set to 2) until the reflection counter is zeroed and for at least 100ms. The generation token is also used during this phase to avoid congestion on the slowest traffic direction; however, in this case, "unmarked" packets are transmitted with the loss bit set to 3. The whole process restarts going back to the first point.

Independently of the current phase of the algorithm, the reflection counter is increased every time a packet carrying a loss value equal to 1 arrives. Moreover, depending on the connection RTT, the client should vary the duration of the generation phase to different values. For example, for connection below 100ms of RTT the client generates for 100ms; for connection below 300ms of RTT it generates for 300ms and for connection below 1s of RTT it generates for 1000ms. This is necessary to ensure that the client has already received generated marked packets before the beginning of the reflection phase.

As regards the role of the server, it simply reflects each incoming marked packet sent by the client. It maintains two different counters for generated and reflected packets (i.e. loss bits to 1 and

2) in concomitance with a mechanism to reflect in output the same number of marked packets in the same order of arrival (with at most the reordering of packets arrived out of sequence).

7.1. Observer's logic for two bits loss signal

The on-path observer, placed in any direction, counts marked packets belonging to different phases simply looking at the loss value carried by each packet (therefore, it does not look at the spin-bit value anymore). Then, in the same way seen for the previous one bit algorithm, it simply computes the difference between a Generation train and a Reflection train to produce a statistical measurement of the Round Trip Packet Loss (RTPL) and of the connection end-to-end loss rate. Moreover, it can also count unmarked packets and, cooperating with a second observer placed in the same direction, compute the one-way loss rate between two intermediate points using the alternate marking methodology (RFC 8321 [RFC8321]).

Here is an example. Packets are represented by a single digit corresponding to the carried two-bits loss value (0 to 3):

Generation	Reflection	Generation
1 1 0 1 1 1 1 0 1 1 0	2 2 2 2 3 2 3 3 2 3 3	1 1 0 1 0 0 1 1 0 1 0

Figure 3: two bits loss signal example

Note that 8 marked packets have been generated of which 6 reflected; then again 6 marked packets are generated.

8. Protocols

8.1. QUIC

The binding of the delay bit signal to QUIC is partially described in [I-D.ietf-quic-spin-exp], which adds the spin bit only to the QUIC protocol. From an implementation point of view, the delay bit is placed in the partially unencrypted (but authenticated) QUIC header, alongside the spin bit, occupying one of the two bits left reserved for future experiments. As things stand, according to [I-D.ietf-quic-transport], the proposed scheme of the first header's byte would be 01SDRKPP.

Regarding the loss signal, since the use of the spin bit is not mandatory and many connection may not have it spinning, two different configuration are proposed:

If the spin-bit IS enabled (i.e. the RTT is already exposed on wire), use the 1 bit loss signal alongside the delay bit to improve delay measurements accuracy; in this configuration, the proposed scheme of the first header's byte would be 01SDLKPP;

If the spin-bit IS NOT enabled, use the 2 bits loss signal just to measure connection loss rate without exposing any RTT related information on wire; in this configuration, the proposed scheme of the first header's byte would be 01SLLKPP.

This implies that an observer must be able to determine whether the spin bit is active and correctly spinning or not (choosing, accordingly, the right version of packet loss measurement to be used).

8.2. TCP

The signal can be added to TCP by defining bit 4 of bytes 13-14 of the TCP header to carry the spin bit, and eventually bits 5 and 6 to carry additional information, like the delay bit and the 1 bit loss signal (or the two bits loss signal).

9. Security Considerations

The privacy considerations for the hybrid RTT measurement signal are essentially the same as those for passive RTT measurement in general.

10. Acknowledgements

tbc

11. IANA Considerations

tbc

12. References

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Guidelines for Internet Congestion Control at Endpoints
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Abstract

This document is for discussion by the TSVWG. It provides guidance on the design of methods to avoid congestion collapse and to provide congestion control. Recommendations and requirements on this topic are distributed across many documents in the RFC series. This therefore seeks to gather and consolidate these recommendations in an annexe. Based on these specifications, and Internet engineering experience, the document provides input to the design of new congestion control methods in protocols.

The present document is for discussion and comment by the IETF. If published, it plans to update or replace the Best Current Practice in BCP 41, which currently includes "Congestion Control Principles" provided in RFC2914.

Status of This Memo

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

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1. Introduction

The IETF has specified Internet transports (e.g., TCP [I-D.ietf-tcpm-rfc793bis], UDP [RFC0768], UDP-Lite [RFC3828], SCTP [RFC4960], and DCCP [RFC4340]) as well as protocols layered on top of these transports (e.g., RTP [RFC3550], QUIC [I-D.ietf-quic-transport], SCTP/UDP [RFC6951], DCCP/UDP [RFC6773]) and transports that work directly over the IP network layer. These

transports are implemented in endpoints (either Internet hosts or routers acting as endpoints), and are designed to detect and react to network congestion. TCP was the first transport to provide this, although the TCP specifications found in RFC 793 predates the inclusion of congestion control and did not contain any discussion of using or managing a congestion window. RFC 793.bis [I-D.ietf-tcpm-rfc793bis] seek to address this.

Recommendations and requirements on this topic are distributed across many documents in the RFC series. The appendix of this document therefore seeks to gather and consolidate these recommendations. This, and Internet engineering experience are used as a basis to provide overall guidelines as input to the design of congestion control methods that are implemented in Internet protocols. The focus of the present document is upon unicast point-to-point transports, this includes migration from using one path to another path.

Some recommendations [RFC5783] and requirements in this document apply to point-to-multipoint transports (e.g., multicast), however this topic extends beyond the current document's scope. [RFC2914] provides additional guidance on the use of multicast.

2. Terminology

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

The path between endpoints (sometimes called "Internet Hosts" or called source and destination nodes in IPv6) consists of the endpoint protocol stack at the sender and the receiver (which together implement the transport service), and a succession of links and network devices (routers or middleboxes) that provide connectivity across the network. The set of network devices forming the path is not usually fixed, and it should generally be assumed that this set can change over arbitrary lengths of time.

[RFC5783] defines congestion control as "the feedback-based adjustment of the rate at which data is sent into the network. Congestion control is an indispensable set of principles and mechanisms for maintaining the stability of the Internet." [RFC5783] also provides an informational snapshot taken by the IRTF's Internet Congestion Control Research Group (ICCRG) from October 2008.

The text draws on language used in the specifications of TCP and other IETF transports. For example, a protocol timer is generally needed to detect persistent congestion, and this document uses the

term Retransmission Timeout (RTO) to refer to the operation of this timer. Similarly, the document refers to a congestion window as the variable that controls the rate of transmission by the congestion controller. The use of these terms does not imply that endpoints need to implement functions in the way that TCP currently does. Each new transport needs to make its own design decisions about how to meet the recommendations and requirements for congestion control.

Other terminology is directly copied from the cited RFCs.

3. Principles of Congestion Control

This section summarises the principles for providing congestion control.

3.1. Internet Requirements

Principles include:

- o Endpoints **MUST** perform congestion control [RFC1122] and **SHOULD** leverage existing congestion control techniques [RFC8085].
- o If an application or protocol chooses not to use a congestion-controlled transport protocol, it **SHOULD** control the rate at which it sends datagrams to a destination host, in order to fulfil the requirements of [RFC2914], as stated in [RFC8085].
- o Transports **SHOULD** control the aggregate traffic they send on a path [RFC8085]. They ought not to use multiple congestion-controlled flows between the same endpoints to gain a performance advantage. An endpoint can become aware of congestion by various means (including, delay variation, timeout, ECN, packet loss). A signal that indicates congestion on the end-to-end network path, **SHOULD** result in a congestion control reaction by the transport that reduces the current rate of the sending endpoint [RFC8087]).
- o Although network devices can be configured to reduce the impact of flow multiplexing on other flows, endpoints **MUST NOT** rely solely on the presence and correct configuration of these methods, except when constrained to operate in a controlled environment. Transports that do not target Internet deployment need to be constrained to only operate in a controlled environment (e.g., see Section 3.6 of [RFC8085]) and provide appropriate mechanisms to prevent traffic accidentally leaving the controlled environment [RFC8084].

3.1.1. Tolerance to a Diversity of Path Characteristics

Principles include:

- o A transport design is REQUIRED be robust to a change in the set of devices forming the network path. A reconfiguration, reset or other event could interrupt this path or trigger a change in the set of network devices forming the path.
- o Transports are REQUIRED to operate safely over the wide range of path characteristics presented by Internet paths.
- o Path characteristics can change over relatively short intervals of time (i.e., characteristics discovered do not necessarily remain valid for multiple Round Trip Times, RTTs). In particular, the transport SHOULD measure and adapt to the characteristics of the path(s) being used.

3.2. Avoiding Congestion Collapse and Flow Starvation

Principles include:

- o Transports MUST avoid inducing flow starvation to other flows that share resources along the path they use.
- o Endpoints MUST treat a loss of all feedback (e.g., expiry of a retransmission time out, RTO) as an indication of persistent congestion (i.e., an indication of potential congestion collapse).
- o When an endpoint detects persistent congestion, it MUST reduce the maximum rate (e.g., reduce its congestion window). This normally involves the use of protocol timers to detect a lack of acknowledgment for transmitted data (Section 3.5).
- o Protocol timers (e.g., used for retransmission or to detect persistent congestion) need to be appropriately initialised. A transport SHOULD adapt its protocol timers to follow the measured the path Round Trip Time (RTT) (e.g., Section 3.1.1 of [RFC8085]).
- o A transport MUST employ exponential backoff each time persistent congestion is detected [RFC1122], until the path characteristics can again be confirmed.
- o Network devices MAY provide mechanisms to mitigate the impact of congestion collapse by transport flows (e.g., priority forwarding of control information, and starvation detection), and SHOULD mitigate the impact of non-conformant and malicious flows

[RFC7567])). These mechanisms complement, but do not replace, the endpoint congestion avoidance mechanisms.

3.3. Connection Initialization

When a connection or flow to a new destination is established, the endpoints have little information about the characteristics of the network path they will use. This section describes how a flow starts transmission over such a path.

Flow Start: A new flow between two endpoints needs to initialise a congestion controller for the path it will use. It **MUST NOT** assume that capacity is available at the start of the flow, unless it uses a mechanism to explicitly reserve capacity. In the absence of a capacity signal, a flow might therefore start slowly. The TCP slow-start algorithm is an accepted standard for flow startup [RFC5681]. TCP uses the notion of an Initial Window (IW) [RFC3390], updated by [RFC6928] to define the initial volume of data that can be sent on a path. This is not the smallest burst, or the smallest window, but it is considered a safe starting point for a path that is not suffering persistent congestion, and is applicable until feedback about the path is received. The initial sending rate (e.g., determined by the IW) needs to be viewed as tentative until the capacity is confirmed to be available.

Initial RTO Interval: When a flow sends the first packet(s), it typically has no way to know the actual RTT of the path it will use. An initial value needs to be used to initialise the principal retransmission timer, which will be used to detect lack of responsiveness from the remote endpoint. In TCP, this is the starting value of the RTO. The selection of a safe initial value is a trade off that has important consequences on the overall Internet stability [RFC6928] [RFC8085]. In the absence of any knowledge about the latency of a path (including the initial value), the RTO **MUST** be conservatively set to no less than 1 second. Values shorter than 1 second can be problematic (see the appendix of [RFC6298]). (Note: Linux TCP has deployed a smaller initial RTO value).

[[Author note: It could be useful to discuss cached values]].

Initial RTO Expiry: If the RTO timer expires while awaiting completion of a connection setup, or handshake (e.g., the three-way handshake in TCP, the ACK of a SYN segment), and the implementation is using an RTO less than 3 seconds, the local endpoint can resend the connection setup. [[Author note: It would be useful to discuss how the timer is managed to protect from multiple handshake failure]].

The RTO MUST then be re-initialized to increase it to 3 seconds when data transmission begins (i.e., after the handshake completes) [RFC6298] [RFC8085]. This conservative increase is necessary to avoid congestion collapse when many flows retransmit across a shared bottleneck with restricted capacity.

Initial Measured RTO: Once an RTT measurement is available (e.g., through reception of an acknowledgement), the timeout value must be adjusted. This adjustment MUST take into account the RTT variance. For the first sample, this variance cannot be determined, and a local endpoint MUST therefore initialise the variance to $RTT/2$ (see equation 2.2 of [RFC6928] and related text for UDP in section 3.1.1 of [RFC8085]).

Current State: A congestion controller MAY assume that recently used capacity between a pair of endpoints is an indication of future capacity available in the next RTT between the same endpoints. It MUST react (reduce its rate) if this is not (later) confirmed to be true. [[Author note: do we need to bound this]].

Cached State: A congestion controller that recently used a specific path could use additional state that lets a flow take-over the capacity that was previously consumed by another flow (e.g., in the last RTT) which it understands is using the same path and no will longer use the capacity it recently used. In TCP, this mechanism is referred to as TCP Control Block (TCB) sharing [RFC2140] [I-D.ietf-tcpm-2140bis]. The capacity and other information can be used to suggest a faster initial sending rate.

Any information used to accelerate the growth of the cwnd MUST be viewed as tentative until the path capacity is confirmed by receiving a confirmation that actual traffic has been sent across the path. (i.e., the new flow needs to either use or loose the capacity that has been tentatively offered to it). A sender MUST reduce its rate if this capacity is not confirmed within the current RTO interval.

3.4. Using Path Capacity

This section describes how a sender needs to regulate the maximum volume of data in flight over the interval of the current RTT, and how it manages transmission of the capacity that it perceives is available.

Transient Path: Unless managed by a resource reservation protocol, path capacity information is transient. A sender that does not use capacity has no understanding whether previously used capacity remains available to use, or whether that capacity has disappeared

(e.g., a change in the path that causes a flow to experience a smaller bottleneck, or when more traffic emerges that consumes previously available capacity resulting in a new bottleneck). For this reason, a transport that is limited by the volume of data available to send MUST NOT continue to grow its congestion window when the current congestion window is more than twice the volume of data acknowledged in the last RTT.

Validating the congestion window Standard TCP states that a TCP sender "SHOULD set the congestion window to no more than the Restart Window (R)" before beginning transmission, if the sender has not sent data in an interval that exceeds the current retransmission timeout, i.e., when an application becomes idle [RFC5681]. An experimental specification [RFC7661] permits TCP senders to tentatively maintain a congestion window larger than the path supported in the last RTT when application-limited, provided that they appropriately and rapidly collapse the congestion window when potential congestion is detected. This mechanism is called Congestion Window Validation (CWV).

Collateral Damage: Even in the absence of congestion, statistical multiplexing of flows can result in transient effects for flows sharing common resources. A sender therefore SHOULD avoid inducing excessive congestion to other flows (collateral damage).

Burst Mitigation: While a congestion controller ought to limit sending at the granularity of the current RTT, this can be insufficient to satisfy the goals of preventing starvation and mitigating collateral damage. This requires moderating the burst rate of the sender to avoid significant periods where a flow(s) consume all buffer capacity at the path bottleneck, which would otherwise prevent other flows from gaining a reasonable share. Endpoints SHOULD provide mechanisms to regulate the bursts of transmission that the application/protocol sends to the network (section 3.1.6 of [RFC8085]). ACK-Clocking [RFC5681] can help mitigate bursts for protocols that receive continuous feedback of reception (such as TCP). Sender pacing can mitigate this [RFC8085], (See Section 4.6 of [RFC3449]), and has been recommended for TCP in conditions where ACK-Clocking is not effective, (e.g., [RFC3742], [RFC7661]). SCTP [RFC4960] defines a maximum burst length (Max.Burst) with a recommended value of 4 segments to limit the SCTP burst size.

3.5. Timers and Retransmission

This section describes mechanisms to detect and provide retransmission, and to protect the network in the absence of timely feedback.

Loss Detection: Loss detection occurs after a sender determines there is no delivery confirmation within an expected period of time (e.g., by observing the time-ordering of the reception of ACKs, as in TCP DupACK) or by utilising a timer to detect loss (e.g., a transmission timer with a period less than the RTO, [RFC8085] [I-D.ietf-tcpm-rack]) or a combination of using a timer and ordering information to trigger retransmission of data.

Retransmission: Retransmission of lost packets or messages is a common reliability mechanism. When loss is detected, the sender can choose to retransmit the lost data, ignore the loss, or send other data (e.g., [RFC8085] [I-D.ietf-quic-recovery]), depending on the reliability model provided by the transport service. Any transmission consumes network capacity, therefore retransmissions MUST NOT increase the network load in response to congestion loss (which worsens that congestion) [RFC8085]. Any method that sends additional data following loss is therefore responsible for congestion control of the retransmissions (and any other packets sent, including FEC information) as well as the original traffic.

Measuring the RTT: Once an endpoint has started communicating with its peer, the RTT be MUST adjusted by measuring the actual path RTT. This adjustment MUST include adapting to the measured RTT variance (see equation 2.3 of [RFC6928]).

Maintaining the RTO: The RTO SHOULD be set based on recent RTT observations (including the RTT variance) [RFC8085].

RTO Expiry: Persistent lack of feedback (e.g., detected by an RTO timer, or other means) MUST be treated an indication of potential congestion collapse. A failure to receive any specific response within a RTO interval could potentially be a result of a RTT change, change of path, excessive loss, or even congestion collapse. If there is no response within the RTO interval, TCP collapses the congestion window to one segment [RFC5681]. Other transports MUST similarly respond when they detect loss of feedback.

An endpoint needs to exponentially backoff the RTO interval [RFC8085] each time the RTO expires. That is, the RTO interval MUST be set to at least the $RTO * 2$ [RFC6298] [RFC8085].

Maximum RTO: A maximum value MAY be placed on the RTO interval. This maximum limit to the RTO interval MUST NOT be less than 60 seconds [RFC6298].

[[Author Note: Check RTO-Consider.]]

3.6. Responding to Potential Congestion

The safety and responsiveness of new proposals need to be evaluated [RFC5166]. In determining an appropriate congestion response, designs could take into consideration the size of the packets that experience congestion [RFC4828].

Congestion Response: An endpoint **MUST** promptly reduce the rate of transmission when it receive or detects an indication of congestion (e.g., loss) [RFC2914].

TCP Reno established a method that relies on multiplicative-decrease to halve the sending rate while congestion is detected. This response to congestion indications is considered sufficient for safe Internet operation, but other decrease factors have also been published in the RFC Series [RFC8312].

ECN Response: A congestion control design should provide the necessary mechanisms to support Explicit Congestion Notification (ECN) [RFC3168] [RFC6679], as described in section 3.1.7 of [RFC8085]. This can help determine an appropriate congestion window when supported by routers on the path [RFC7567] to enable rapid early indication of incipient congestion.

The early detection of incipient congestion justifies a different reaction to an explicit congestion signal compared to the reaction to detected packet loss [RFC8311] [RFC8087]. Simple feedback of received Congestion Experienced (CE) marks [RFC3168], relies only on an indication that congestion has been experienced within the last RTT. This style of response is appropriate when a flow uses ECT(0). The reaction to reception of this indication was modified in TCP ABE [RFC8511]. Further detail about the received CE-marking can be obtained by using more accurate receiver feedback (e.g., [I-D.ietf-tcpm-accurate-ecn] and extended RTP feedback). The more detailed feedback provides an opportunity for a finer-granularity of congestion response.

Current work-in-progress [I-D.ietf-tsvwg-l4s-arch] defines a reaction for packets marked with ECT(1), building on the style of detailed feedback provided by [I-D.ietf-tcpm-accurate-ecn] and a modified marking system [I-D.ietf-tsvwg-aqm-dualq-coupled].

Robustness to Path Change: The detection of congestion and the resulting reduction **MUST NOT** solely depend upon reception of a signal from the remote endpoint, because congestion indications could themselves be lost under persistent congestion.

The only way to surely confirm that a sending endpoint has successfully communicated with a remote endpoint is to utilise a timer (seeSection 3.5) to detect a lack of response that could result from a change in the path or the path characteristics (usually called the RTO). Congestion controllers that are unable to react after one (or at most a few) RTTs after receiving a congestion indication should observe the guidance in section 3.3 of the UDP Guidelines [RFC8085].

Persistent Congestion: Persistent congestion can result in congestion collapse, which **MUST** be aggressively avoided [RFC2914]. Endpoints that experience persistent congestion and have already exponentially reduced their congestion window to the restart window (e.g., one packet), **MUST** further reduce the rate if the RTO timer continues to expire. For example, TFRC [RFC5348] continues to reduce its sending rate under persistent congestion to one packet per RT, and then exponentially backs off the time between single packet transmissions if the congestion continues to persist [RFC2914].

[RFC8085] provides guidelines for a sender that does not, or is unable to, adapt the congestion window.

3.7. Using More Capacity

In the absence of persistent congestion, an endpoint **MAY** increase its congestion window and hence the sending rate. An increase should only occur when there is additional data available to send across the path (i.e., the sender will utilise the additional capacity in the next RTT).

Increasing Congestion Window: A sender **MUST NOT** continue to increase its rate for more than an RTT after a congestion indication is received. The transport **SHOULD** stop increasing its congestion window as soon as it receives indication of congestion.

While the sender is increasing the congestion window, a sender will transmit faster than the last confirmed safe rate. Any increase above the last confirmed rate needs to be regarded as tentative and the sender reduce their rate below the last confirmed safe rate when congestion is experienced (a congestion event).

Congestion: An endpoint **MUST** utilise a method that assures the sender will keep the rate below the previously confirmed safe rate for multiple RTT periods after an observed congestion event. In TCP, this is performed by using a linear increase from a slow

start threshold that is re-initialised when congestion is experienced.

Avoiding Overshoot: Overshoot of the congestion window beyond the point of congestion can significantly impact other flows sharing resources along a path. It is important to note that as endpoints experience more paths with a large BDP and a wider range of potential path RTT, that variability or changes in the path can have very significant constraints on appropriate dynamics for increasing the congestion window (see also burst mitigation, Section 3.4).

3.8. Network Signals

An endpoint can utilise signals from the network to help determine how to regulate the traffic it sends.

Network Signals: Mechanisms **MUST NOT** solely rely on transport messages or specific signalling messages to perform safely. (See section 5.2 of [RFC8085] describing use of ICMP messages). They need to be designed so that they safely operate when path characteristics change at any time. Transport mechanisms **MUST** be robust to potential black-holing of any signals (i.e., need to be robust to loss or modification of packets, noting that this can occur even after successful first use of a signal by a flow, as occurs when the path changes, see Section 3.1.1).

A mechanism that utilises signals originating in the network (e.g., RSVP, NSIS, Quick-Start, ECN), **MUST** assume that the set of network devices on the path can change. This motivates the use of soft-state when designing protocols that interact with signals originating from network devices [I-D.irtf-panrg-what-not-to-do] (e.g., ECN). This can include context-sensitive treatment of "soft" signals provided to the endpoint [RFC5164].

3.9. Protection of Protocol Mechanisms

An endpoint needs to provide protection from attacks on the traffic it generates, or attacks that seek to increase the capacity it consumes (impacting other traffic that shared a bottleneck).

Off Path Attack: A design **MUST** protect from off-path attack to the protocol [RFC8085] (i.e., one by an attacker that is unable to see the contents of packets exchanged across the path). An attack on the congestion control can lead to a Denial of Service (DoS) vulnerability for the flow being controlled and/or other flows that share network resources along the path.

Validation of Signals: Network signalling and control messages (e.g., ICMP [RFC0792]) MUST be validated before they are used to protect from malicious abuse. This MUST at least include protection from off-path attack [RFC8085].

On Path Attack: A protocol can be designed to protect from on-path attacks, but this requires more complexity and the use of encryption/authentication mechanisms (e.g., IPsec [RFC4301], QUIC [I-D.ietf-quic-transport]).

4. IETF Guidelines on Evaluation of Congestion Control

The IETF has provided guidance [RFC5033] for considering alternate congestion control algorithms.

The IRTF has also described a set of metrics and related trade-off between metrics that can be used to compare, contrast, and evaluate congestion control techniques [RFC5166]. [RFC5783] provides a snapshot of congestion-control research in 2008.

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6. IANA Considerations

This memo includes no request to IANA.

RFC Editor Note: If there are no requirements for IANA, the section will be removed during conversion into an RFC by the RFC Editor.

7. Security Considerations

This document introduces no new security considerations. Each RFC listed in this document discusses the security considerations of the specification it contains. The security considerations for the use of transports are provided in the references section of the cited RFCs. Security guidance for applications using UDP is provided in the UDP Usage Guidelines [RFC8085].

Section 3.9 describes general requirements relating to the design of safe protocols and their protection from on and off path attack.

Section 3.8 follows current best practice to validate ICMP messages prior to use.

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Appendix A. Internet Congestion Control

Internet transports can reserve capacity at routers or on the links being used. This is sometimes used in controlled environments, but most uses across the Internet do not rely upon prior reservation of capacity along the path they use. In the absence of such a reservation, endpoints are unable to determine a safe rate at which to start or continue their transmission. The use of an Internet path therefore requires a combination of end-to-end transport mechanisms to detect and then respond to changes in the capacity that it discovers is available across the network path.

Buffering (an increase in latency) or congestion loss (discard of a packet) arises when the traffic arriving at a link or network exceeds the resources available. Loss can also occur for other reasons, but it is usually not possible for an endpoint to reliably disambiguate the cause of packet loss (e.g., loss could be due to link corruption, receiver overrun, etc. [RFC3819]). A network device typically uses

a drop-tail policy to drop excess IP packets when its queue(s) becomes full. This use of buffers can also be managed using Active Queue Management (AQM) [RFC7567], which can be combined withb Explicit Congestion Notification signalling.

Internet transports need to react to avoid congestion that impacts other flows sharing a path. The Requirements for Internet Hosts [RFC1122] formally mandates that endpoints perform congestion control. "Because congestion control is critical to the stable operation of the Internet, applications and other protocols that choose to use UDP as an Internet transport must employ mechanisms to prevent congestion collapse and to establish some degree of fairness with concurrent traffic [RFC2914].

The general recommendation in the UDP Guidelines [RFC8085] is that applications SHOULD leverage existing congestion control techniques, such as those defined for TCP [RFC5681], TCP-Friendly Rate Control (TFRC) [RFC5348], SCTP [RFC4960], and other IETF-defined transports. This is because there are many trade offs and details that can have a serious impact on the performance of congestion control for the application they support and other traffic that seeks to share the resources along the path over which they communicate.

Network devices can be configured to isolate the queuing of packets for different flows, or aggregates of flows, and thereby assist in reducing the impact of flow multiplexing on other flows. This could include methods seeking to equally distribute resources between sharing flows, but this is explicitly not a requirement for a network device [Flow-Rate-Fairness]. Endpoints can not rely on the presence and correct configuration of these methods, and therefore even when a path is expected to support such methods, also need to employ methods that work end-to-end.

Experience has shown that successful protocols developed in a specific context or for a particular application tend to also become used in a wider range of contexts. Therefore, IETF specifications by default target deployment on the general Internet, or need to be defined for use only within a controlled environment.

A.1. Flow Multiplexing and Congestion

When a transport uses a path to send packets (i.e. a flow), this impacts any other Internet flows (possibly from or to other endpoints) that share the capacity of any common network device or link (i.e., are multiplexed) along the path. As with loss, latency can also be incurred for other reasons [RFC3819] (Quality of Service link scheduling, link radio resource management/bandwidth on demand,

transient outages, link retransmission, and connection/resource setup below the IP layer, etc).

When choosing an appropriate sending rate, packet loss needs to be considered. Although losses are not always due to congestion, endpoint congestion control needs to conservatively react to loss as a potential signal of reduced available capacity and reduce the sending rate. Many designs place the responsibility of rate-adaption at the sender (source) endpoint, utilising feedback information provided by the remote endpoint (receiver). Congestion control can also be implemented by determining an appropriate rate limit at the receiver and using this limit to control the maximum transport rate (e.g., using methods such as [RFC5348] and [RFC4828]).

It is normal to observe some perturbation in latency and/or loss when flows shares a common network bottleneck with other traffic. This impact needs to be considered and Internet flows ought to implement appropriate safeguards to avoid inappropriate impact on other flows that share the resources along a path. Congestion control methods satisfy this requirement and therefore can help avoid congestion collapse.

"This raises the issue of the appropriate granularity of a "flow", where we define a 'flow' as the level of granularity appropriate for the application of both fairness and congestion control. [RFC2309] states: "There are a few 'natural' answers: 1) a TCP or UDP connection (source address/port, destination address/port); 2) a source/destination host pair; 3) a given source host or a given destination host. We would guess that the source/destination host pair gives the most appropriate granularity in many circumstances. The granularity of flows for congestion management is, at least in part, a policy question that needs to be addressed in the wider IETF community." [RFC2914]

Endpoints can send more than one flow. "The specific issue of a browser opening multiple connections to the same destination has been addressed by [RFC2616]. Section 8.1.4 states that "Clients that use persistent connections SHOULD limit the number of simultaneous connections that they maintain to a given server. A single-user client SHOULD NOT maintain more than 2 connections with any server or proxy." [RFC2140].

This suggests that there are opportunities for transport connections between the same endpoints (from the same or differing applications) might share some information, including their congestion control state, if they are known to share the same path. [RFC8085] adds "An application that forks multiple worker processes or otherwise uses

multiple sockets to generate UDP datagrams SHOULD perform congestion control over the aggregate traffic."

In the absence of persistent congestion, an endpoint is permitted to increase its congestion window and hence the sending rate. An increase should only occur when there is additional data available to send across the path (i.e., the sender will utilise the additional capacity in the next RTT).

TCP Reno [RFC5681] defines an algorithm, known as the Additive-Increase/ Multiplicative-Decrease (AIMD) algorithm, which allows a sender to exponentially increase the congestion window each RTT from the initial window to the first detected congestion event. This is designed to allow new flows to rapidly acquire a suitable congestion window. Where the bandwidth delay product (BDP) is large, it can take many RTT periods to determine a suitable share of the path capacity. Such high BDP paths benefit from methods that more rapidly increase the congestion window, but in compensation these need to be designed to also react rapidly to any detected congestion (e.g., TCP Cubic [RFC8312]).

A.2. Avoiding Congestion Collapse and Flow Starvation

A significant pathology can arise when a poorly designed transport creates congestion. This can result in severe service degradation or "Internet meltdown". This phenomenon was first observed during the early growth phase of the Internet in the mid 1980s [RFC0896] [RFC0970]. It is technically called "Congestion Collapse". [RFC2914] notes that informally, "congestion collapse occurs when an increase in the network load results in a decrease in the useful work done by the network."

Transports need to be specifically designed with measures to avoid starving other flows of capacity (e.g., [RFC7567]). [RFC2309] also discussed the dangers of congestion-unresponsive flows, and states that "all UDP-based streaming applications should incorporate effective congestion avoidance mechanisms." [RFC7567] and [RFC8085] both reaffirm this, encouraging development of methods to prevent starvation.

A.3. Adjusting the Rate

Congestion Management: The capacity available to a flow could be expressed as the number of bytes in flight, the sending rate or a limit on the number of unacknowledged segments. When determining the capacity used, all data sent by a sender needs to be accounted, this includes any additional overhead or data generated by the transport. A transport performing congestion management

will usually optimise performance for its application by avoiding excessive loss or delay and maintain a congestion window. In steady-state this congestion window reflects a safe limit to the sending rate that has not resulted in persistent congestion. A congestion controller for a flow that uses packet Forward Error Correction (FEC) encoding (e.g., [RFC6363]) needs to consider all additional overhead introduced by packet FEC when setting and managing its congestion window.

One common model views the path between two endpoints as a "pipe". New packets enter the pipe at the sending endpoint, older ones leave the pipe at the receiving endpoint. Congestion and other forms of loss result in "leakage" from this pipe. Received data (leaving the network path at the remote endpoint) is usually acknowledged to the congestion controller.

The rate that data leaves the pipe indicates the share of the capacity that has been utilised by the flow. If, on average (over an RTT), the sending rate equals the receiving rate, this indicates the path capacity. This capacity can be safely used again in the next RTT. If the average receiving rate is less than the sending rate, then the path is either queuing packets, the RTT/path has changed, or there is packet loss.

Appendix B. Best Current Practice in the RFC-Series

Like RFC2119, this document borrows heavily from earlier publications addressing the need for end-to-end congestion control, and this subsection provides an overview of key topics.

[RFC2914] provides a general discussion of the principles of congestion control. Section 3 discussed Fairness, stating "The equitable sharing of bandwidth among flows depends on the fact that all flows are running compatible congestion control algorithms". Section 3.1 describes preventing congestion collapse.

Congestion collapse was first reported in the mid 1980s [RFC0896], and at that time was largely due to TCP connections unnecessarily retransmitting packets that were either in transit or had already been received at the receiver. We call the congestion collapse that results from the unnecessary retransmission of packets classical congestion collapse. Classical congestion collapse is a stable condition that can result in throughput that is a small fraction of normal [RFC0896]. Problems with classical congestion collapse have generally been corrected by improvements to timer and congestion control mechanisms, implemented in modern implementations of TCP [Jacobson88]. This classical congestion collapse was a key focus of [RFC2309].

A second form of congestion collapse occurs due to undelivered packets, where Section 5 of [RFC2914] notes: "Congestion collapse from undelivered packets arises when bandwidth is wasted by delivering packets through the network that are dropped before reaching their ultimate destination. This is probably the largest unresolved danger with respect to congestion collapse in the Internet today. Different scenarios can result in different degrees of congestion collapse, in terms of the fraction of the congested links' bandwidth used for productive work. The danger of congestion collapse from undelivered packets is due primarily to the increasing deployment of open-loop applications not using end-to-end congestion control. Even more destructive would be best-effort applications that *increase* their sending rate in response to an increased packet drop rate (e.g., automatically using an increased level of FEC (Forward Error Correction))."

Section 3.3 of [RFC2914] notes: "In addition to the prevention of congestion collapse and concerns about fairness, a third reason for a flow to use end-to-end congestion control can be to optimize its own performance regarding throughput, delay, and loss. In some circumstances, for example in environments with high statistical multiplexing, the delay and loss rate experienced by a flow are largely independent of its own sending rate. However, in environments with lower levels of statistical multiplexing or with per-flow scheduling, the delay and loss rate experienced by a flow is in part a function of the flow's own sending rate. Thus, a flow can use end-to-end congestion control to limit the delay or loss experienced by its own packets. We would note, however, that in an environment like the current best-effort Internet, concerns regarding congestion collapse and fairness with competing flows limit the range of congestion control behaviors available to a flow."

In addition to the prevention of congestion collapse and concerns about fairness, a flow using end-to-end congestion control can optimize its own performance regarding throughput, delay, and loss [RFC2914].

The standardization of congestion control in new transports can avoid a congestion control "arms race" among competing protocols [RFC2914]. That is, avoid designs of transports that could compete for Internet resource in a way that significantly reduces the ability of other flows to use the Internet. The use of standard methods is therefore encouraged.

The popularity of the Internet has led to a proliferation in the number of TCP implementations [RFC2914]. A variety of non-TCP transports have also been deployed. Some transport implementations fail to use standardised congestion avoidance mechanisms correctly

because of poor implementation [RFC2525]. However, this is not the only reason for not using standard methods. Some transports have chosen mechanisms that are not presently standardised, or have adopted approaches to their design that differ from present standards. Guidance is needed therefore not only for future standardisation, but to ensure safe and appropriate evolution of transports that have not presently been submitted for standardisation.

Appendix C. Revision Notes

Note to RFC-Editor: please remove this entire section prior to publication.

Individual draft -00:

- o Comments and corrections are welcome directly to the authors or via the IETF TSVWG, working group mailing list.

Individual draft -01:

- o This update is proposed for initial WG comments.
- o If there is interest in progressing this document, the next version will include more complete referencing to cited material.

Individual draft -02:

- o Correction of typos.

Individual draft -03:

- o Added section 1.1 with text on current BCP status with additional alignment and updates to RFC2914 on Congestion Control Principles (after question from M. Scharf).
- o Edits to consolidate starvation text.
- o Added text that multicast currently noting that this is out of scope.
- o Revised sender-based CC text after comment from C. Perkins (Section 3.1, 3.3 and other places).
- o Added more about FEC after comment from C. Perkins.
- o Added an explicit reference to RFC 5783 and updated this text (after question from M. Scharf).

- o To avoid doubt, added a para about "Each new transport needs to make its own design decisions about how to meet the recommendations and requirements for congestion control."
- o Upated references.

Individual draft -04:

- o Correction of NiTs. Further clarifications.
- o This draft does not attempt to address further alignment with draft-ietf-tcpm-rto-consider. This will form part of a future revision.

Individual draft -05:

- o Moved intro to appendix and re-issued as a live draft.
- o This draft does not attempt to address further alignment with draft-ietf-tcpm-rto-consider. This will form part of a future revision.

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Packet Loss Signaling for Encrypted Protocols
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Abstract

This document describes a protocol-independent method that employs two bits to allow endpoints to signal packet loss in a way that can be used by network devices to measure and locate the source of the loss. The signaling method applies to all protocols with a protocol-specific way to identify packet loss. The method is especially valuable when applied to protocols that encrypt transport header and do not allow an alternative method for loss detection.

Status of This Memo

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1. Introduction

1.1. Motivation for Passive On-Path Loss Observation

Packet loss is hard and pervasive problem of day-to-day network operation. Proactively detecting, measuring, and locating it is crucial to maintaining high QoS and timely resolution of crippling end-to-end throughput issues. To this effect, in a TCP-dominated world, network operators have been heavily relying on information present in the clear in TCP headers: sequence and acknowledgment numbers and SACKs when enabled (see [RFC8517]). These allow for quantitative estimation of packet loss by passive on-path observation. Additionally, the lossy segment (upstream or downstream from the observation point) can be quickly identified by moving the passive observer around.

With encrypted protocols, the equivalent transport headers are encrypted and passive packet loss observation is not possible, as described in [TRANSPORT-ENCRYPT].

Measuring TCP loss between similar endpoints cannot be relied upon to evaluate encrypted protocol loss. Different protocols could be routed by the network differently and the fraction of Internet traffic delivered using protocols other than TCP is increasing every year. It is imperative to measure packet loss experienced by encrypted protocol users directly.

1.2. On-Path Loss Observation

There are three sources of loss that network operators need to observe to guarantee high QoS:

- `_upstream loss_` - loss between the sender and the observation point (Section 4.2)
- `_downstream loss_` - loss between the observation point and the destination (Section 4.4)
- `_observer loss_` - loss by the observer itself that does not cause downstream loss (Section 4.5)

The upstream and downstream loss together constitute `_end-to-end loss_` (Section 4.1).

1.3. On-Path Loss Signaling

Following the recommendation in [RFC8558] of making path signals explicit, this document proposes adding two explicit loss bits to the clear portion of the protocol headers to restore network operators' ability to maintain high QoS. These bits can be added to an unencrypted portion of a header belonging to any protocol layer, e.g. IP (see [IP]) and IPv6 (see [IPv6]) headers or extensions, such as [IPv6AltMark], UDP surplus space (see [UDP-OPTIONS] and [UDP-SURPLUS]), reserved bits in a QUIC v1 header (see [QUIC-TRANSPORT]).

1.4. Recommended Use of the Signals

The loss signal is not designed for use in automated control of the network in environments where loss bits are set by untrusted hosts. Instead, the signal is to be used for troubleshooting individual flows as well as for monitoring the network by aggregating information from multiple flows and raising operator alarms if aggregate statistics indicate a potential problem.

2. Notational Conventions

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

3. Loss Bits

The draft introduces two bits that are to be present in packets capable of loss reporting. These are packets that include protocol headers with the loss bits. Only loss of packets capable of loss reporting is reported using loss bits.

Whenever this specification refers to packets, it is referring only to packets capable of loss reporting.

- Q: The "sQuare signal" bit is toggled every N outgoing packets as explained below in Section 3.1.
- L: The "Loss event" bit is set to 0 or 1 according to the Unreported Loss counter, as explained below in Section 3.2.

Each endpoint maintains appropriate counters independently and separately for each separately identifiable flow (each subflow for multipath connections).

3.1. Setting the sSquare Bit on Outgoing Packets

The sSquare Value is initialized to the Initial Q Value (0) and is reflected in the Q bit of every outgoing packet. The sSquare value is inverted after sending every N packets (a Q Run). Hence, Q Period is $2*N$. The Q bit represents "packet color" as defined by [RFC8321]. The sSquare Bit can also be called an Alternate Marking bit.

Observation points can estimate the upstream losses by counting the number of packets during a half period of the square signal, as described in Section 4.

3.1.1. Q Run Length Selection

The sender is expected to choose N (Q run length) based on the expected amount of loss and reordering on the path. The choice of N strikes a compromise - the observation could become too unreliable in case of packet reordering and/or severe loss if N is too small, while short flows may not yield a useful upstream loss measurement if N is too large (see Section 4.2).

The value of N MUST be at least 64 and be a power of 2. This requirement allows an Observer to infer the Q run length by observing one period of the square signal. It also allows the Observer to identify flows that set the loss bits to arbitrary values (see Section 6).

If the sender does not have sufficient information to make an informed decision about Q run length, the sender SHOULD use $N=64$, since this value has been extensively tried in large-scale field tests and yielded good results. Alternatively, the sender MAY also choose a random N for each flow, increasing the chances of using a Q run length that gives the best signal for some flows.

The sender MUST keep the value of N constant for a given flow.

3.2. Setting the Loss Event Bit on Outgoing Packets

The Unreported Loss counter is initialized to 0, and L bit of every outgoing packet indicates whether the Unreported Loss counter is positive ($L=1$ if the counter is positive, and $L=0$ otherwise). The value of the Unreported Loss counter is decremented every time a packet with $L=1$ is sent.

The value of the Unreported Loss counter is incremented for every packet that the protocol declares lost, using whatever loss detection machinery the protocol employs. If the protocol is able to rescind the loss determination later, a positive Unreported Loss counter MAY

be decremented due to the rescission, but it SHOULD NOT become negative due to the rescission.

This loss signaling is similar to loss signaling in [ConEx], except the Loss Event bit is reporting the exact number of lost packets, whereas Echo Loss bit in [ConEx] is reporting an approximate number of lost bytes.

For protocols, such as TCP ([TCP]), that allow network devices to change data segmentation, it is possible that only a part of the packet is lost. In these cases, the sender MUST increment Unreported Loss counter by the fraction of the packet data lost (so Unreported Loss counter may become negative when a packet with L=1 is sent after a partial packet has been lost).

Observation points can estimate the end-to-end loss, as determined by the upstream endpoint, by counting packets in this direction with the L bit equal to 1, as described in Section 4.

4. Using the Loss Bits for Passive Loss Measurement

4.1. End-To-End Loss

The Loss Event bit allows an observer to calculate the end-to-end loss rate by counting packets with L bit value of 0 and 1 for a given flow. The end-to-end loss rate is the fraction of packets with L=1.

The assumption here is that upstream loss affects packets with L=0 and L=1 equally. If some loss is caused by tail-drop in a network device, this may be a simplification. If the sender's congestion controller reduces the packet send rate after loss, there may be a sufficient delay before sending packets with L=1 that they have a greater chance of arriving at the observer.

4.2. Upstream Loss

Blocks of N (Q Run length) consecutive packets are sent with the same value of the Q bit, followed by another block of N packets with an inverted value of the Q bit. Hence, knowing the value of N, an on-path observer can estimate the amount of upstream loss after observing at least N packets. The upstream loss rate ("u") is one minus the average number of packets in a block of packets with the same Q value ("p") divided by N ($u = 1 - \text{avg}(p)/N$).

The observer needs to be able to tolerate packet reordering that can blur the edges of the square signal.

The observer needs to differentiate packets as belonging to different flows, since they use independent counters.

4.3. Correlating End-to-End and Upstream Loss

Upstream loss is calculated by observing packets that did not suffer the upstream loss. End-to-end loss, however, is calculated by observing subsequent packets after the sender's protocol detected the loss. Hence, end-to-end loss is generally observed with a delay of between 1 RTT (loss declared due to multiple duplicate acknowledgments) and 1 RTO (loss declared due to a timeout) relative to the upstream loss.

The flow RTT can sometimes be estimated by timing protocol handshake messages. This RTT estimate can be greatly improved by observing a dedicated protocol mechanism for conveying RTT information, such as the Latency Spin bit of [QUIC-TRANSPORT].

Whenever the observer needs to perform a computation that uses both upstream and end-to-end loss rate measurements, it SHOULD use upstream loss rate leading the end-to-end loss rate by approximately 1 RTT. If the observer is unable to estimate RTT of the flow, it should accumulate loss measurements over time periods of at least 4 times the typical RTT for the observed flows.

If the calculated upstream loss rate exceeds the end-to-end loss rate calculated in Section 4.1, then either the Q Period is too short for the amount of packet reordering or there is observer loss, described in Section 4.5. If this happens, the observer SHOULD adjust the calculated upstream loss rate to match end-to-end loss rate.

4.4. Downstream Loss

Because downstream loss affects only those packets that did not suffer upstream loss, the end-to-end loss rate ("e") relates to the upstream loss rate ("u") and downstream loss rate ("d") as $(1-u)(1-d)=1-e$. Hence, $d=(e-u)/(1-u)$.

4.5. Observer Loss

A typical deployment of a passive observation system includes a network tap device that mirrors network packets of interest to a device that performs analysis and measurement on the mirrored packets. The observer loss is the loss that occurs on the mirror path.

Observer loss affects upstream loss rate measurement since it causes the observer to account for fewer packets in a block of identical Q

bit values (see {{upstreamloss}}). The end-to-end loss rate measurement, however, is unaffected by the observer loss, since it is a measurement of the fraction of packets with the set L bit value, and the observer loss would affect all packets equally (see Section 4.1).

The need to adjust the upstream loss rate down to match end-to-end loss rate as described in Section 4.3 is a strong indication of the observer loss, whose magnitude is between the amount of such adjustment and the entirety of the upstream loss measured in Section 4.2. Alternatively, a high apparent upstream loss rate could be an indication of significant reordering, possibly due to packets belonging to a single flow being multiplexed over several upstream paths with different latency characteristics.

5. ECN-Echo Event Bit

While the primary focus of the draft is on exposing packet loss, modern networks can report congestion before they are forced to drop packets, as described in [ECN]. When transport protocols keep ECN-Echo feedback under encryption, this signal cannot be observed by the network operators. When tasked with diagnosing network performance problems, knowledge of a congestion downstream of an observation point can be instrumental.

If downstream congestion information is desired, this information can be signaled with an additional bit.

- E: The "ECN-Echo Event" bit is set to 0 or 1 according to the Unreported ECN Echo counter, as explained below in Section 5.1.

5.1. Setting the ECN-Echo Event Bit on Outgoing Packets

The Unreported ECN-Echo counter operates identically to Unreported Loss counter (Section 3.2), except it counts packets delivered by the network with CE markings, according to the ECN-Echo feedback from the receiver.

This ECN-Echo signaling is similar to ECN signaling in [ConEx]. ECN-Echo mechanism in QUIC provides the number of packets received with CE marks. For protocols like TCP, the method described in [ConEx-TCP] can be employed. As stated in [ConEx-TCP], such feedback can be further improved using a method described in [ACCURATE].

5.2. Using E Bit for Passive ECN-Reported Congestion Measurement

A network observer can count packets with CE codepoint and determine the upstream CE-marking rate directly.

Observation points can also estimate ECN-reported end-to-end congestion by counting packets in this direction with a E bit equal to 1.

The upstream CE-marking rate and end-to-end ECN-reported congestion can provide information about downstream CE-marking rate. Presence of E bits along with L bits, however, can somewhat confound precise estimates of upstream and downstream CE-markings in case the flow contains packets that are not ECN-capable.

6. Protocol Ossification Considerations

Accurate loss information is not critical to the operation of any protocol, though its presence for a sufficient number of flows is important for the operation of networks.

The loss bits are amenable to "greasing" described in [RFC8701], if the protocol designers are not ready to dedicate (and ossify) bits used for loss reporting to this function. The greasing could be accomplished similarly to the Latency Spin bit greasing in [QUIC-TRANSPORT]. Namely, implementations could decide that a fraction of flows should not encode loss information in the loss bits and, instead, the bits would be set to arbitrary values. The observers would need to be ready to ignore flows with loss information more resembling noise than the expected signal.

7. Security Considerations

Passive loss observation has been a part of the network operations for a long time, so exposing loss information to the network does not add new security concerns for protocols that are currently observable.

In the absence of upstream packet loss, the Q bit signal does not provide any information that cannot be observed by simply counting packets transiting a network path. In the presence of upstream packet loss, the Q bit will disclose the loss, but this is information about the environment and not the endpoint state. The L bit signal discloses internal state of the protocol's loss detection machinery, but this state can often be gleamed by timing packets and observing congestion controller response. Hence, loss bits do not provide a viable new mechanism to attack data integrity and secrecy.

7.1. Optimistic ACK Attack

A defense against an Optimistic ACK Attack, described in [QUIC-TRANSPORT], involves a sender randomly skipping packet numbers to detect a receiver acknowledging packet numbers that have never been received. The Q bit signal may inform the attacker which packet numbers were skipped on purpose and which had been actually lost (and are, therefore, safe for the attacker to acknowledge). To use the Q bit for this purpose, the attacker must first receive at least an entire Q Run of packets, which renders the attack ineffective against a delay-sensitive congestion controller.

A protocol that is more susceptible to an Optimistic ACK Attack with the loss signal provided by Q bit and uses a loss-based congestion controller, SHOULD shorten the current Q Run by the number of skipped packets numbers. For example, skipping a single packet number will invert the sQuare signal one outgoing packet sooner.

8. Privacy Considerations

To minimize unintentional exposure of information, loss bits provide an explicit loss signal - a preferred way to share information per [RFC8558].

New protocols commonly have specific privacy goals, and loss reporting must ensure that loss information does not compromise those privacy goals. For example, [QUIC-TRANSPORT] allows changing Connection IDs in the middle of a connection to reduce the likelihood of a passive observer linking old and new subflows to the same device. A QUIC implementation would need to reset all counters when it changes the destination (IP address or UDP port) or the Connection ID used for outgoing packets. It would also need to avoid incrementing Unreported Loss counter for loss of packets sent to a different destination or with a different Connection ID.

9. IANA Considerations

This document makes no request of IANA.

10. Change Log

10.1. Since version 02

- Minor improvement and clarifications

10.2. Since version 01

- Clarified Q Period selection
- Added an optional E (ECN-Echo Event) bit
- Clarified L bit calculation for protocols that allow partial data loss due to a change in segmentation (such as TCP)

10.3. Since version 00

- Addressed review comments
- Improved guidelines for privacy protections for QIUC

11. Acknowledgments

The sSquare bit was originally suggested by Kazuho Oku in early proposals for loss measurement and is an instance of the "alternate marking" as defined in [RFC8321].

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TCP Maintenance and Minor Extensions
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Some Congestion Experienced in TCP
draft-grimes-tcpm-tcpsce-01

Abstract

This memo classifies a TCP code point ESCE ("Echo Some Congestion Experienced") for use in feedback of IP code point SCE ("Some Congestion Experienced").

Status of This Memo

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

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1. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119] and [RFC8174] when, and only when, they appear in all capitals, as shown here.

2. Introduction

This memo requests a TCP header codepoint for use as ESCE.

This memo limits its scope to the definition of the TCP codepoint ESCE, with a few brief illustrations of how it may be used.

SCE provides early and proportional feedback to the CC (congestion control) algorithms for transport protocols, including but not limited to TCP. The [sce-repo] is a Linux kernel modified to support SCE, including:

- * Enhancements to Linux's [cake] (Common Applications Kept Enhanced) AQM to support SCE signaling
- * Modifications to the TCP receive path to reflect SCE signals back to the sender

- * The addition of three new TCP CC algorithms that modify the originals to add SCE support: Reno-SCE, DCTCP-SCE and Cubic-SCE (work in progress as of this writing)

3. Background

[I-D.morton-tsvwg-sce] defines the IP SCE codepoint.

4. TCP Receiver

The mechanism defined to feed back SCE signals to the sender explicitly makes use of the ESCE ("Echo Some Congestion Experienced") code point in the TCP header.

4.1. Single ACK implementation

Upon receipt of a packet an ACK is immediately generated, the SCE codepoint is copied into the ESCE codepoint of the ACK. This keeps the count of bytes SCE marked or not marked properly reflected in the ACK packet(s). This valid implementation has the downside of increasing ACK traffic. This implementation is NOT RECOMMENDED, but useful for experimental work.

4.2. Simple Delayed ACK implementation

Upon receipt of a packet without an SCE codepoint traditional delayed ACK processing is performed. Upon receipt of a packet with an SCE codepoint immediate ACK processing SHOULD be done, this allows some delaying of ACK's, but creates earlier feedback of the congested state. This has the negative effect of over signalling ESCE.

4.3. Dithered Delayed ACK implementation

Upon receipt of a packet the SCE codepoint is stored in the TCP state. Multiple packets state may be stored. Upon generation of an ACK, normal or delayed, the stored SCE state is used to set the state of ESCE. If no SCE state is in the TCP state, then the ESCE code point MUST NOT be set. If all of the packets to be ACKed have SCE state set then the ESCE code point MUST be set in the ACK. If some of the packets to be ACKed have SCE state set then some proportional number of ACK packets SHOULD be sent with the ESCE code point set. Though this may defer a ESCE congestion signal when there is not a next packet for some time it is generally accepted that such sparse flows are not the source of congestion and thus the delayed signal is of low impact. The goal is to have the same number of bytes marked with ESCE as arrived with SCE.

4.4. Advanced ACK implementation

The Advanced ACK implementation actually immediately flushes any pending ACK's up to the `_previous_` segment when the state of the SCE marking `_changes_`, allowing consecutive packets with the same SCE state to be coalesced by the normal delayed-ack logic. The ACK volume is then inflated only slightly compared to an unmarked connection, and may actually involve fewer acks than a connection involving CE marks or losses, during which delayed acks are temporarily disabled.

4.5. ACK Thinning

Ack thinning is something that has been considered, given that [cake] includes an optional ack-filter which does thinning. We have, for example, added consideration of the ESCE bit to Cake's ack-filter. Mathematically, the most extreme errors possible in either direction, due to ack thinning, are easily corrected during subsequent RTTs.

5. TCP Sender

The recommended response to each single segment marked with ESCE is to reduce cwnd by an amortised $1/\sqrt{\text{cwnd}}$ segments. If the growth rate is greater than that provided by the Reno-linear algorithm - eg. slow-start exponential or CUBIC polynomial - then the growth rate SHOULD also be reduced.

Other responses, such as the $1/\text{cwnd}$ from DCTCP, are also acceptable but may perform less well.

There are no changes to the response functions with respect to CE or packet loss specified by this draft, hence [RFC3168] and [RFC8511] are still applicable

This is still an area of continued investigation.

6. Related Work

6.1. More Accurate ECN Feedback in TCP [I-D.ietf-tcpm-accurate-ecn]

AccECN replaces the [RFC3168] definition of the ECE and CWR bits (and the former NS bit) with its own three-bit field. This new interpretation is predicated on successfully negotiating AccECN, and is not useful to SCE implementations because it provides no information about any ECT(1) codepoints received, and SCE does not need or use the extra information about CE marks that the three-bit field does provide. Hence SCE may be considered mutually exclusive with AccECN on any given connection.

AccECN supports a fallback to [RFC3168] style signalling during the three-way handshake by recognising the normal requests and responses of an [RFC3168] endpoint. SCE endpoints also exhibit [RFC3168] behaviour during the handshake, so this mutual exclusivity occurs naturally. There will therefore be no confusion on the wire between the two experiments, even though SCE does not explicitly negotiate its upgrade from plain [RFC3168] behaviour.

The latter is consistent with the (now historic) Nonce Sum specification, which also did not explicitly negotiate support, and used the same additional ECN codepoint and TCP header bit that SCE is now requesting.

7. IANA Considerations

This document requests one of the reserved bits in the TCP header, with the former TCP NS ("Nonce Sum") bit (bit 7) being suggested due to similarities with its previous usage. [RFC8311] (section 3) obsoletes the NS codepoint making it available for use.

8. Security Considerations

There are no Security considerations.

9. Acknowledgements

TBD

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Enhanced DiffServ by In-band Signaling
draft-han-tsvwg-enhanced-diffserv-00

Abstract

There are real-time applications requiring deterministic services in terms of bandwidth and latency in various industries, such as network based AR/VR (Augmented Reality and Virtual Reality), industrial internet. This document proposes a solution which enhances DiffServ in IPv6 to provide end to end bandwidth guaranteed service and latency guaranteed service.

Status of This Memo

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1. Introduction

[RFC8557][RFC8578] identifies some use cases from different industries which have a common need for "deterministic flows". Such flows require guaranteed bandwidth and bounded latency.

This document proposes to use in-band signaling together with DiffServ to provide Latency Guaranteed Service and Bandwidth Guaranteed Service. Network traffic is classified according to application requirements and resources such as bandwidth and buffer are dedicated to provide a reliable and scalable service, while unused reserved resources could be used by best effort traffic.

The solution is compatible with existing Quality-of-Service (QoS) mechanism as long as there is enough network resources, and is not restricted to network topologies. Resource Reservation is done by in-band signaling [I-D.han-tsvwg-ip-transport-qos], and IP path can

be achieved using existing routing protocols (IGP or BPG), or the explicit path routing such as segment routing [RFC8402].

2. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] RFC 8174 [RFC8174] when, and only when, they appear in all capitals, as shown here.

Abbreviations used in this documents:

E2E

End-to-end.

EH

IPv6 Extension Header or Extension Option.

QoS

Quality of Service.

OAM

Operation and Management.

In-band Signaling

In telecommunications, in-band signaling is sending control information within the same band or channel used for voice or video.

Out-of-band Signaling

out-of-band signaling is that the control information sent over a different channel, or even over a separate network.

IP flow

For non-IPSec, an IP flow is identified by the source, destination IP address, the protocol number, the source and destination port number.

IP path

An IP path is the route that IP flow will traverse. It could be the shortest path determined by routing protocols (IGP or BPG), or the explicit path such as segment routing [RFC8402].

QoS channel

A forwarding channel that is QoS guaranteed. It provides additional QoS service to IP forwarding. A QoS channel can be

used for one or multiple IP flows depending on the granularity of in-band signaling.

srTCM

Single Rate Three Color Marker [RFC2697]

trTCM

Two Rate Three Color Marker [RFC4115]

LGS

Latency Guaranteed Service

BGS

Bandwidth Guaranteed Service

BES

Best Effort Service

CIR

Committed Information Rate.

PIR

Peak Information Rate.

HbH-EH

IPv6 Hop-by-Hop Extension Header.

Dst-EH

IPv6 Destination Extension Header.

HbH-EH-aware node

Network nodes that are configured to process the IPv6 Hop-by-Hop Extension Header.

3. DiffServ and IntServ

3.1. Differentiated Services

Differentiated services or DiffServ [RFC2475] provides a simple, scalable and coarse-grained mechanism to support quality of services on IP networks by classifying and managing network traffic.

DiffServ uses DiffServ Code Point (DSCP) which is the first 6 bits of the TOS field [RFC2474] to classify packets when they enter the network and then are separated into different queues. With DiffServ each router handles each packet differently. Per-Hop Forwarding Behavior (PHB) is a way of forwarding a particular flow or group of

flows marked with a particular DSCP value using a particular method on a DiffServ node.

3.2. Integrated Services

IntServ [RFC3175] or integrated services specifies more fine-grained QoS, which is often contrasted with DiffServ's coarse-grained control system. IntServ definitely can support the applications requiring special QoS guarantee if it is deployed in a network, supported by Host OS and integrated with application. However, IntServ works on a small-scale only. When you scale up the network, it is difficult to keep track of all of the reservations and session states. Thus, IntServ is not scalable. Another problem of IntServ is it is not application driven, tedious provisioning cross different network must be done earlier. The provisioning is slow and hard to maintain.

4. In-band Signaling

In-band signaling messages are carried along with the payload, and it is guaranteed that the signaling follows the same path as the data flow.

In-band QoS signaling for IPv6 was discussed by Lawrence Roberts in 2005 [I-D.roberts-inband-qos-ipv6]. The requirements of IP in-band signaling was proposed by Jon Haper in 2007 [I-D.harper-inband-signalling-requirements]. Telecommunications Industry Association (TIA) published a standard for "QoS Signaling for IP QoS Support and Sender Authentication" in 2006 [TIA].

This draft uses the in-band signaling method proposed by [I-D.han-tsvwg-ip-transport-qos], which offers a solution in IPv6 for QoS service using in-band signaling to guarantee certain level of service quality. Advantages of using in-band signaling and what parameters are carried etc. are discussed in great details. Please refer to [I-D.han-tsvwg-ip-transport-qos] for signaling details, which is not in the scope of this document.

5. Design Targets

The design targets of the proposal are outlined as follows:

1. Service Guarantee:

- * Provide guaranteed and minimized (queuing) latency for LGS (Latency Guaranteed Service) flows. The maximum latency is guaranteed, minimized and predictable at each hop, if LGS flow rate is confirmed with their pre-claimed parameters (CIR, PIR, CBS, EBS).

- * Provide guaranteed bandwidth for BGS (Bandwidth Guaranteed Service) flows. The bandwidth of CIR is guaranteed at each hop, if BGS flow rate is confirmed with their pre-claimed parameters (CIR,PIR,CBS,EBS).
 - 2. No Starvation: LGS flows will never starve other types of lower priority flows (BGS and BES). BGS flows will never starve BE flows.
 - 3. No sacrifice of link utilization: When the total rate of LGS flow is less than the committed rate (sum of CIR of all LGS flows), other class flows (BGS and BES) can use the remained bandwidth. When the total rate of LGS flow is less than the committed rate (sum of CIR of all LGS flows), and, the total rate of BGS flow is less than the committed rate (sum of CIR of all BGS flows), other class flows (BES) can use the remained bandwidth.
 - 4. Fairness within the same class: All LGS flows will share the bandwidth within its class. All BGS flows will share the bandwidth within its class. All BE flows will share the bandwidth within BE class.
 - 5. Backward compatible with current DiffServ, and can coexist and can be deployed incrementally.
6. Processing Procedure
- 6.1. Introduction

The section will describe more details about the in-band signaling integrated with DiffServ to achieve the guaranteed service. It is composed of following contents:

1. The purpose of in-band signaling for enhanced DiffServ.
2. The info carried in the signaling.
3. The admission process with in-band signaling.
4. User traffic classification.
5. User traffic classification and New DSCP class.
6. Queuing and Scheduling Consideration.
7. Class Based Traffic Shaper.

The document [Enhanced_DSCP] has more details for the experiments using the proposal to achieve the latency guaranteed service. The theoretical formula to estimate the maximum latency is given. Different tests with different combination of traffic are done. Results are analyzed and further researches are discussed.

6.2. The purpose of in-band signaling

The in-band signaling procedures described in [I-D.han-tsvwg-ip-transport-qos] is to make network support new services. There are two types of new services provided:

1. Latency Guaranteed Service (LGS): Provide guaranteed and minimized queuing latency. The maximum latency of a LGS flow is guaranteed, minimized and predictable at each hop if the LGS flow rate is confirmed with its pre-claimed parameters (CIR, PIR, CBS, EBS).
2. Bandwidth Guaranteed Service (BGS): provide guaranteed bandwidth. The CIR bandwidth of a BGS flow is guaranteed at each hop if the BGS flow rate is confirmed with its pre-claimed parameters.

In addition to above, the use of in-band signaling with DiffServ will make the flow level QoS expectation available at each network device on the IP path. Then network device could collect all flows signaling info to form an accurate instruction for QoS programming automatically. Without such automation process, DiffServ will only be able to rely on the network management procedures to configure the QoS for each class at each hop one by one, this is called Per Hop Behavior.

6.3. The info carried in in-band signaling

The detailed info carried in the signaling was described in the draft [I-D.han-tsvwg-ip-transport-qos].

In order to achieve the service guarantees, admission control is needed for user's service expectation. Also to get the best service guarantee in network, some network device need to do traffic shaping to measure and control traffic for difference service.

To satisfy above two requirements, each flow's service expectation and traffic pattern should be carried in the signaling.

To describe one or multiple flows traffic character, Single Rate Three Color Marker (srTCM from [RFC2697]) or Two Rate Three Color Marker (trTCM from RFC4115) could be used. The minimum requirement from user's signaling is the Cir (Committed information rate) when

srTCM is used, CIR plus PIR (Peak information rate) when trTCM is used. Optionally, the signaling can carry more info about the traffic, such as CBS, EBS. The exact requirements for signaling will be decided by IETF.

6.4. Admission process with in-band signaling

The admission control should be enforced at any ingress interface connecting to user's device or accepting user's traffic, and also enforced at the egress interfaces that will send the user's traffic to next network device. In addition to other irrelevant admission checking, the admission process for the E-DiffServ minimally has three major steps:

1. If a user is allowed to request the desired service described in its in-band signaling. This may include the security, authentication and billing checks.
2. After adding the new flow for a user's request, if the total rate for all flows in the same class at the same ingress interface exceeds the total bandwidth allocated for the class for the ingress interface, this allocated bandwidth is from management procedures and is less than the ingress interface link rate.
3. After adding the new flow for a user's request, if the total rate for all flows in the same class at the same egress interface exceeds the total bandwidth allocated for the class for the egress interface, this allocated bandwidth is from management procedures and is less than the egress interface link rate.

In above admission control process, if a flow is not allowed, the in-band signaling of the flow is updated accordingly to indicate the desired service is rejected, and the in-band signaling will finally notify the user's host. If a flow is allowed, the in-band signaling of the flow will proceed to continue. For the details of how the closed loop control is formed by in-band signaling, please refer to [I-D.han-tsvwg-ip-transport-qos].

6.5. User traffic classification and New DSCP class

After passed the admission control process, user's traffic is classified to different class based on its requested service. This is also done at the ingress interface like the admission control. There are two options for the classification:

1. For a network without DiffServ enabled, the LGS flows can be classified as EF class to achieve the shortest latency; The BGS

flows can be classified as any type of AFxy class; The BE flows can be classified as BE class.

2. For a network that DiffServ is enabled, the LGS and BGS flows can be classified as two new class; The BE flows can be classified as BE class. To be backward compatible and not interrupting existing DiffServ services, new DSCP classes are proposed for LGS and BGS. The values are TBD.

6.6. User traffic conformity checking

After a user's request service passes the admission control and is accepted, user's traffic is still needed to be checked for its conformity. i.e, if the traffic is conforming to the traffic pattern claimed in the service request.

This is done by a traffic meter (shaper) associated to ingress interface. The algorithm in the meter is using either srTCM or trTCM. The action of the meter will determine what exact class the traffic will be classified as. Only the Green traffic will be marked to the designated class described in 5.5. Other colored traffic (Yellow or Red) will be classified as lower priority class or just discarded. The exact action is up to the decision of IETF.

6.7. Queuing and Scheduling Consideration

The most important factor to provide the guaranteed service is the queuing and scheduling selection at egress interface on each router. There are many researches about this topic. The current widely accepted methods are Class based Strict Priority Queuing (PQ) and Weighed Fair Queuing (WFQ). It is recommended that the PQ+WFQ are used for different service flows. PQ is the best candidate to provide the shortest latency for LGS flows. The WFQ can be used for BGS flows. There are two options for the Queuing and Scheduling selection:

1. For a network without DiffServ enabled, the LGS flows can be queued into the Highest Priority Queue to achieve the shortest latency; The BGS and BE flows can be queued into different Weighted Fair Queues to share bandwidth. The weight for BGS can be obtained from the sum of all BGS flows' CIR. The weight for BE can be calculated from the link rate deducting the bandwidth allocated for LGS and BGS flows.
2. For a network with DiffServ enabled, the LGS flows can be queued into the Secondary Highest Priority Queue just below EF queue; The BGS and BE flows can be queued into different Weighted Fair Queues to share bandwidth with other existing classes. The

weight for BGS can be obtained from the sum of all BGS flows' CIR. The weight for BE can be calculated from the link rate deducting the bandwidth allocated for all AFxy classes, LGS and BGS flows.

6.8. Class Based Traffic Shaper

To guarantee each class's traffic are conforming to the pre-allocated bandwidth or resource for the class, class-based traffic shaper can be used for traffics before they are queued into different queues. Only Green traffic will be queued into the queue described above. Other colored traffic (Yellow or Red) will be queued into a lower priority queue or discard. The exact actions will be decided by IETF. The purpose of using traffic shaper for each class is to protect the service for each class when other class traffic is exceeding its allocation. This is critical to satisfy the design target outlined in section 5.

The algorithm for the shaper is similar to the ingress traffic shaper described in section 6.6. The algorithm in the meter is using either srTCM or trTCM.

The Traffic Shaper's Parameter such as PIR, CIR, CBS, EBS can be obtained from all flow's traffic parameters embedded into the in-band signaling. i.e, PIR or CIR can be calculated by the sum of all allowed (same class) flow's PIR or CIR values. CBS and EBS can also be obtained from flow's signaling message if it carries, or from management process. The detailed standard will be decided by IETF.

7. IANA Considerations

The new DSCP values are required to be allocated by IANA.

8. Security Considerations

TBD.

9. References

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Appendix A. Acknowledgements

TBD.

Appendix B. Testing Results

The analysis of the queuing delay and testing results can be found at [Enhanced_DSCP] .

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Diffserv to QCI Mapping
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Abstract

As communication devices become more hybrid, smart devices include more media-rich communication applications, and the boundaries between telecommunication and other applications becomes less clear. Simultaneously, as the end-devices become more mobile, application traffic transits more often between enterprise networks, the Internet, and cellular telecommunication networks, sometimes using simultaneously more than one path and network type. In this context, it is crucial that quality of service be aligned between these different environments. However, this is not always the case by default, and cellular communication networks use a different QoS nomenclature from the Internet and enterprise networks. This document specifies a set of 3rd Generation Partnership Project (3GPP) Quality of Service (QoS) Class Identifiers (QCI) and 5G QoS Identifiers (5QI) to Differentiated Services Code Point (DSCP) mappings, to reconcile the marking recommendations offered by the 3GPP with the recommendations offered by the IETF, so as to maintain a consistent QoS treatment between cellular networks and the Internet. This mapping can be used by enterprises or implementers expecting traffic to flow through both types of network, and wishing to align the QoS treatment applied to one network under their control with the QoS treatment applied to the other network.

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1. Introduction

3GPP has become the preferred set of standards to define cellular communication principles and protocols. With the augmented capabilities of smartphones, cellular networks increasingly carry non-communication traffic and interconnect with the Internet and Enterprise IP networks. The access networks defined by the 3GPP present several design challenges for ensuring end-to-end quality of service when these networks interconnect with the Internet or to enterprise networks. Some of these challenges relate to the nature of the cellular network itself, being centrally controlled, collision-free and primarily designed around subscription level and associated services, while other challenges relate to the fact that the 3GPP standards are not administered by the same standards body as Internet protocols. While 3GPP has developed tools to enable QoS over cellular networks, little guidance exists on how to maintain consistency of QoS treatment between cellular networks and the Internet, or IP-based Enterprise networks. As such, enterprises and

other operators managing traffic flowing through both 3GPP and Internet Protocol links do not always know how to translate 3GPP QoS identifiers into Internet Protocol QoS identifiers and vice versa. The purpose of this document is to provide such guidance.

1.1. Related Work

Several RFCs outline Diffserv QoS recommendations over IP networks, including:

[RFC2474] specifies the Diffserv Codepoint Field. This RFC also details Class Selectors, as well as the Default Forwarding (DF) treatment. [RFC2475] defines a Diffserv architecture [RFC3246] specifies the Expedited Forwarding (EF) Per-Hop Behavior (PHB) [RFC2597] specifies the Assured Forwarding (AF) PHB. [RFC3662] specifies a Lower Effort Per-Domain Behavior (PDB) [RFC4594] presents Configuration Guidelines for Diffserv Service Classes [RFC5127] presents the Aggregation of Diffserv Service Classes [RFC5865] specifies a DSCP for Capacity Admitted Traffic [RFC8622] presents the Lower-Effort Per-Hop Behavior (LE-PHB) for Diffserv

Note: [RFC4594] is intended to be viewed as a framework for supporting Diffserv in any network, regardless of the underlying data-link or physical layer protocols. Its principles could apply to IP traffic carried over cellular DataLink and Physical Layer mediums. Additionally, the principles of [RFC4594] apply to any traffic entering the Internet, regardless of its original source location. Thus, [RFC4594] describes different types of traffic expected in IP networks and provides guidance as to what DSCP marking(s) should be associated with each traffic type. As such, this document draws heavily on [RFC4594], as well as [RFC5127], and [RFC8100].

In turn, the relevant standard for cellular LTE QoS is 3GPP [TS 23.107], which defines more than 1600 General Packet Radio Service (GPRS) QoS profiles across multiple classes and associated attributes. As this quantity is large and source of potential complexity, the 3GPP Technical Specification Group Services and System Aspects, defining the Policy Charging Control Architecture, leverages a subset of QoS profiles used as QoS Class Identifiers (QCI). For 5G communications, [TS 23.501] defines 5G QoS Identifiers. This document draws on these specifications, which are being progressively updated; the current version of which (at the time of writing) are 3GPP [TS 23.203] v16.2.0 and 3GPP [TS 23.501] v16.3.0.

1.2. Applicability Statement

This document is applicable to the use of Differentiated Services that interconnect with 3GPP LTE or 5G cellular networks (referred to as cellular, throughout this document, for simplicity). These guidelines are applicable whether cellular network endpoints are IP-enabled, in which case these guidelines can apply end-to-end, starting from the endpoint operating system, or whether cellular network endpoints are either not IP-enabled, or do not enable QoS, in which case these guidelines apply at the interconnection point between the cellular access network and the Internet or IP network. Such interconnection point can commonly occur at the infrastructure Radio Unit (eNodeB), within the infrastructure core network (CN), or at the edge of the core network toward the Internet or an Enterprise IP network, for example within the Packet Data Network Gateway (P-GW).

1.3. Document Organization

This document is organized as follows:

- o Section 2 introduces the QoS logic marking applicable to each domain. We introduce the general logic of Diffserv and the notion of domain boundary. We then examine the 3GPP QoS logic, detailing the concept of bearer, QCI and 5QIs, and showing how QCIs and 5QIs are implemented and used.
- o Section 3 provides general recommendations for QoS support at the 3GPP / Diffserv domains boundaries.
- o Section 4 proposes a Diffserv to QCI translation scheme, so as to suggest DSCP values that can be directly translated into QCIs or 5QIs values, when traffic moves into a 3GPP domain where QCIs or 5QIs must be used.
- o Section 5 proposes a reverse mapping, from QCI to Diffserv. As many QCIs intents do not match existing DSCP values, new DSCP values are proposed wherever needed.
- o Section 6 underlines the resulting IANA requirements for this mapping.
- o Section 7 and Section 8 examine the security consequences of these new mapping schemes.

1.4. Requirements language

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

1.5. Terminology Used in this Document

Key terminology used in this document includes:

EPS Bearer: a path that user traffic (IP flows) uses between the UE and the PGW.

GGSN: Gateway GPRS Support Node, responsible for the internetworking between the GPRS network and external networks. PGW performs the GGSN functionalities in EPC.

IP BS Manager: Internet Protocol Bearer Service Manager, a function that manages the IP bearer services. Part of this function can include translation of QoS parameters between EPS and external networks.

UE: User Equipment, the end-device.

EPS Session: a PDN connection, comprised of one or more IP flows, that a UE established and maintains to the EPS.

SAE: System Architecture Evolution.

RAN: Radio access network, the radio segment of the LTE network EPS.

EPC: Evolved Packet Core, the core segment of the LTE network EPS.

EPS: Evolved Packet System, the LTE network, comprised of the RANs and EPC.

HSS: Home Subscriber Server, the database that contains user-related and subscriber-related information.

LUS: Live Uplink Streaming, a video flow (often real-time) sent from a source to a sink.

SGW: Serving Gateway, the point of interconnection between the RAN and the EPC.

PGW: Packet Data Network Gateway, point of interconnection between the EPC and external IP networks.

MME: Mobility Management Entity: software function that handles the signaling related to mobility and security for the access network.

PCEF: Policy and Charging Enforcement Function, provides user traffic handling and QoS within the PGW.

PCRF: Policy and Charging Rules Function, a functional entity that provides policy, bandwidth and charging functions for each EPS user.

2. Service Comparison and Default Interoperation of Diffserv and 3GPP LTE and 5G

2.1. Diffserv Domain Boundaries

It is important to recognize that 3GPP standards allow support for principles of [RFC2475]. The user equipment (UE) application function may have no active QoS support, or may support Diffserv or IntServ functions [TS 23.207] v15 5.2.2. When Diffserv is supported, an Internet Protocol Bearer Service Manager (IP BS Manager) function integrated to the UE can translate Diffserv parameters into LTE QoS parameters (e.g. QCI). As such, the UE IP BS Manager function may act as a Diffserv domain boundary (as defined in [RFC2475]) between a Diffserv domain present within the UE networking stack and the LTE Radio Access Network.

Additionally, the P-GW interconnects the UE data plane to the external networks. The P-GW is the element that implements Gateway GPRS (General Packet Radio Service) Support Node (GGSN) functionalities in Evolved Packet Core (EPS) networks. The GGSN includes an IP BS manager function that acts as a Diffserv Edge function, and can translate Diffserv parameters to 3GPP QoS parameters (e.g. QCI or 5G NSA 5QI) and vice versa. In SA 5G, the user plane and control plane are separated, and the P-GW for the user plane (PGW-U) joins the Service Gateway (SGW-U) into the User Plane Function (UPF).

As such, 3GPP standards allow the existence of a Diffserv domain within the UE and outside of the EPS boundaries. The Diffserv domain is not considered within the EPS, where QCIs or 5QIs are used to define and transport QoS parameters.

2.2. QCI and Bearer Model in 3GPP

It is important to note that LTE (4G) and 5G standards are an evolution of UMTS standards (2G, 3G) developed in the 1990s. As such, these standards recognize [RFC2475] (1998), but not [RFC4594] (2006). EPS networks rely on the notion of bearers. A bearer is a conduit between the UE and the P-GW, and LTE supports two types of bearers:

- o GBR: Guaranteed Bit Rate bearers. These bearers allocate network resources associated to a GBR value associated to the bearer. These resources stay allocated (reserved) for the duration of the existence of the GBR bearer and the flow it carries.
- o Non-GBR bearers: also called default bearers, non-GBR are bearers for which network resources are not permanently allocated during the existence of the bearer and the flow it carries. As such, one or more non-GBR bearer may share the same set of temporal resources.

Each EPS bearer is identified by a name and number, and is associated with specific QoS parameters of various types:

1. QoS Class Identifiers (QCI). A QCI is a scalar associated to a bearer, and is used to define the type of traffic and service expected in the bearer. [TS 23.107] v15 defines 4 basic classes: conversational, streaming, interactive and background. These classes are defined more in details in Section 2.3. Each class includes multiple types of traffic, each associated with sets of attributes, thus permitting the definition of more than 1600 different QoS profiles. [TS 23.203] v16 6.1.7.2 reduces the associated complexity by characterizing traffic based on up to 6 attributes, resulting in 26 types of traffic and their associated expected service requirements through the use of 26 scalars (QCI). Each QCI is defined in the relation to the following six performance characteristics:
2. Resource Type (GBR or Non-GBR).
3. Priority: a scalar used as a tie breaker if two packets compete for a given network resource. A lower value indicates a higher priority.
4. Packet Delay Budget: marks the upper bound for the time that a packet may be delayed between the UE and the PCRF (Policy and Charging Rules Function) or the PCEF function (Policy and Charging Enforcement Function) residing inside the P-GW. PCEF supports offline and online charging while PCRF is real-time.

Either component, being in charge of policing and charging, can determine resource reservation actions and policies.

5. Packet Error Loss Rate, defines an upper bound for a rate of non-congestion related packet losses. The purpose of the PELR is to allow for appropriate link layer protocol configurations when needed.
6. Maximum Burst Size (only for some GBR QCIs), defines the amount of data which the Radio Access Network (RAN) is expected to deliver within the part of the Packet Delay Budget allocated to the link between the UE and the radio base station. If more data is transmitted from the application, the Packet Delay Budget may be exceeded.
7. Data rate Averaging Window (only for some GBR QCIs), defines the 'sliding window' duration over which the GBR and MBR are calculated.

Although [TS 23.203] v16 6.1.7.2 associates each QCI with up to 6 characteristics, it is clear that these characteristics are constrained by bandwidth allocation, in particular on the radio link that are associated with three commonly used parameters:

1. Maximum Bit Rate (MBR), only valid for GBR bearers, defines the maximum sustained traffic rate that the bearer can support.
2. Guaranteed Bit Rate (GBR), only valid for GBR bearers, defines the minimum traffic rate reserved for the bearer.
3. Aggregate MBR (AMBR), defines the total amount of bit rate available for a group of non-GBR bearers. AMBR is often used to provide differentiated service levels to different types of customers.

2.3. QCI Definition and Logic

[TS 23.107] v15 6.3 defines four possible traffic classes. These four general classes are used as the foundation from which QCI categories are defined in [TS 23.203]. The categorization is made around the notion of sensitivity to delay.

2.3.1. Conversational

The conversational class is intended to carry real-time traffic flows. The expectation of such class is a live conversation between two humans or a group. Examples of such flows include [TS 23.107] v15 6.3.1 telephony speech, but also VoIP and video conferencing.

Video conference would be seen as a different class from telephony in the Diffserv model. However, 3GPP positions them in the same general class, as all of them include live conversations. Sensitivity to delay is high because of the real-time nature of the flows. The time relation between the stream entities have to be preserved (to maintain the same experience for all flows and all parties involved in the conversation).

2.3.2. Streaming

The streaming class is intended for flows where the user is watching real time video, or listening to real-time audio (or both). The real-time data flow is always aiming at a live (human) destination. It is important to note that the Streaming class is intended to be both a real-time flow and a one-way transport. Two-way real-time traffic belongs to the conversational class, and non-real-time flows belong to the interactive or the background classes. The delay sensitivity is lower than that of Conversational flows, because it is expected that the receiving end includes a time alignment function (e.g. buffering). As the flow is unidirectional, variations in delay do not conversely affect the user experience as long as the variation is within the alignment function boundaries.

2.3.3. Interactive

The interactive class is intended for flows where a machine or human is requesting data from a remote equipment (e.g. a server). Examples of human interaction with the remote equipment are: web browsing, data base retrieval, server access. Examples of machines interaction with remote equipment are: polling for measurement records and automatic data base enquiries (tele-machines). Delay sensitivity is average, and is based on round trip time (overall time between emission of the request and reception of the response).

2.3.4. Background

The background class applies to flows where the equipment is sending or receiving data files without direct user interaction (e.g. emails, SMS, database transfers etc.) As such, delay sensitivity is low. Background is described as delivery-time insensitive.

Based upon the above principles, [TS 23.203] has defined several QCI's. [TS 23.203] Release 16 6.1.7-A defines 26 QCIs:

QC I	Resource Type	Priority Level	Packet Delay Budget	Packet Error Loss	Example Services
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1	GBR	2	100 ms	10.E-2	Conversational Voice
2	GBR	4	150 ms	10.E-3	Conversational Video (Live Streaming)
3	GBR	3	50 ms	10.E-3	Real Time Gaming, V2X messages, Electricity distribution (medium voltage) Process automation (monitoring)
4	GBR	5	300 ms	10.E-6	Non-Conversational Video (Buffered Streaming)
65	GBR	0.7	75 ms	10.E-2	Mission Critical user plane Push To Talk voice (e.g., MCPTT)
66	GBR	2	100 ms	10.E-2	Non-Mission-Critical user plane Push To Talk voice
67	GBR	1.5	100 ms	10.E-3	Mission Critical Video user plane
75	GBR	2.5	50 ms	10.E-2	V2X messages
71	GBR	5.6	150 ms	10.E-6	"Live" Uplink Streaming
72	GBR	5.6	300 ms	10.E-4	"Live" Uplink Streaming
73	GBR	5.6	300 ms	10.E-8	"Live" Uplink Streaming
74	GBR	5.6	500 ms	10.E-8	"Live" Uplink Streaming
76	GBR	5.6	500 ms	10.E-4	"Live" Uplink Streaming
5	Non-GBR	1	100 ms	10.E-6	IMS Signalling

6	Non-GBR	6	300 ms	10.E-6	Video (Buffered Streaming) TCP-based (e.g. www, email, chat, ftp, p2p file sharing, progressive video)
7	Non-GBR	7	100 ms	10.E-3	Voice, Video (live streaming), interactive gaming
8	Non-GBR	8	300 ms	10.E-6	Video (buffered streaming) TCP-based (e.g. www, email, chat, ftp, p2p file sharing, progressive video)
9	Non-GBR	9	300 ms	10.E-6	Same as 8
69	Non-GBR	0.5	60 ms	10.E-6	Mission Critical delay sensitive signalling (e.g., MC-PTT signalling, MC Video signalling)
70	Non-GBR	5.5	200 ms	10.E-6	Mission Critical Data (e.g. example services are the same as QCI 6/8/9)
79	Non-GBR	6.5	50 ms	10.E-2	V2X messages
80	Non-GBR	6.8	10 ms	10.E-2	Low latency eMMB applications (TCP/UDP-based); augmented reality
82	GBR	1.9	10 ms	10.E-6	Discrete automation (small packets)
83	GBR	2.2	10 ms	10.E-4	Discrete automation (large packets)
84	GBR	2.4	30 ms	10.E-5	Intelligent Transport Systems

85	GBR	2.1	5 ms	10.E-5	Electricity Distribution - High Voltage
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Several QCIs cover the same application types. For example, QCIs 6, 8 and 9 all apply to buffered streaming video and web applications. However, LTE context distinguishes several types of customers and environments. As such, QCI 6 can be used for the prioritization of non-real-time data (i.e. most typically TCP-based services/applications) of MPS (multimedia priority services) subscribers, when the network supports MPS. QCI 8 can be used for a dedicated "premium bearer" (e.g. associated with premium content) for any subscriber or subscriber group, while QCI 9 can be used for the default bearer for non-privileged subscribers.

2.4. QCI implementations

[TS 23.203] v16 defines multiple QCIs. However, a UE or a EPS does not need to implement all supported QCIs, even when all matching types of traffic are expected between the UE and the network. In practical implementations, it is common for an EPS to implement one GBR bearer where at least QCI 1 is directed (and optionally other GBR QCIs), and another default bearer where all other traffic to and from the same UE is directed. The QCI associated to that second bearer may depend on the subscriber category. As such, the QCI listed in Section 2.3 are indicative of performance and traffic type classifications, and are not strict in their implementation mandate.

2.5. 5QI and flow-based QoS Model in 3GPP 5G

While 4G LTE QoS is enforced at the EPS bearer level, 5G QoS focuses on the transported flows. A QoS Flow ID (QFI) identifies a given QoS Flow. In the User Plane, the traffic with a given QFI within a PDU session is treated in the same way. The 5G QoS Identifier (5QI) is used in 3GPP to identify a specific QoS forwarding behavior for a 5G QoS Flow (similar to the QCI value for LTE, with the difference that 5QI applies to a flow, carried at some point in a bearer, while QCI applies to a bearer within which certain types of flows are expected). As such, the 5QI defines packet loss rate, packet delay budget etc. In the 5G system, the entity named Session Management Function (SMF) manages the QoS information. The SMF provides QFI information to the Radio Access Network (RAN) for mapping the various QoS flows to access network resources (i.e., data radio bearers). The RAN performs packet marking in the uplink on a per QoS Flow basis, with a marking value determined by the QFI and a treatment matching the associated 5QI. The SMF also instructs the User Plane Function (UPF) for classification, bandwidth enforcement and marking

of the user plane traffic in downlink. Such packet marking information includes the QFI and the transport level packet marking value (i.e., the value of the DSCP field in the outer IP header). In [TS 23.501], 3GPP provides the 5G QoS characteristics associated with the 5QIs, and specifies the packet forwarding treatment that a QoS Flow receives end-to-end, from the UE up to the UPF (and back). The characteristics considered are:

- o Resource type, i.e., if the flow requires resources to be allocated for Guaranteed Bandwidth Rate (GBR), delay critical GBR (DCGBR), or non-GBR.
- o Default priority level
- o Packet delay budget (PDB), including the PDB consumed in the 5G core network
- o Packet Error Rate (PER)
- o Averaging window (in milliseconds), applicable for GBR and delay-critical GBR
- o Default maximum data burst volume (in bytes), applicable for delay-critical GBR only

The following table shows a simplified version from the standardized [TS 23.501] 5QI to QoS characteristics mapping.

5QI	Resource Type	Priority Level	Packet Delay Budget	Packet Error Rate	Default Max Burst	Default Avg Window	Example Services
1	GBR	20	100 ms	10.E-2	N/A	2000	Conversational voice
2	GBR	40	150 ms	10.E-3	N/A	2000	Conversational video (live streaming)
3	GBR	30	50 ms	10.E-3	N/A	2000	Real time gaming, V2X messages, medium voltage electricity

								dist.
4	GBR	50	300 ms	10.E-6	N/A	2000		non-conversational video (buffered streaming)
65	GBR	7	75 ms	10.E-2	N/A	2000		Mission critical user plane push-to-talk voice (e.g. MCPTT)
66	GBR	20	100 ms	10.E-3	N/A	2000		Non-mission critical user plane push-to-talk voice
67	GBR	15	100 ms	10.E-3	N/A	2000		Mission critical user plane video
71	GBR	56	150 ms	10.E-6	N/A	2000		"Live" uplink streaming
72	GBR	56	300 ms	10.E-4	N/A	2000		"Live" uplink streaming
73	GBR	56	300 ms	10.E-8	N/A	2000		"Live" uplink streaming
74	GBR	56	500 ms	10.E-8	N/A	2000		"Live" uplink streaming
76	GBR	56	500 ms	10.E-4	N/A	2000		"Live" uplink streaming
5	non-GBR	10	100 ms	10.E-6	N/A	N/A		IMS signaling
6	non-GBR	60	300 ms	10.E-6	N/A	N/A		Video (Buffered Streaming) TCP-based (e.g. www, email, chat, etc.)

7	non-GBR	70	100 ms	10.E-3	N/A	N/A	Voice, Video (live streaming), interactive gaming
8	non-GBR	80	300 ms	10.E-6	N/A	N/A	Video (Buffered Streaming) TCP-based (e.g. www, email, chat, etc.)
9	non-GBR	90	300 ms	10.E-6	N/A	N/A	Same as 8
69	non-GBR	5	60 ms	10.E-6	N/A	N/A	Mission Critical delay sensitive signalling (e.g., MC-PMC)
70	non-GBR	55	200 ms	10.E-6	N/A	N/A	Mission critical data (e.g. same examples as QCI/5QI 6,7,8
79	non-GBR	65	50 ms	10.E-2	N/A	N/A	V2X messages
80	non-GBR	68	10 ms	10.E-6	N/A	N/A	Low latency eMMB applications (TCP/UDP-based); augmented reality
82	DCGBR	19	10 ms	10.E-4	255 B	2000 ms	Discrete automation
83	DCGBR	22	10 ms	10.E-4	1354 B	2000 ms	Discrete automation

84	DCGBR	24	30 ms	10.E-5	1354 B	2000 ms	Intelligent Transport Systems
85	DCGBR	21	5 ms	10.E-5	255 B	2000 ms	Electricity distribution, High voltage, V2X
86	DCGBR	18	5 ms	10.E-4	1354 B	2000 ms	V2X, collision avoidance, platooning, self driving

Although the focus of 5QI and that of QCI is different, it should be noted that the traffic examples provided by each QCI match the traffic intent for a 5QI with matching number. The 5QI default priority level is a tenfold expression of the QCI priority level (and this document will refer to the QCI priority levels for simplicity). As such, any given QCI or 5QI can be equivalised to the same DSCP value. In turn, an application and its given DSCP value can be expressed either in a QCI or a 5QI (provided that both exist for the associated traffic or application).

2.6. GSMA IPX Guidelines Interpretation and Conflicts

3GPP standards do not define or recommend any specific mapping between each QCI or 5QI and Diffserv, and leaves that mapping choice to the operator of the Edge domain boundary (e.g. UE software stack developer, P-GW operator). However, 3GPP defines that "for the IP based backbone, Differentiated Services defined by IETF shall be used" ([TS 23.107] v15 6.4.7).

The GSM Association (GSMA) has published an Inter-Service Provider IP Backbone Guideline reference document [ir.34] that provides technical guidance to participating service providers for connecting IP based networks and services to achieve roaming and inter-working services. The document built upon [RFC3246] and [RFC2597], and upon the initial definition of 4 service classes in [TS 23.107] v15 to recommend a mapping to EF for conversational traffic, to AF41 for Streaming traffic, to AF31, AF21 and AF11 for different traffic in the Interactive class, and to BE for background traffic.

These GSMA Guidelines were developed without reference to existing IETF specifications for various services, referenced in Section 1.1. Additionally, the same recommendations remained while new traffic

types under each 3GPP general class were added. As such, the GSMA recommendations yield to several inconsistencies with [RFC4594], including:

- o Recommending EF for real-time (conversational) video, for which [RFC4594] recommends AF41.
- o Recommending AF31 for DNS traffic, for which [RFC4594] recommends the standard service class (DF)
- o Recommending AF31 for all types of signaling traffic, thus losing the ability to differentiate between the various types of signaling flows, as recommended in [RFC4594] section 5.1.
- o Recommending AF21 for WAP browsing and WEB browsing, for which [RFC4594] recommends the High Throughput data class
- o Recommending AF11 for remote connection protocols, such as telnet or SSH, for which [RFC4594] recommends the OAM class.
- o Recommending DF for file transfers, for which [RFC4594] recommends the High Throughput Data class.
- o Recommending DF for email exchanges, for which [RFC4594] recommends the High Throughput Data class.
- o Recommending DF for MMS exchanged over SMTP, for which [RFC4594] recommends the High Throughput Data class.

The document [ir.34] also does not provide guidance for QCIs other than 1 to 9, leaving the case of the 12 other QCIs unaddressed.

Thus, document [ir.34] conflicts with the overall Diffserv traffic-conditioning service plan, both in the services specified and the code points specified for them. As such, these two plans cannot be normalized. Rather, as discussed in [RFC2474] Section 2, the two domains (GSMA and other IP networks) are different Differentiated Services Domains separated by a Differentiated Services Boundary. At that boundary, code points from one domain are translated to code points for the other, and maybe to Default (zero) if there is no corresponding service to translate to.

3. P-GW Device Marking and Mapping Capability Recommendations

This document assumes and RECOMMENDS that all P-GWs (as the interconnects between cellular and other IP networks) and all other interconnection points between cellular and other IP networks support the ability to:

- o mark DSCP, per Diffserv standards
- o mark QCI, per the [TS 23.203] standard, or 5QI, as per the [TS 23.501] standard
- o support fully-configurable mappings between DSCP and QCI or 5QI
- o process DSCP markings set by cellular endpoint devices

This document further assumes and RECOMMENDS that all cellular endpoint devices (UE) support the ability to:

- o mark DSCP, per Diffserv standards
- o mark QCI, per the [TS 23.203] standard, OR 5QI, per the [TS 23.501] standard
- o support fully-configurable mappings between DSCP (set by applications in software) and QCI or 5QI (set by the operating system and/or the LTE infrastructure)

Having made the assumptions and recommendations above, it bears mentioning that while the mappings presented in this document are RECOMMENDED to replace the current common default practices (as discussed in Section 2.3 and Section 2.4), these mapping recommendations are not expected to fit every last deployment model, and as such MAY be overridden by network administrators, as needed.

4. DSCP to QCI or 5QI Mapping Recommendations

4.1. Control Traffic

4.1.1. Network Control Protocols

The Network Control service class is used for transmitting packets between network devices (e.g., routers) that require control (routing) information to be exchanged between nodes within the administrative domain, as well as across a peering point between different administrative domains.

[RFC4594] Section 3.2 recommends that Network Control Traffic be marked CS6 DSCP. Additionally, as stated in [RFC4594] Section 3.1: "CS7 DSCP value SHOULD be reserved for future use, potentially for future routing or control protocols."

Network Control service is not directly called by any specific QCI or 5QI description, because 3GPP network control does not operate over UE data channels. It should be noted that encapsulated routing

protocols for encapsulated or overlay networks (e.g., VPN, Network Virtualization Overlays, etc.) are not Network Control Traffic for any physical network at the cellular space; hence, they SHOULD NOT be marked with CS6 in the first place, and are not expected to be forwarded to the cellular data plane.

However, when such network control traffic is forwarded, it is expected to receive a high priority and level of service. As such, packets marked to CS7 DSCP are RECOMMENDED to be mapped to QCI 82, thus benefiting from a dedicated bearer with low packet error loss rate (10^{-4}) and low budget delay (10 ms). Similarly, it is RECOMMENDED to map Network Control Traffic marked CS6 to QCI/5QI 82, thereby admitting it to the Discrete Automation (GBR) category with a relative priority level of 1.9/19.

4.1.2. Operations, Administration, and Maintenance (OAM)

The OAM (Operations, Administration, and Maintenance) service class is recommended for OAM&P (Operations, Administration, and Maintenance and Provisioning). The OAM service class can include network management protocols, such as SNMP, Secure Shell (SSH), TFTP, Syslog, etc., as well as network services, such as NTP, DNS, DHCP, etc.

[RFC4594] Section 3.3, recommends that OAM traffic be marked CS2 DSCP.

Applications using this service class require a low packet loss but are relatively not sensitive to delay. This service class is configured to provide good packet delivery for intermittent flows. As such, packets marked to CS2 are RECOMMENDED to be mapped to QCI/5QI 9, thus admitting it to the non-GBR Buffered video traffic, with a relative priority of 9/90.

4.2. User Traffic

User traffic is defined as packet flows between different users or subscribers. It is the traffic that is sent to or from end-terminals and that supports a very wide variety of applications and services [RFC4594] Section 4.

Network administrators can categorize their applications according to the type of behavior that they require and MAY choose to support all or a subset of the defined service classes.

4.2.1. Telephony

The Telephony service class is recommended for applications that require real-time, very low delay, very low jitter, and very low packet loss for relatively constant-rate traffic sources (inelastic traffic sources). This service class SHOULD be used for IP telephony service. The fundamental service offered to traffic in the Telephony service class is minimum jitter, delay, and packet loss service up to a specified upper bound. [RFC4594] Section 4.1 recommends that Telephony traffic be marked EF DSCP.

3GPP [TS 23.203] describes two QCIs adapted to Voice traffic: QCI 1 (GBR) and QCI 7 (non-GBR). The same logic is found in [TS 23.501] for the same 5QIs. However, Telephony traffic as intended in [RFC4594] supposes resource allocation control. Telephony SHOULD be configured to receive guaranteed forwarding resources so that all packets are forwarded quickly. The Telephony service class SHOULD be configured to use Priority Queuing system. QCI 7 does not match these conditions. As such, packets marked to EF are RECOMMENDED to be mapped to QCI/5QI 1, thus admitting it to the GBR Conversational Voice category, with a relative priority of 2/20.

4.2.2. Signaling

The Signaling service class is recommended for delay-sensitive client-server (e.g., traditional telephony) and peer-to-peer application signaling. Telephony signaling includes signaling between 1) IP phone and soft-switch, 2) soft-client and soft-switch, and 3) media gateway and soft-switch as well as peer-to-peer using various protocols. This service class is intended to be used for control of sessions and applications. [RFC4594] Section 4.2 recommends that Signaling traffic be marked CS5 DSCP.

While Signaling is recommended to receive a superior level of service relative to the default class (e.g., relative to QCI 7), it does not require the highest level of service (i.e., GBR and very high priority). As such, it is RECOMMENDED to map Signaling traffic marked CS5 DSCP to QCI/5QI 4, thereby admitting it to the GBR Non-conversational video category, with a relative priority level of 5/50.

Note: Signaling traffic for native Voice dialer applications should be exchanged over a control channel, and is not expected to be forwarded in the data-plane. However, Signaling for non-native (OTT) applications may be carried in the data-plane. In this case, Signaling traffic is control-plane traffic from the perspective of the voice/video telephony overlay-infrastructure. As such, Signaling

should be treated with preferential servicing versus other data-plane flows.

4.2.3. Multimedia Conferencing

The Multimedia Conferencing service class is recommended for applications that require real-time service for rate-adaptive traffic. [RFC4594] Section 4.3 recommends Multimedia Conferencing traffic be marked AF4x (that is, AF41, AF42, and AF43, according to the rules defined in [RFC2475]). The Diffserv model allows for three values to allow for different relative priorities of flows of the same nature.

The primary media type typically carried within the Multimedia Conferencing service class marked AF41 is video intended to be a component of a real-time exchange; as such, it is RECOMMENDED to map AF41 into the Conversational Video (Live Streaming) category, with a GBR. Specifically, it is RECOMMENDED to map AF41 to QCI/5QI 2, thereby admitting AF41 into the GBR Conversational Video, with a relative priority of 4/40.

AF42 is typically reserved for video intended to be a component of real-time exchange, but which criticality is less than traffic carried with a marking of AF41. As such, it is RECOMMENDED to map AF42 into the Conversational Video (Live Streaming) category, with a GBR, but a lower priority than QCI/5QI 2. Specifically, it is RECOMMENDED to map AF42 to QCI/5QI 4, thereby admitting AF42 into the GBR Conversational Video, with a relative priority of 5/50.

Traffic marked AF43 is typically used for real-time video exchange of lower criticality. As such, it is RECOMMENDED to map AF43 into the Conversational Video (Live Streaming) category, but without a GBR. Specifically, it is RECOMMENDED to map AF43 to QCI/5QI 7, thereby admitting AF43 into the non-GBR Voice, Video and Interactive gaming, with a relative priority of 7/70.

4.2.4. Real-Time Interactive

The Real-Time Interactive service class is recommended for applications that require low loss and jitter and very low delay for variable-rate inelastic traffic sources. Such applications may include inelastic video-conferencing applications, but may also include gaming applications (as pointed out in [RFC4594] Sections 2.1 through 2.3 and Section 4.4. [RFC4594] Section 4.4 recommends Real-Time Interactive traffic be marked CS4 DSCP.

The primary media type typically carried within the Real-Time Interactive service class is video; as such, it is RECOMMENDED to map

this class into a low latency Category. Specifically, it is RECOMMENDED to map CS4 to QCI 80, thereby admitting Real-Time Interactive traffic into the non-GBR category Low Latency eMBB (enhanced Mobile Broadband) applications with a relative priority of 6.8. In cases where GBR is required, for example because a single bearer is allocated for all non-GBR traffic, using a GBR equivalent is also acceptable. In this case, it is RECOMMENDED to map CS4 to QCI/5QI 3, thereby admitting Real-Time Interactive traffic into the GBR category Real-time gaming, with a relative priority of 3/30.

4.2.5. Multimedia Streaming

The Multimedia Streaming service class is recommended for applications that require near-real-time packet forwarding of variable-rate elastic traffic sources. Typically, these flows are unidirectional. [RFC4594] Section 4.5 recommends Multimedia Streaming traffic be marked AF3x (that is, AF31, AF32, and AF33, according to the rules defined in [RFC2475]).

The primary media type typically carried within the Multimedia Streaming service class is video; as such, it is RECOMMENDED to map this class into a Video Category. Specifically, it is RECOMMENDED to map AF31 to QCI/5QI 4, thereby admitting AF31 into the GBR Non Conversational Video category, with a relative priority of 5/50.

Flows marked with AF32 are expected to be of the same nature as flows marked with AF32, but with a lower criticality. As such, these flows may not require a dedicated bearer with GBR. Therefore, it is RECOMMENDED to map AF32 to QCI/5QI 6, thereby admitting AF32 traffic into the non-GBR category Video (Buffered Streaming) with a relative priority of 6/60.

Flows marked with AF33 are expected to be of the same nature as flows marked with AF31 and AF32, but with the lowest criticality. As such, it is RECOMMENDED to map AF33 to QCI/5QI 8, thereby admitting AF33 traffic into the non-GBR category Video (Buffered Streaming) with a relative priority of 8/80.

4.2.6. Broadcast Video

The Broadcast Video service class is recommended for applications that require near-real-time packet forwarding with very low packet loss of constant rate and variable-rate inelastic traffic sources. Typically, these flows are unidirectional. [RFC4594] Section 4.6 recommends Broadcast Video traffic be marked CS3 DSCP.

As directly implied by the name, the primary media type typically carried within the Broadcast Video service class is video; as such,

it is RECOMMENDED to map this class into a Video Category. Specifically, it is RECOMMENDED to map CS3 to QCI/5QI 4, thereby admitting Multimedia Streaming into the GBR Non Conversational Video category, with a relative priority of 5/50. In cases where GBR availability is constrained, using a non-GBR equivalent is also acceptable. In this case, it is RECOMMENDED to map CS3 to QCI/5QI 6, thereby admitting Real-Time Interactive traffic into the non-GBR category Video with a relative priority of 6/60.

4.2.7. Low-Latency Data

The Low-Latency Data service class is recommended for elastic and time-sensitive data applications, often of a transactional nature, where a user is waiting for a response via the network in order to continue with a task at hand. As such, these flows are considered foreground traffic, with delays or drops to such traffic directly impacting user productivity. [RFC4594] Section 4.7 recommends Low-Latency Data be marked AF2x (that is, AF21, AF22, and AF23, according to the rules defined in [RFC2475]).

The primary media type typically carried within the Low-Latency Data service class is data; as such, it is RECOMMENDED to map this class into a data Category. Specifically, it is RECOMMENDED to map AF21 to QCI/5QI 70, thereby admitting AF21 into the non-GBR Mission Critical Data category, with a relative priority of 5.5/55.

Flows marked with AF22 are expected to be of the same nature as flows marked with AF21, but with a lower criticality. Therefore, it is RECOMMENDED to map AF22 to QCI/5QI 6, thereby admitting AF22 traffic into the non-GBR category Video and TCP-based traffic, with a relative priority of 6/60.

Flows marked with AF23 are expected to be of the same nature as flows marked with AF21 and AF22, but with the lowest criticality. As such, it is RECOMMENDED to map AF23 to QCI/5QI 8, thereby admitting AF23 traffic into the non-GBR category Video and TCP-based traffic, with a relative priority of 8/80.

It should be noted that a consequence of such classification is that AF22 is mapped to the same QCI and 5QI as CS3, and AF23 is mapped to the same QCI and 5QI as AF33. However, this overlap is unavoidable, as some QCIs and 5QIs express intents that are expressed in the Diffserv domain through distinct marking values, grouped in the 3GPP domain under the same general category.

4.2.8. High-Throughput Data

The High-Throughput Data service class is recommended for elastic applications that require timely packet forwarding of variable-rate traffic sources and, more specifically, is configured to provide efficient, yet constrained (when necessary) throughput for TCP longer-lived flows. These flows are typically not user interactive.

According to [RFC4594] Section 4.8 it can be assumed that this class will consume any available bandwidth and that packets traversing congested links may experience higher queuing delays or packet loss. It is also assumed that this traffic is elastic and responds dynamically to packet loss. [RFC4594] Section 4.8 recommends High-Throughput Data be marked AF1x (that is, AF11, AF12, and AF13, according to the rules defined in [RFC2475]).

The primary media type typically carried within the High-Throughput Data service class is data; as such, it is RECOMMENDED to map this class into a data Category. Specifically, it is RECOMMENDED to map AF11 to QCI/5QI 6, thereby admitting AF11 into the non-GBR Video and TCP-based traffic category, with a relative priority of 6/60.

Flows marked with AF12 are expected to be of the same nature as flows marked with AF11, but with a lower criticality. Therefore, it is RECOMMENDED to map AF12 to QCI/5QI 8, thereby admitting AF12 traffic into the non-GBR category Video and TCP-based traffic, with a relative priority of 8/80.

Flows marked with AF13 are expected to be of the same nature as flows marked with AF11 and AF12, but with the lowest criticality. As such, it is RECOMMENDED to map AF13 to QCI/5QI 9, thereby admitting AF13 traffic into the non-GBR category Video and TCP-based traffic, with a relative priority of 9/90.

It should be noted that a consequence of such classification is that AF11 is mapped to the same QCI as CS3 and AF22, AF12 is mapped to the same QCI and 5QI as Af33 and AF23, and AF13 is mapped to the same QCI and 5QI as CS2. However, this overlap is unavoidable, as some QCIs and 5QIs express intents that are expressed in the Diffserv domain through distinct marking values, grouped in the 3GPP domain under the same general category.

4.2.9. Standard

The Standard service class is recommended for traffic that has not been classified into one of the other supported forwarding service classes in the Diffserv network domain. This service class provides the Internet's "best-effort" forwarding behavior. [RFC4594]

Section 4.9 states that the "Standard service class MUST use the Default Forwarding (DF) PHB".

The Standard service class loosely corresponds to the default non-GBR bearer practice in 3GPP. Therefore, it is RECOMMENDED to map Standard service class traffic marked DF DSCP to QCI/5QI 9, thereby admitting it to the low priority Video and TCP-based traffic category, with a relative priority of 9/90.

4.2.10. Low-Priority Data

The Low-Priority Data service class serves applications that the user is willing to accept without service assurances. This service class is specified in [RFC3662] and [RFC8622]. [RFC3662] and [RFC4594] both recommend Low-Priority Data be marked CS1 DSCP. [RFC8622] updates these recommendations and suggests the LE (000001) marking. As such, this document aligns with this recommendation and notes that CS1 marking has become ambiguous.

The Low-Priority Data service class does not have equivalent in the 3GPP domain, where all service is controlled and allocated differentially. As such, there is no clear QCI or 5QI that could be labelled low priority below the best effort category. As such, it is RECOMMENDED to map Low-Priority Data traffic marked CS1 DSCP and LE DSCP to QCI/5QI 9, thereby admitting it to the low priority Video and TCP-based traffic category, with a relative priority of 9/90.

4.3. Summary of Recommendations for DSCP-to-QCI Mapping

The table below summarizes the [RFC4594] DSCP marking recommendations mapped to 3GPP:

DSCP	Recommended QCI/5QI	Resource Type	Priority Level (QCI/5QI)
CS7	82	GBR	1.9 / 19
CS6	82	GBR	1.9 / 19
EF	1	GBR	2 / 20
CS5	4	GBR	5 / 50
AF43	7	non-GBR	7 / 70
AF42	4	GBR	5 / 50
AF41	2	GBR	4 / 40
CS4	80 3	non-GBR GBR	6.8 / 68, 3 / 30
AF33	8	non-GBR	8 / 80
AF32	6	non-GBR	6 / 60
AF31	4	GBR	5 / 50
CS3	85	GBR	2.1 / 21
AF23	8	Non-GBR	8 / 80
AF22	6	Non-GBR	6 / 60
AF21	70	Non-GBR	5.5 / 55
CS2	9	Non-GBR	9 / 90
AF13	9	Non-GBR	9 / 90
AF12	8	Non-GBR	8 / 80
AF11	6	Non-GBR	6 / 60
CS0	9	Non-GBR	9 / 90
CS1	9	Non-GBR	6.8 / 68
LE	9	Non-GBR	6.8 / 68

5. QCI and 5QI to DSCP Mapping Recommendations

Traffic travelling from the 3GPP domain toward the Internet or the enterprise domain may already display DSCP marking, if the UE is capable of marking DSCP along with, or without, upstream QCI bearer or 5QI marking, as detailed in Section 2.1.

When Diffserv marking is present in the flows originating from the UE and transiting through the CN (Core Network), and if Diffserv marking are not altered or removed on the path toward the Diffserv domain, then the network can be considered as end-to-end Diffserv compliant. In this case, it is RECOMMENDED that the entity providing the translation from 3GPP to Diffserv ignores the QCI or 5QI value and simply forwards unchanged the Diffserv values expressed by the UE in its various flows.

This general recommendation is not expected to fit every last deployment model, and as such Diffserv marking MAY be overridden by network administrators, as needed, before the flows are forwarded to the Internet, the enterprise network or the Diffserv domain in general. Additionally, within a given Diffserv domain, it is generally NOT RECOMMENDED to pass through DSCP markings from unauthenticated, unidentified or unauthorized devices, as these are typically considered untrusted sources, as detailed in Section 7. Such risk is limited within the 3GPP domain where no upstream traffic is admitted without prior authentication of the UE. However, this risk exists when UE traffic is forwarded to an enterprise domain to which the UE does not belong.

In cases where the UE is unable to apply Diffserv marking, or if these markings are modified or removed within the 3GPP domain, such that these markings may not represent the intent expressed by the UE, and in cases where the QCI is available to represent the flow intent, the recommendations in this section apply. These recommendations MAY apply to the boundary between the 3GPP and the Diffserv model, and MAY also apply to the Diffserv domain, when a given applicaiton traffic flows through both the 3GPP and the Diffserv domains (e.g. multiple paths) and when the enterprise administrator wishes to ensure that the same QoS intent is applied for both paths.

5.1. QCI, 5QI and Diffserv Logic Reconciliation

The QCIs and 5QIs are defined as relative priorities for traffic flows which are described by combinations of 6 or more parameters, as expressed in Section 2.2. As such, QCIs and 5QIs also represent flows in terms of multi-dimensional needs, not just in terms of relative priorities. This multi-dimensional logic is different from the Diffserv logic, where each traffic class is represented as a

combination of needs relative to delay, jitter and loss. This characterization around three parameters allows for the construction of a fairly hierarchical traffic categorization infrastructure, where traffic with high sensitivity to delay and jitter also typically has high sensitivity to loss.

By contrast, the 3GPP QCI and 5QI structure presents multiple points where dimensions cross one another with different or opposing vectors. For example, IMS signaling (QCI or 5QI 5) is defined with very high priority (1/10), low loss tolerance (10-6), but is non-GBR and belongs to the signaling category. By contrast, Conversational voice (QCI or 5QI 1) has lower priority (2/20) than IMS signaling, higher loss tolerance (10-2), yet benefits from a GBR. Fitting both QCIs or 5QIs 5 and 1 in a hierarchical model is challenging.

At the same time, QCIs and 5QIs represent needs that can apply to different applications of various criticality but sending flows of the same nature. For example, QCIs or 5QIs 6, 8 and 9 all include voice traffic, video traffic, but also email or FTP. What distinguish these QCIs/5QIs is the criticality of the associated traffic. Diffserv does not envision voice and FTP as possibly belonging to the same class. As the same time, QCIs or 5QIs 2 and 9 include real-time voice traffic. Diffserv does not allow a type of traffic with stated sensitivity to loss, delay and jitter to be split into categories at both end of the priority spectrum.

As such, it is not expected that QCIs and 5QIs can be mapped to the Diffserv model strictly and hierarchically. Instead, a better approach is to observe the various QCI and 5QI categories, and analyze their intent. This process allows for the grouping of several QCIs or 5QIs into hierarchical groups, that can then be translated into ensembles coherent with the Diffserv logic. This approach, in turn, allows for incorporation of new QCIs and 5QIs as the 3GPP model continues to evolve.

It should be noted, however, that such approach results in partial incompatibility. Some QCIs or 5QIs represent an intent that is simply not present in the Diffserv model. In that case, attempting to artificially stitch the QCI/5QI to an existing Diffserv traffic class and marking would be dangerous. QCI or 5QI traffic forwarded to the Diffserv domain would be mixed with Diffserv traffic that would represent a very different intent.

As such, the result of this classification is that some QCIs and 5QIs call for new Diffserv traffic classes and markings. This consequence is preferable to mixing traffic of different natures into the same pre-existing category.

Each QCI is represented with 6 parameters and each 5QI with 7 parameters, including an Example Services value. This parameter is representative of the QCI or 5QI intent. Although [TS 23.203] and [TS 23.501] summarize each QCI or 5QI intent, these standards contain only summaries of more complex classifications expressed in other 3GPP standards. It is often necessary to refer to these other standards to obtain a more complete description of each QCI/5QI and the multiple type of flows that each QCI or 5QI represents.

For the purpose of this document, the QCI or 5QI intent is the primary classification driver, along with the priority level. The secondary elements, such as priority, delay budget and loss tolerance allow for better refinement of the relative classifications of the QCIs and 5QIs. The resource types (GBR, DElay-critical GBR, non-GBR) provide additional visibility into the intent.

Although 26 QCIs are listed in [TS 23.203] and 27 5QIs in [TS 23.501], representing two (GBR, non-GBR) or three resource types (GBR, non-GBR, Delay-Critical GBR) respectively, 21 and 22 priority values, 9 delay budget values, and 7 loss tolerance values, examining the intent in fact surfaces 9 traffic families:

1. Voice QCI/5QI [1] (dialer / conversational voice) is its own group
2. Voice signaling [5] (IMS) is its own group
3. Voice related (other voice applications, including PTT) [65, 66, 69]
4. Video (conversational or not, mission critical or not) [67, 2, 4, 71, 72, 73, 74, 76]
5. Live streaming / interactive gaming is its own group [7]
6. Low latency eMBB, AR/VR is its own group [80]
7. V2X messaging [75, 3, 9]
8. Automation and Transport [82, 83, 84, 85, 86]
9. Non-mission-critical data [6, 8, 9]
10. Mission-critical data is its own group [70]

5.2. Voice [1]

Several QCIs or 5QIs are intended to carry voice traffic. However, QCI/5QI 1 stands apart from the others. Its category is Conversational Voice, but this QCI/5QI is intended to represent the VoLTE voice bearer, for dialer and emergency services. QCI/5QI 1 uses a GBR, and has a priority level of 2/20. Its packet delay budget is 100 ms (from UE to P-GW) with a packet error loss of at most $10.E-2$. As the GBR is allocated by the infrastructure, QCI/5QI 1 is both admitted and allocated dedicated resources. As such, QCI/5QI 1 maps in intent and function to [RFC5865], Admitted Voice, and is RECOMMENDED for mapping to DSCP 44.

5.3. IMS Signaling [5]

QCI/5QI 5 is intended for Signaling. This category does not represent signaling for VoLTE, as such signaling is not conducted over the UE data channels. Instead, QCI/5QI 5 is intended for IMS services. IP Multimedia System (IMS) is a framework for delivering multimedia services over IP networks. These services include real-time and video applications, and their signaling is recommended to be carried, whenever possible, using IETF protocols such as SIP. Being of signaling nature, QCI/5QI 5 is non-GBR. However, being critical to enabling IMS real-time applications, QCI/5QI 5 has a high priority of 1/10. Its packet delay budget is 100 ms, but packet error loss rate very low, at less than $10.E-6$. Overall, QCI/5QI 5 maps rather well to the intent of [RFC4594] signaling for real time applications, and as such is RECOMMENDED to map to [RFC4594] Signaling, CS5.

5.4. Voice-related QCIs and 5QIs [65, 66, 69]

Several QCIs/5QIs display the commonality of targeting voice (non-VoLTE) traffic:

- o QCI/5QI 65 is GBR, mission critical PTT voice, priority 0.7/7
- o QCI/5QI 66 is GBR, non-mission critical PTT voice, priority 2/20
- o QCI/5QI 69 is non-GBR, mission-critical PTT signaling, priority 0.5/5

These QCIs/5QIs are Voice in nature, and naturally fit into a proximity marking model with DSCP 46 and 44.

Additionally, lower priority marks higher precedence intent in QCI and 5QI. However, there is no model in [RFC4594] that distinguishes 3 classes of voice traffic. Therefore, new markings are unavoidable. As such, there is a need to group these markings in the Voice

category (101 xxx), and to order 69, 65 and 66 with different markings to reflect their different priority levels.

Among these three QCIs/5QIs, 69 is non-GBR, intended for mission-critical PTT signaling, with the highest priority of the three, at 0.5/5. 69 is intended for signaling, but is latency sensitive, with a low 60 ms delay budget and a low $10 \cdot 10^{-6}$ loss tolerance. Being of Signaling nature for real time applications, QCI/5QI 69 has proximity of intent with CS5 (Voice signaling, 40), but this marking is already used by QCI/5QI 5. Therefore, it is RECOMMENDED to map QCI/5QI 69 to a new DSCP marking, 41.

Similarly, QCI/5QI 66 is GBR and targeted for non-mission critical PTT voice, with a priority level of 2/20. 66 is Voice in nature, and GBR. However, 66 is intended for non-mission-critical traffic, and has a lower priority than mission-critical Voice, a higher tolerance for delay (100 ms vs 75). As such, 66 cannot fit within [RFC4594] model mapping real-time voice to the class EF (DSCP 46). Here again, a new marking is needed. As such, this QCI/5QI fits in intent and proximity closest to Admitted Voice, but is non-GBR, and therefore non-admitted, guiding a new suggested DSCP marking of 43.

Then, QCI/5QI 65 is GBR, intended for mission critical PTT voice, with a relative low priority index of 0.7/7. QCI/5QI 65 receives GBR and is intended for mission critical traffic. Its priority is higher (0.7 vs 2) than QCI/5QI 66, but a lower priority (0.7/7 vs 0.5/5) than QCI/5QI 69. Additionally, 65 cannot be represented by DSCP 44 (used by QCI/5QI 1), or DSCP 46 (used by non-GBR voice). As such, QCI/5QI 65 fits between QCIs/5QIs 69 66, with a new suggested DSCP marking of 42.

5.5. Video QCIs and 5QIs [67, 2, 4, 71, 72, 73, 74, 76]

Although six different QCIs and 5QIs have example services that include some form of video traffic, eight QCIs and 5QIs are video in nature, 67, 2, 4, 71, 72, 73, 74, and 76.

All eight QCIs/5QIs represent video streams and fit naturally in the AF4x category. However, these QCIs/5QIs do not match [RFC4594] intent for multimedia conferencing, in that they are all admitted (being associated to a GBR). They also do not match the category described by [RFC5865] for capacity-admitted traffic. Therefore, there is not a clear possible mapping for any of these QCIs and 5QIs to an existing AF4x category. In order to avoid mixing admitted and non-admitted video in the same class, it is necessary to associate these QCIs/5QIs to new Diffserv classes.

In particular, QCI/5QI 67 is GBR, intended for mission-critical video user plane. This QCI/5QI is video in nature, and matches traffic that is rate-adaptive, and real time. 67 priority is high (1.5/15), with a tolerant delay budget (100ms) and rather low loss tolerance ($10.E-3$). 67 is GBR.

As such, it is RECOMMENDED to map QCI/5QI 67 against the DSCP value closest to AF4x video with lowest discard eligibility (AF41), namely DSCP 33.

Similarly, QCI/5QI 2 is intended for conversational video (live streaming). 2 is also video in nature and associated to a GBR, however its priority is lower than 67 (4/40 vs 1.5/15). Additionally, its delay budget is also larger (150 ms vs 100 ms). Its packet error loss is also $10.E-3$. As such, 2 fits well within a video queue, with a larger drop probability than 67. Therefore, it is RECOMMENDED to map QCI/5QI 2 to the video category with a Diffserv marking of 35.

QCIs/5QIs 71, 72, 73, 74 and 76 are intended for "Live" Uplink Streaming (LUS) services, where an end-user with a radio connection (for example a reporter or a drone) streams live video feed into the network or to a second party ([TS 26.939]). This traffic is GBR. However, [TS 26.939] defines LUS and also differentiates GBR from MBR and TBR. At the time of the admission, the infrastructure can offer a Guaranteed Bit Rate, which should match the bare minimum rate expected by the application (and its codec). Because of the burstiness nature of video, the Maximum Bit Rate (MBR) available to the transmission should be much higher than the GBR. In fact, the Target Bit Rate (TBR), which is the preferred service operation point for that application, is likely close to the MBR. Thus, the application will receive a treatment between the GBR and the TBR. This allocated bit rate will directly translate in video quality changes, where an available bit rate close to the GBR will result in a lower Mean Opinion Score than a bit rate close to the TBR. As the application detects the constraints on the available bit rate, it may adapt by changing its codec and compression scheme accordingly. Flows with higher compression will have higher delay tolerance and budget (as a single packet burst represents a larger segment of the video flow) but lower loss tolerance (as each lost packet represents a larger segment of the video flow). As such, 71, 72, 73, 74 and 76 express intents similar to QCI/5QI 2, with additional constraints on the directionality of the flow (upstream only) and the bit rate applied by the infrastructure. These constraints are orthogonal to the intent of the flow. As such, it is RECOMMENDED to map QCIs/5QIs 71, 72, 73, 74 and 76 to the same DSCP value as QCI/5QI 2, and thus to the video category with a Diffserv marking of 35.

QCI/5QI 4 is intended for non-conversational video (buffered streaming), with a priority of 5/50. 4 is also video in nature. Although it is buffered, it is admitted, being associated to a GBR. QCI/5QI 4 as a lower priority than QCIs/5QIs 67 and 2, and a larger delay budget (300 ms vs 150/100). However, its packet loss tolerance is low (10.E-6). This combination makes it eligible for a video category, but with a higher drop probability than 67 and 2. Therefore, it is RECOMMENDED to map QCI/5QI 4 to DSCP 37.

5.6. Live streaming and interactive gaming [7]

QCI/5QI 7 is non-GBR and intended for live streaming voice or video interactive gaming. Its priority is 7/70. It is the only QCI/5QI targeting this particular traffic mix. In the Diffserv model, voice and video are different categories, and are also different from interactive gaming (real time interactive). In the 3GPP model, live streaming video and mission-critical video are defined in other queues with high priority (e.g. QCI or 5QI 2 for video Live streaming, with a priority of 2/20, or QCI/5QI 67 for mission-critical video, with a priority of 1.5/15). By comparison, QCI/5QI 7 priority is relatively low (7/70), with a 100 ms budget delay and a comparatively rather high loss tolerance (10.E-3).

As such, 7 fits well with bursty (e.g. video) and possibly rate adaptive flows, with possible drop probability. It is also non-admitted (non-GBR), and as such, fits close to [RFC4594] intent for multimedia conferencing, with high discard eligibility. Therefore, it is RECOMMENDED to map QCI/5QI 7 to the existing Diffserv category AF43.

5.7. Low latency eMBB and AR/VR [80]

QCI/5QI 80 is intended for low latency eMBB (enhanced Mobile Broadband) applications, such as Augmented Reality or Virtual Reality (AR/VR). 80 priority is 6.8/68, with a low packet delay budget of 10 ms, and a packet error loss rate of at most 10.E-6. 80 is non-GBR, yet intended for real time applications. Traffic in the AR/VR category typically does not react dynamically to losses, requires bandwidth and a low and predictable delay.

As such, QCI/5QI 80 matches closely the specifications for CS4. Therefore, it is RECOMMENDED to map QCI/5QI 80 to the existing category CS4.

5.8. V2X messaging [75,3,9]

Three QCIs/5QIs are intended specifically to carry Vehicle to Anything (V2X) traffic, 75, 3, and 79. All 3 QCIs/5QIs are data in nature, and fit naturally into the AF2x category. However, two of these (75 and 3) are admitted (GBR), and therefore do not fit in the current Diffserv model. 79 is non-admitted, but matches none of the AF2X categories in [RFC4594].

In particular, QCI/5QI 75 is GBR, with a rather high priority (2.5/25), a low delay budget (50 ms), but tolerance to losses ($10E-2$). Being low latency data in nature, 75 fits well in the AF2X category. However, being admitted, it fits none of the existing markings. Being the highest traffic (in priority) in this low latency data family, 75 is recommended to be mapped to a new category, as close as possible to the AF2X class, and with a low drop probability. As such, it is RECOMMENDED to map QCI/5QI 75 to DSCP 17.

Similarly, QCI/5QI 3 is intended for V2X messages, but can also be used for Real time gaming, or Utility traffic (medium voltage distribution) or process automation monitoring. QCI/5QI 3 priority is 3/30. 3 is data in nature, but GBR. Its delay budget is low (50 ms), but with some tolerance to loss ($10E-3$).

QCI/5QI 3 is of the same type as QCI/5QI 75, but with a lower priority. Therefore, 3 should be mapped to a category close to the category to which 75 is mapped, but with a higher drop probability. As such, it is RECOMMENDED to map QCI/5QI 3 to DSCP 19.

Additionally, QCI/5QI 79 is also intended for V2X messages. 79 is similar in nature to 75 and 3, but is non-critical (non-GBR). Its priority is also lower (6.5/65). Its budget delay is similar to that of 75 and 3 (50 ms), and its packet error loss rate is similar to that of 75 ($10E-2$).

79 partially matches AF2X, but is not elastic, and therefore cannot fit exactly in [RFC4594] model. As such, it is recommended to a mapping similar to QCI/5QIs 75 and 3, with a higher drop probability. Therefore, it is RECOMMENDED to map QCI/5QI 79 to DSCP 21.

5.9. Automation and Transport [82, 83, 84, 85, 86]

QCI/5QI 84 is intended for intelligent transport systems. As such, its intent is close to the V2X messaging category. QCI 84 is also admitted (GBR in [TS 23.203] and Delay-Critical GBR in [TS 23.501]). However, 84 is intended for traffic with a smaller packet delay budget (30 ms vs 50 ms for QCI/5QI 75) and a smaller packet error

loss maximum rate (10.E-6 vs 10.E-2 for QCI/5QI 75). As such, 84 should be mapped against a category above that of 75 or 3. Being admitted, 84 does not map easily into an existing category. As such, it is RECOMMENDED to map QCI/5QI 84 to DSCP category 31.

5QI 86 is also intended for intelligent transport systems, and fits in the same general category as 84. 86 is also admitted (Delay-Critical GBR), with a higher priority (18) than 84 but similar burst rate (1354 bytes). 5QI 86 therefore fits into a category close to that of 84. As such, it is RECOMMENDED to map 5QI 86 to DSCP category 29.

QCI/5QI 85 is intended for electricity distribution (high voltage) communication. As such, it is close in intent to QCI/5QI 3. 85 is also GBR. However, 85 priority is lower than that of QCI/5QI 3 (2.1/21 vs 3/30). 85 has also a very low packet delay budget (5 ms vs 50 ms for QCI/5QI 3) and low packet error loss rate (10.E-6 vs 10.E-3 for QCI/5QI 3). As such, 84 should be mapped to a category higher than that of QCI/5QI 3, with a very low drop probability. As such, it is RECOMMENDED to map QCI/5QI 85 to DSCP category 23.

QCIs/5QIs 82 and 83 are both intended for discrete automation control traffic. 82 represents traffic with a higher priority (1.9/19) than traffic matched to 83 (priority 2.2/22). 82 also expects smaller data bursts (255 bytes) than 83 (1358 bytes). However, both QCIs are admitted (GBR), with the same low packet delay budget (10 ms) and packet error loss maximum rate (10.E-4).

As such, 82 and 83 fit in the same general category, with a higher drop probability assigned to 83. They also fit the general intent category of automation traffic types, with a priority higher than that of other M2M traffic types (e.g. V2X messages). As such, they fit well into the AF3X category. However, being both admitted (GBR), they do not easily map to any existing AF3X category, and require new categories.

As such, it is RECOMMENDED to map QCI/5QI 82 to DSCP category 25. Similarly, it is RECOMMENDED to map QCI/5QI 83 to DSCP category 79.

5.10. Non-mission-critical data [6,8,9]

QCIs/5QIs 6, 8 and 8 are intended for non-GBR, Video or TCP data traffic. All 3 QCIs/5QIs are data in nature, non-mission critical, relative low priority and therefore fit naturally into the AF1x category. The inclusion in these QCIs/5QIs' intent of buffered video is an imperfect fit for AF1X. However, the intent of these QCIs/5QIs is to match buffered, and non-mission critical traffic. As such, they match the intent of AF1X, even if the Diffserv model would not

associate buffered video to non-mission critical, buffered and low priority traffic.

The intent of all three QCIs/5QIs is similar. The difference lies in their priority and criticality.

QCI/5QI 6 has priority 6/60, a packet delay budget of 300 ms, and a packet error loss rate of at most 10.E-6. QCI/5QI 8 has a priority 8/80, a packet delay budget of 300 ms, and a packet error loss rate of at most 10.E-6. QCI/5QI 9 has priority 9/90, and also a packet delay budget of 300 ms and a packet error loss rate of at most 10.E-6. As these three QCIs/5QIs represent the same intent and are only different in their priority level, using discard eligibility to differentiate them is logical. As such, it is RECOMMENDED to map QCI/5QI 6 to category AF11. Similarly, it is RECOMMENDED to map QCI/5QI 8 to AF12. And logically, it is RECOMMENDED to map QCI/5QI 9 to AF13.

5.11. Mission-critical data [70]

QCI/5QI 70 is non-GBR, intended for mission critical data, with a priority of 5.5/55, a packet delay budget of 200 ms and a packet error loss rate tolerance of at most 10.E-6. The traffic types intended for 70 are the same as for QCIs/5QIs 6,8,9 categories, namely buffered streaming video and TCP-based traffic, such as www, email, chat, FTP, P2P and other file sharing applications. However, 70 is specifically intended for applications that are mission critical. For this reason, 70 priority is higher than 6, 8 or 9 priorities (5.5/55 vs 6/60, 8/80 and 9/90 respectively). Therefore, 70 fits well in the AF2x family, while 6,8,9 are in AF1x. As 70 displays intermediate differentiated treatment, it also fits well with an intermediate discard eligibility. As such, it is RECOMMENDED to map QCI/5QI 70 to DSCP 20 (AF22).

5.12. Summary of Recommendations for QCI or 5QI to DSCP Mapping

The table below summarizes the 3GPP QCI and 5QI to [RFC4594] DSCP marking recommendations:

QCI/5QI	Resource Type	Priority Level	Example Services	Recommended DSCP (PHB)
1	GBR	2	Conversational Voice	44 (VA)
2	GBR	4	Conversational Video (Live Streaming)	35 (N.A.)

3	GBR	3	Real Time Gaming, V2X messages, Electricity distribution (medium voltage) Process automation (monitoring)	19 (N.A.)
4	GBR	5	Non-Conversational Video (Buffered Streaming)	37 (N.A.)
65	GBR	0.7	Mission Critical user plane Push To Talk voice (e.g., MCPTT)	42 (N.A.)
66	GBR	2	Non-Mission-Critical user plane Push To Talk voice	43 (N.A.)
67	GBR	1.5	Mission Critical Video user plane	33 (N.A.)
75	GBR	2.5	V2X messages	17 (N.A.)
71	GBR	5.6	Live uplink streaming	35 (N.A.)
72	GBR	5.6	Live uplink streaming	35 (N.A.)
73	GBR	5.6	Live uplink streaming	35 (N.A.)
74	GBR	5.6	Live uplink streaming	35 (N.A.)
76	GBR	5.6	Live uplink streaming	35 (N.A.)
82	GBR	1.9	Discrete automation (small packets)	25 (N.A.)
83	GBR	2.2	Discrete automation (large packets)	27 (N.A.)
84	GBR	2.4	Intelligent	31 (N.A.)

				Transport Systems	
85	GBR	2.1		Electricity Distribution - High Voltage	23 (N.A.)
86	GBR	1.8		Intelligent Transport Systems	29 (N.A.)
5	Non-GBR	1		IMS Signalling	40 (CS5)
6	Non-GBR	6		Video (Buffered Streaming) TCP-based (e.g. www, email, chat, ftp, p2p file sharing, progressive video)	10 (AF11)
7	Non-GBR	7		Voice, Video (live streaming), interactive gaming	38 (AF43)
8	Non-GBR	8		Video (buffered streaming) TCP-based (e.g. www, email, chat, ftp, p2p file sharing, progressive video)	12 (AF12)
9	Non-GBR	9		Same as 8	14 (AF13)
69	Non-GBR	0.5		Mission Critical delay sensitive signalling (e.g., MC-PTT signalling, MC Video signalling)	41 (N.A.)
70	Non-GBR	5.5		Mission Critical Data (e.g. example services are the same as QCI 6/8/9)	20 (AF22)
79	Non-GBR	6.5		V2X messages	21 (N.A.)
80	Non-GBR	6.8		Low latency eMMB applications (TCP/UDP-based); augmented reality	32 (CS4)

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6. IANA Considerations

This document has no IANA actions. Although this document suggests the use of codepoints in the Pool 1 of the codespace defined in [RFC2474], no exclusive attribution is requested. The recommended utilisation of seven codepoints in Pool 2 and six codepoints in pool 3 is also intended as a recommendation for experimental or Local Use, as defined in [RFC2474].

7. Specific Security Considerations

The recommendations in this document concern widely deployed wired and wireless network functionality, and, for that reason, do not present additional security concerns that do not already exist in these networks.

8. Security Recommendations for General QoS

It may be possible for a wired or wireless device (which could be either a host or a network device) to mark packets (or map packet markings) in a manner that interferes with or degrades existing QoS policies. Such marking or mapping may be done intentionally or unintentionally by developers and/or users and/or administrators of such devices.

To illustrate: A gaming application designed to run on a smartphone may request that all its packets be marked DSCP EF. Although the 3GPP infrastructure may only allocate a non-GBR default QCI (e.g. QCI 9) for this traffic, the translation point into the Internet domain may consider the DSCP marking instead of the allocated QCI, and forward this traffic with a marking of EF. This traffic may then interfere with QoS policies intended to provide priority services for business voice applications.

To mitigate such scenarios, it is RECOMMENDED to implement general QoS security measures, including:

- o Setting a traffic conditioning policy reflective of business objectives and policy, such that traffic from authorized users and/or applications and/or endpoints will be accepted by the network; otherwise, packet markings will be "bleached" (i.e., re-marked to DSCP DF). Additionally, Section 5 made it clear that it is generally NOT RECOMMENDED to pass through DSCP markings from unauthorized, unidentified and/or unauthenticated devices, as these are typically considered untrusted sources. This is especially relevant for Internet of Things (IoT) deployments,

where tens of billions of devices with little or no security capabilities are being connected to LTE and IP networks, leaving them vulnerable to be utilized as agents for DDoS attacks. These attacks can be amplified with preferential QoS treatments, should the packet markings of such devices be trusted.

- o Policing EF marked packet flows, as detailed in [RFC2474] Section 7 and [RFC3246] Section 3.

Finally, it should be noted that the recommendations put forward in this document are not intended to address all attack vectors leveraging QoS marking abuse. Mechanisms that may further help mitigate security risks of both wired and wireless networks deploying QoS include strong device- and/or user-authentication, access-control, rate-limiting, control-plane policing, encryption, and other techniques; however, the implementation recommendations for such mechanisms are beyond the scope of this document to address in detail. Suffice it to say that the security of the devices and networks implementing QoS, including QoS mapping between wired and wireless networks, merits consideration in actual deployments.

9. References

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Abstract

This specification defines a framework for coupling the Active Queue Management (AQM) algorithms in two queues intended for flows with different responses to congestion. This provides a way for the Internet to transition from the scaling problems of standard TCP Reno-friendly ('Classic') congestion controls to the family of 'Scalable' congestion controls. These are designed for consistently very Low queuing Latency, very Low congestion Loss and Scaling of per-flow throughput (L4S) by using Explicit Congestion Notification (ECN) in a modified way. Until the Coupled DualQ, these L4S senders could only be deployed where a clean-slate environment could be arranged, such as in private data centres. The coupling acts like a semi-permeable membrane: isolating the sub-millisecond average queuing delay and zero congestion loss of L4S from Classic latency and loss; but pooling the capacity between any combination of Scalable and Classic flows with roughly equivalent throughput per flow. The DualQ achieves this indirectly, without having to inspect transport layer flow identifiers and without compromising the performance of the Classic traffic, relative to a single queue. The DualQ design has low complexity and requires no configuration for the public Internet.

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1. Introduction

This document specifies a framework for DualQ Coupled AQMs, which is the network part of the L4S architecture [I-D.ietf-tsvwg-l4s-arch]. L4S enables both very low queuing latency (sub-millisecond on average) and high throughput at the same time, for ad hoc numbers of capacity-seeking applications all sharing the same capacity.

1.1. Outline of the Problem

Latency is becoming the critical performance factor for many (most?) applications on the public Internet, e.g. interactive Web, Web services, voice, conversational video, interactive video, interactive remote presence, instant messaging, online gaming, remote desktop, cloud-based applications, and video-assisted remote control of machinery and industrial processes. In the developed world, further increases in access network bit-rate offer diminishing returns, whereas latency is still a multi-faceted problem. In the last decade or so, much has been done to reduce propagation time by placing caches or servers closer to users. However, queuing remains a major intermittent component of latency.

Traditionally very low latency has only been available for a few selected low rate applications, that confine their sending rate within a specially carved-off portion of capacity, which is prioritized over other traffic, e.g. Diffserv EF [RFC3246]. Up to now it has not been possible to allow any number of low latency, high throughput applications to seek to fully utilize available capacity, because the capacity-seeking process itself causes too much queuing delay.

To reduce this queuing delay caused by the capacity seeking process, changes either to the network alone or to end-systems alone are in progress. L4S involves a recognition that both approaches are yielding diminishing returns:

- * Recent state-of-the-art active queue management (AQM) in the network, e.g. FQ-CoDel [RFC8290], PIE [RFC8033], Adaptive RED [ARED01]) has reduced queuing delay for all traffic, not just a select few applications. However, no matter how good the AQM, the capacity-seeking (sawtooth) rate of TCP-like congestion controls represents a lower limit that will either cause queuing delay to vary or cause the link to be under-utilized. These AQMs are tuned to allow a typical capacity-seeking Reno-friendly flow to induce an average queue that roughly doubles the base RTT, adding 5-15 ms of queuing on average (cf. 500 microseconds with L4S for the same mix of long-running and web traffic). However, for many applications low delay is not useful unless it is consistently low. With these AQMs, 99th percentile queuing delay is 20-30 ms (cf. 2 ms with the same traffic over L4S).
- * Similarly, recent research into using e2e congestion control without needing an AQM in the network (e.g. BBR [I-D.cardwell-iccr-g-bbr-congestion-control]) seems to have hit a similar lower limit to queuing delay of about 20ms on average but there are also regular 25ms delay spikes due to bandwidth probes and 60ms spikes due to flow-starts.

L4S learns from the experience of Data Center TCP [RFC8257], which shows the power of complementary changes both in the network and on end-systems. DCTCP teaches us that two small but radical changes to congestion control are needed to cut the two major outstanding causes of queuing delay variability:

1. Far smaller rate variations (sawteeth) than Reno-friendly congestion controls;
2. A shift of smoothing and hence smoothing delay from network to sender.

Without the former, a 'Classic' (e.g. Reno-friendly) flow's round trip time (RTT) varies between roughly 1 and 2 times the base RTT between the machines in question. Without the latter a 'Classic' flow's response to changing events is delayed by a worst-case (transcontinental) RTT, which could be hundreds of times the actual smoothing delay needed for the RTT of typical traffic from localized CDNs.

These changes are the two main features of the family of so-called 'Scalable' congestion controls (which includes DCTCP, TCP Prague and SCReAM). Both these changes only reduce delay in combination with a complementary change in the network and they are both only feasible with ECN, not drop, for the signalling:

1. The smaller sawteeth allow an extremely shallow ECN packet-marking threshold in the queue.
2. And no smoothing in the network means that every fluctuation of the queue is signalled immediately.

Without ECN, either of these would lead to very high loss levels. But, with ECN, the resulting high marking levels are just signals, not impairments. BBRv2 combines the best of both worlds - it works as a scalable congestion control when ECN is available, but also aims to minimize delay when it isn't.

However, until now, Scalable congestion controls (like DCTCP) did not co-exist well in a shared ECN-capable queue with existing ECN-capable TCP Reno [RFC5681] or Cubic [RFC8312] congestion controls -- Scalable controls are so aggressive that these 'Classic' algorithms would drive themselves to a small capacity share. Therefore, until now, L4S controls could only be deployed where a clean-slate environment could be arranged, such as in private data centres (hence the name DCTCP).

This document specifies a 'DualQ Coupled AQM' extension that solves the problem of coexistence between Scalable and Classic flows, without having to inspect flow identifiers. It is not like flow-queuing approaches [RFC8290] that classify packets by flow identifier into separate queues in order to isolate sparse flows from the higher latency in the queues assigned to heavier flows. If a flow needs both low delay and high throughput, having a queue to itself does not isolate it from the harm it causes to itself. In contrast, DualQ Coupled AQMs address the root cause of the latency problem -- they are an enabler for the smooth low latency scalable behaviour of Scalable congestion controls, so that every packet in every flow can potentially enjoy very low latency, then there would be no need to isolate each flow into a separate queue.

1.2. Scope

L4S involves complementary changes in the network and on end-systems:

Network: A DualQ Coupled AQM (defined in the present document) or a modification to flow-queue AQMs (described in section 4.2.b of the L4S architecture [I-D.ietf-tsvwg-l4s-arch]);

End-system: A Scalable congestion control (defined in section 4 of the L4S ECN protocol [I-D.ietf-tsvwg-ecn-l4s-id]).

Packet identifier: The network and end-system parts of L4S can be deployed incrementally, because they both identify L4S packets using the experimentally assigned explicit congestion notification (ECN) codepoints in the IP header: ECT(1) and CE [RFC8311] [I-D.ietf-tsvwg-ecn-l4s-id].

Data Center TCP (DCTCP [RFC8257]) is an example of a Scalable congestion control for controlled environments that has been deployed for some time in Linux, Windows and FreeBSD operating systems. During the progress of this document through the IETF a number of other Scalable congestion controls were implemented, e.g. TCP Prague [I-D.briscoe-iccrp-prague-congestion-control] [PragueLinux], BBRv2 [BBRv2], [I-D.cardwell-iccrp-bbr-congestion-control], QUIC Prague and the L4S variant of SCREAM for real-time media [RFC8298].

The focus of this specification is to enable deployment of the network part of the L4S service. Then, without any management intervention, applications can exploit this new network capability as their operating systems migrate to Scalable congestion controls, which can then evolve *while* their benefits are being enjoyed by everyone on the Internet.

The DualQ Coupled AQM framework can incorporate any AQM designed for a single queue that generates a statistical or deterministic mark/drop probability driven by the queue dynamics. Pseudocode examples of two different DualQ Coupled AQMs are given in the appendices. In many cases the framework simplifies the basic control algorithm, and requires little extra processing. Therefore it is believed the Coupled AQM would be applicable and easy to deploy in all types of buffers; buffers in cost-reduced mass-market residential equipment; buffers in end-system stacks; buffers in carrier-scale equipment including remote access servers, routers, firewalls and Ethernet switches; buffers in network interface cards, buffers in virtualized network appliances, hypervisors, and so on.

For the public Internet, nearly all the benefit will typically be achieved by deploying the Coupled AQM into either end of the access link between a 'site' and the Internet, which is invariably the bottleneck (see section 6.4 of [I-D.ietf-tsvwg-l4s-arch] about deployment, which also defines the term 'site' to mean a home, an office, a campus or mobile user equipment).

Latency is not the only concern of L4S:

- * The "Low Loss" part of the name denotes that L4S generally achieves zero congestion loss (which would otherwise cause retransmission delays), due to its use of ECN.
- * The "Scalable throughput" part of the name denotes that the per-flow throughput of Scalable congestion controls should scale indefinitely, avoiding the imminent scaling problems with 'TCP-Friendly' congestion control algorithms [RFC3649].

The former is clearly in scope of this AQM document. However, the latter is an outcome of the end-system behaviour, and therefore outside the scope of this AQM document, even though the AQM is an enabler.

The overall L4S architecture [I-D.ietf-tsvwg-l4s-arch] gives more detail, including on wider deployment aspects such as backwards compatibility of Scalable congestion controls in bottlenecks where a DualQ Coupled AQM has not been deployed. The supporting papers [DualPI2Linux], [PI2], [DCTtH19] and [PI2param] give the full rationale for the AQM's design, both discursively and in more precise mathematical form, as well as the results of performance evaluations. The main results have been validated independently when using the Prague congestion control [Boru20] (experiments are run using Prague and DCTCP, but only the former are relevant for validation, because Prague fixes a number of problems with the Linux DCTCP code that make it unsuitable for the public Internet).

1.3. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119] when, and only when, they appear in all capitals, as shown here.

The DualQ Coupled AQM uses two queues for two services. Each of the following terms identifies both the service and the queue that provides the service:

Classic service/queue: The Classic service is intended for all the

congestion control behaviours that co-exist with Reno [RFC5681] (e.g. Reno itself, Cubic [RFC8312], TFRC [RFC5348]).

Low-Latency, Low-Loss Scalable throughput (L4S) service/queue: The 'L4S' service is intended for traffic from scalable congestion control algorithms, such as TCP Prague [I-D.briscoe-iccrp-prague-congestion-control], which was derived from Data Center TCP [RFC8257]. The L4S service is for more general traffic than just TCP Prague -- it allows the set of congestion controls with similar scaling properties to Prague to evolve, such as the examples listed earlier (Relentless, SCReAM, etc.).

Classic Congestion Control: A congestion control behaviour that can co-exist with standard TCP Reno [RFC5681] without causing significantly negative impact on its flow rate [RFC5033]. With Classic congestion controls, such as Reno or Cubic, because flow rate has scaled since TCP congestion control was first designed in 1988, it now takes hundreds of round trips (and growing) to recover after a congestion signal (whether a loss or an ECN mark) as shown in the examples in section 5.1 of the L4S architecture [I-D.ietf-tsvwg-l4s-arch] and in [RFC3649]. Therefore control of queuing and utilization becomes very slack, and the slightest disturbances (e.g. from new flows starting) prevent a high rate from being attained.

Scalable Congestion Control: A congestion control where the average time from one congestion signal to the next (the recovery time) remains invariant as the flow rate scales, all other factors being equal. This maintains the same degree of control over queueing and utilization whatever the flow rate, as well as ensuring that high throughput is robust to disturbances. For instance, DCTCP averages 2 congestion signals per round-trip whatever the flow rate, as do other recently developed scalable congestion controls, e.g. Relentless TCP [Mathis09], TCP Prague [I-D.briscoe-iccrp-prague-congestion-control], [PragueLinux], BBRv2 [BBRv2], [I-D.cardwell-iccrp-bbr-congestion-control] and the L4S variant of SCREAM for real-time media [SCReAM], [RFC8298]). For the public Internet a Scalable transport has to comply with the requirements in Section 4 of [I-D.ietf-tsvwg-ecn-l4s-id] (aka. the 'Prague L4S requirements').

C: Abbreviation for Classic, e.g. when used as a subscript.

L: Abbreviation for L4S, e.g. when used as a subscript.

The terms Classic or L4S can also qualify other nouns, such as 'codepoint', 'identifier', 'classification', 'packet', 'flow'. For example: an L4S packet means a packet with an L4S identifier sent from an L4S congestion control.

Both Classic and L4S services can cope with a proportion of unresponsive or less-responsive traffic as well, but in the L4S case its rate has to be smooth enough or low enough not to build a queue (e.g. DNS, VoIP, game sync datagrams, etc). The DualQ Coupled AQM behaviour is defined to be similar to a single FIFO queue with respect to unresponsive and overload traffic.

Reno-friendly: The subset of Classic traffic that is friendly to the standard Reno congestion control defined for TCP in [RFC5681]. Reno-friendly is used in place of 'TCP-friendly', given the latter has become imprecise, because the TCP protocol is now used with so many different congestion control behaviours, and Reno is used in non-TCP transports such as QUIC.

Classic ECN: The original Explicit Congestion Notification (ECN) protocol [RFC3168], which requires ECN signals to be treated the same as drops, both when generated in the network and when responded to by the sender.

For L4S, the names used for the four codepoints of the 2-bit IP-ECN field are unchanged from those defined in [RFC3168]: Not ECT, ECT(0), ECT(1) and CE, where ECT stands for ECN-Capable Transport and CE stands for Congestion Experienced. A packet marked with the CE codepoint is termed 'ECN-marked' or sometimes just 'marked' where the context makes ECN obvious.

1.4. Features

The AQM couples marking and/or dropping from the Classic queue to the L4S queue in such a way that a flow will get roughly the same throughput whichever it uses. Therefore both queues can feed into the full capacity of a link and no rates need to be configured for the queues. The L4S queue enables Scalable congestion controls like DCTCP or TCP Prague to give very low and predictably low latency, without compromising the performance of competing 'Classic' Internet traffic.

Thousands of tests have been conducted in a typical fixed residential broadband setting. Experiments used a range of base round trip delays up to 100ms and link rates up to 200 Mb/s between the data centre and home network, with varying amounts of background traffic in both queues. For every L4S packet, the AQM kept the average queuing delay below 1ms (or 2 packets where serialization delay

exceeded 1ms on slower links), with 99th percentile no worse than 2ms. No losses at all were introduced by the L4S AQM. Details of the extensive experiments are available [DualPI2Linux], [PI2], [DCTtH19].

In all these experiments, the host was connected to the home network by fixed Ethernet, in order to quantify the queuing delay that can be achieved by a user who cares about delay. It should be emphasized that L4S support at the bottleneck link cannot 'undelay' bursts introduced by another link on the path, for instance by legacy WiFi equipment. However, if L4S support is added to the queue feeding the `_outgoing_` WAN link of a home gateway, it would be counterproductive not to also reduce the burstiness of the `_incoming_` WiFi. Also, trials of WiFi equipment with an L4S DualQ Coupled AQM on the `_outgoing_` WiFi interface are in progress, and early results of an L4S DualQ Coupled AQM in a 5G radio access network testbed with emulated outdoor cell edge radio fading are given in [L4S_5G].

Subjective testing has also been conducted by multiple people all simultaneously using very demanding high bandwidth low latency applications over a single shared access link [L4Sdemo16]. In one application, each user could use finger gestures to pan or zoom their own high definition (HD) sub-window of a larger video scene generated on the fly in 'the cloud' from a football match. Another user wearing VR goggles was remotely receiving a feed from a 360-degree camera in a racing car, again with the sub-window in their field of vision generated on the fly in 'the cloud' dependent on their head movements. Even though other users were also downloading large amounts of L4S and Classic data, playing a gaming benchmark and watchings videos over the same 40Mb/s downstream broadband link, latency was so low that the football picture appeared to stick to the user's finger on the touch pad and the experience fed from the remote camera did not noticeably lag head movements. All the L4S data (even including the downloads) achieved the same very low latency. With an alternative AQM, the video noticeably lagged behind the finger gestures and head movements.

Unlike Diffserv Expedited Forwarding, the L4S queue does not have to be limited to a small proportion of the link capacity in order to achieve low delay. The L4S queue can be filled with a heavy load of capacity-seeking flows (TCP Prague etc.) and still achieve low delay. The L4S queue does not rely on the presence of other traffic in the Classic queue that can be 'overtaken'. It gives low latency to L4S traffic whether or not there is Classic traffic. The tail latency of traffic served by the Classic AQM is sometimes a little better sometimes a little worse, when a proportion of the traffic is L4S.

The two queues are only necessary because:

- * the large variations (sawteeth) of Classic flows need roughly a base RTT of queuing delay to ensure full utilization
- * Scalable flows do not need a queue to keep utilization high, but they cannot keep latency predictably low if they are mixed with Classic traffic,

The L4S queue has latency priority within sub-round trip timescales, but over longer periods the coupling from the Classic to the L4S AQM (explained below) ensures that it does not have bandwidth priority over the Classic queue.

2. DualQ Coupled AQM

There are two main aspects to the approach:

- * The Coupled AQM that addresses throughput equivalence between Classic (e.g. Reno, Cubic) flows and L4S flows (that satisfy the Prague L4S requirements).
- * The Dual Queue structure that provides latency separation for L4S flows to isolate them from the typically large Classic queue.

2.1. Coupled AQM

In the 1990s, the 'TCP formula' was derived for the relationship between the steady-state congestion window, $cwnd$, and the drop probability, p of standard Reno congestion control [RFC5681]. To a first order approximation, the steady-state $cwnd$ of Reno is inversely proportional to the square root of p .

The design focuses on Reno as the worst case, because if it does no harm to Reno, it will not harm Cubic or any traffic designed to be friendly to Reno. TCP Cubic implements a Reno-compatibility mode, which is relevant for typical RTTs under 20ms as long as the throughput of a single flow is less than about 350Mb/s. In such cases it can be assumed that Cubic traffic behaves similarly to Reno. The term 'Classic' will be used for the collection of Reno-friendly traffic including Cubic and potentially other experimental congestion controls intended not to significantly impact the flow rate of Reno.

A supporting paper [PI2] includes the derivation of the equivalent rate equation for DCTCP, for which `cwnd` is inversely proportional to `p` (not the square root), where in this case `p` is the ECN marking probability. DCTCP is not the only congestion control that behaves like this, so the term 'Scalable' will be used for all similar congestion control behaviours (see examples in Section 1.2). The term 'L4S' is used for traffic driven by a Scalable congestion control that also complies with the additional 'Prague L4S' requirements [I-D.ietf-tsvwg-ecn-l4s-id].

For safe co-existence, under stationary conditions, a Scalable flow has to run at roughly the same rate as a Reno TCP flow (all other factors being equal). So the drop or marking probability for Classic traffic, `p_C` has to be distinct from the marking probability for L4S traffic, `p_L`. The original ECN specification [RFC3168] required these probabilities to be the same, but [RFC8311] updates RFC 3168 to enable experiments in which these probabilities are different.

Also, to remain stable, Classic sources need the network to smooth `p_C` so it changes relatively slowly. It is hard for a network node to know the RTTs of all the flows, so a Classic AQM adds a `_worst-case_` RTT of smoothing delay (about 100-200 ms). In contrast, L4S shifts responsibility for smoothing ECN feedback to the sender, which only delays its response by its `_own_` RTT, as well as allowing a more immediate response if necessary.

The Coupled AQM achieves safe coexistence by making the Classic drop probability `p_C` proportional to the square of the coupled L4S probability `p_CL`. `p_CL` is an input to the instantaneous L4S marking probability `p_L` but it changes as slowly as `p_C`. This makes the Reno flow rate roughly equal the DCTCP flow rate, because the squaring of `p_CL` counterbalances the square root of `p_C` in the 'TCP formula' of Classic Reno congestion control.

Stating this as a formula, the relation between Classic drop probability, `p_C`, and the coupled L4S probability `p_CL` needs to take the form:

$$p_C = (p_{CL} / k)^2 \quad (1)$$

where `k` is the constant of proportionality, which is termed the coupling factor.

2.2. Dual Queue

Classic traffic needs to build a large queue to prevent under-utilization. Therefore a separate queue is provided for L4S traffic, and it is scheduled with priority over the Classic queue. Priority is conditional to prevent starvation of Classic traffic in certain conditions (see Section 2.4).

Nonetheless, coupled marking ensures that giving priority to L4S traffic still leaves the right amount of spare scheduling time for Classic flows to each get equivalent throughput to DCTCP flows (all other factors such as RTT being equal).

2.3. Traffic Classification

Both the Coupled AQM and DualQ mechanisms need an identifier to distinguish L4S (L) and Classic (C) packets. Then the coupling algorithm can achieve coexistence without having to inspect flow identifiers, because it can apply the appropriate marking or dropping probability to all flows of each type. A separate specification [I-D.ietf-tsvwg-ecn-l4s-id] requires the network to treat the ECT(1) and CE codepoints of the ECN field as this identifier. An additional process document has proved necessary to make the ECT(1) codepoint available for experimentation [RFC8311].

For policy reasons, an operator might choose to steer certain packets (e.g. from certain flows or with certain addresses) out of the L queue, even though they identify themselves as L4S by their ECN codepoints. In such cases, the L4S ECN protocol [I-D.ietf-tsvwg-ecn-l4s-id] says that the device "MUST NOT alter the end-to-end L4S ECN identifier", so that it is preserved end-to-end. The aim is that each operator can choose how it treats L4S traffic locally, but an individual operator does not alter the identification of L4S packets, which would prevent other operators downstream from making their own choices on how to treat L4S traffic.

In addition, an operator could use other identifiers to classify certain additional packet types into the L queue that it deems will not risk harm to the L4S service. For instance addresses of specific applications or hosts; specific Diffserv codepoints such as EF (Expedited Forwarding), Voice-Admit or the Non-Queue-Building (NQB) per-hop behaviour; or certain protocols (e.g. ARP, DNS) (see Section 5.4.1 of [I-D.ietf-tsvwg-ecn-l4s-id]). Note that the mechanism only reads these identifiers. [I-D.ietf-tsvwg-ecn-l4s-id] says it "MUST NOT alter these non-ECN identifiers". Thus, the L queue is not solely an L4S queue, it can be considered more generally as a low latency queue.

2.4. Overall DualQ Coupled AQM Structure

Figure 1 shows the overall structure that any DualQ Coupled AQM is likely to have. This schematic is intended to aid understanding of the current designs of DualQ Coupled AQMs. However, it is not intended to preclude other innovative ways of satisfying the normative requirements in Section 2.5 that minimally define a DualQ Coupled AQM. Also, the schematic only illustrates operation under normally expected circumstances; behaviour under overload or with operator-specific classifiers is deferred to Section 2.5.1.1.

The classifier on the left separates incoming traffic between the two queues (L and C). Each queue has its own AQM that determines the likelihood of marking or dropping (p_L and p_C). It has been proved [PI2] that it is preferable to control load with a linear controller, then square the output before applying it as a drop probability to Reno-friendly traffic (because Reno congestion control decreases its load proportional to the square-root of the increase in drop). So, the AQM for Classic traffic needs to be implemented in two stages: i) a base stage that outputs an internal probability p' (pronounced p-prime); and ii) a squaring stage that outputs p_C , where

$$p_C = (p')^2. \quad (2)$$

Substituting for p_C in Eqn (1) gives:

$$p' = p_{CL} / k$$

So the slow-moving input to ECN marking in the L queue (the coupled L4S probability) is:

$$p_{CL} = k * p'. \quad (3)$$

The actual ECN marking probability p_L that is applied to the L queue needs to track the immediate L queue delay under L-only congestion conditions, as well as track p_{CL} under coupled congestion conditions. So the L queue uses a native AQM that calculates a probability p'_L as a function of the instantaneous L queue delay. And, given the L queue has conditional priority over the C queue, whenever the L queue grows, the AQM ought to apply marking probability p'_L , but p_L ought not to fall below p_{CL} . This suggests:

$$p_L = \max(p'_L, p_{CL}), \quad (4)$$

which has also been found to work very well in practice.

The two transformations of p' in equations (2) and (3) implement the required coupling given in equation (1) earlier.

The constant of proportionality or coupling factor, k , in equation (1) determines the ratio between the congestion probabilities (loss or marking) experienced by L4S and Classic traffic. Thus k indirectly determines the ratio between L4S and Classic flow rates, because flows (assuming they are responsive) adjust their rate in response to congestion probability. Appendix C.2 gives guidance on the choice of k and its effect on relative flow rates.

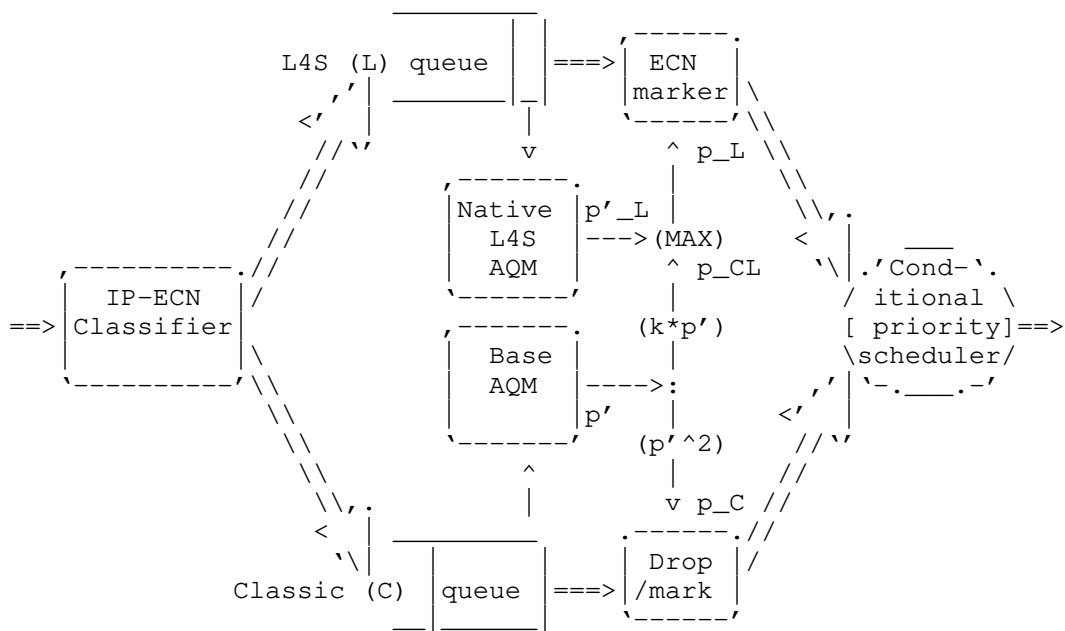


Figure 1: DualQ Coupled AQM Schematic

Legend: ==> traffic flow; ---> control dependency.

After the AQMs have applied their dropping or marking, the scheduler forwards their packets to the link. Even though the scheduler gives priority to the L queue, it is not as strong as the coupling from the C queue. This is because, as the C queue grows, the base AQM applies more congestion signals to L traffic (as well as C). As L flows reduce their rate in response, they use less than the scheduling share for L traffic. So, because the scheduler is work preserving, it schedules any C traffic in the gaps.

Giving priority to the L queue has the benefit of very low L queue delay, because the L queue is kept empty whenever L traffic is controlled by the coupling. Also there only has to be a coupling in one direction - from Classic to L4S. Priority has to be conditional in some way to prevent the C queue being starved in the short-term (see Section 4.2.2) to give C traffic a means to push in, as explained next. With normal responsive L traffic, the coupled ECN marking gives C traffic the ability to push back against even strict priority, by congestion marking the L traffic to make it yield some space. However, if there is just a small finite set of C packets (e.g. a DNS request or an initial window of data) some Classic AQMs will not induce enough ECN marking in the L queue, no matter how long the small set of C packets waits. Then, if the L queue happens to remain busy, the C traffic would never get a scheduling opportunity from a strict priority scheduler. Ideally the Classic AQM would be designed to increase the coupled marking the longer that C packets have been waiting, but this is not always practical - hence the need for L priority to be conditional. Giving a small weight or limited waiting time for C traffic improves response times for short Classic messages, such as DNS requests, and improves Classic flow startup because immediate capacity is available.

Example DualQ Coupled AQM algorithms called DualPI2 and Curvy RED are given in Appendix A and Appendix B. Either example AQM can be used to couple packet marking and dropping across a dual Q.

DualPI2 uses a Proportional-Integral (PI) controller as the Base AQM. Indeed, this Base AQM with just the squared output and no L4S queue can be used as a drop-in replacement for PIE [RFC8033], in which case it is just called PI2 [PI2]. PI2 is a principled simplification of PIE that is both more responsive and more stable in the face of dynamically varying load.

Curvy RED is derived from RED [RFC2309], except its configuration parameters are delay-based to make them insensitive to link rate and it requires less operations per packet than RED. However, DualPI2 is more responsive and stable over a wider range of RTTs than Curvy RED. As a consequence, at the time of writing, DualPI2 has attracted more development and evaluation attention than Curvy RED, leaving the Curvy RED design not so fully evaluated.

Both AQMs regulate their queue in units of time rather than bytes. As already explained, this ensures configuration can be invariant for different drain rates. With AQMs in a dualQ structure this is particularly important because the drain rate of each queue can vary rapidly as flows for the two queues arrive and depart, even if the combined link rate is constant.

It would be possible to control the queues with other alternative AQMs, as long as the normative requirements (those expressed in capitals) in Section 2.5 are observed.

The two queues could optionally be part of a larger queuing hierarchy, such as the initial example ideas in [I-D.briscoe-tsvwg-l4s-diffserv].

2.5. Normative Requirements for a DualQ Coupled AQM

The following requirements are intended to capture only the essential aspects of a DualQ Coupled AQM. They are intended to be independent of the particular AQMs used for each queue.

2.5.1. Functional Requirements

A Dual Queue Coupled AQM implementation **MUST** comply with the prerequisite L4S behaviours for any L4S network node (not just a DualQ) as specified in section 5 of [I-D.ietf-tsvwg-ecn-l4s-id]. These primarily concern classification and remarking as briefly summarized in Section 2.3 earlier. But there is also a subsection (5.5) giving guidance on reducing the burstiness of the link technology underlying any L4S AQM.

A Dual Queue Coupled AQM implementation **MUST** utilize two queues, each with an AQM algorithm.

The AQM algorithm for the low latency (L) queue **MUST** be able to apply ECN marking to ECN-capable packets.

The scheduler draining the two queues **MUST** give L4S packets priority over Classic, although priority **MUST** be bounded in order not to starve Classic traffic (see Section 4.2.2). The scheduler **SHOULD** be work-conserving, or otherwise close to work-conserving. This is because Classic traffic needs to be able to efficiently fill any space left by L4S traffic even though the scheduler would otherwise allocate it to L4S.

[I-D.ietf-tsvwg-ecn-l4s-id] defines the meaning of an ECN marking on L4S traffic, relative to drop of Classic traffic. In order to ensure coexistence of Classic and Scalable L4S traffic, it says, "The likelihood that an AQM drops a Not-ECT Classic packet (p_C) **MUST** be roughly proportional to the square of the likelihood that it would have marked it if it had been an L4S packet (p_L)." The term 'likelihood' is used to allow for marking and dropping to be either probabilistic or deterministic.

For the current specification, this translates into the following requirement. A DualQ Coupled AQM MUST apply ECN marking to traffic in the L queue that is no lower than that derived from the likelihood of drop (or ECN marking) in the Classic queue using Eqn. (1).

The constant of proportionality, k , in Eqn (1) determines the relative flow rates of Classic and L4S flows when the AQM concerned is the bottleneck (all other factors being equal). The L4S ECN protocol [I-D.ietf-tsvwg-ecn-l4s-id] says, "The constant of proportionality (k) does not have to be standardised for interoperability, but a value of 2 is RECOMMENDED."

Assuming Scalable congestion controls for the Internet will be as aggressive as DCTCP, this will ensure their congestion window will be roughly the same as that of a standards track TCP Reno congestion control (Reno) [RFC5681] and other Reno-friendly controls, such as TCP Cubic in its Reno-compatibility mode.

The choice of k is a matter of operator policy, and operators MAY choose a different value using the guidelines in Appendix C.2.

If multiple customers or users share capacity at a bottleneck (e.g. in the Internet access link of a campus network), the operator's choice of k will determine capacity sharing between the flows of different customers. However, on the public Internet, access network operators typically isolate customers from each other with some form of layer-2 multiplexing (OFDM(A) in DOCSIS3.1, CDMA in 3G, SC-FDMA in LTE) or L3 scheduling (WRR in DSL), rather than relying on host congestion controls to share capacity between customers [RFC0970]. In such cases, the choice of k will solely affect relative flow rates within each customer's access capacity, not between customers. Also, k will not affect relative flow rates at any times when all flows are Classic or all flows are L4S, and it will not affect the relative throughput of small flows.

2.5.1.1. Requirements in Unexpected Cases

The flexibility to allow operator-specific classifiers (Section 2.3) leads to the need to specify what the AQM in each queue ought to do with packets that do not carry the ECN field expected for that queue. It is expected that the AQM in each queue will inspect the ECN field to determine what sort of congestion notification to signal, then it will decide whether to apply congestion notification to this particular packet, as follows:

- * If a packet that does not carry an ECT(1) or CE codepoint is classified into the L queue:

- if the packet is ECT(0), the L AQM SHOULD apply CE-marking using a probability appropriate to Classic congestion control and appropriate to the target delay in the L queue
- if the packet is Not-ECT, the appropriate action depends on whether some other function is protecting the L queue from misbehaving flows (e.g. per-flow queue protection [I-D.briscoe-docsis-q-protection] or latency policing):
 - o If separate queue protection is provided, the L AQM SHOULD ignore the packet and forward it unchanged, meaning it should not calculate whether to apply congestion notification and it should neither drop nor CE-mark the packet (for instance, the operator might classify EF traffic that is unresponsive to drop into the L queue, alongside responsive L4S-ECN traffic)
 - o if separate queue protection is not provided, the L AQM SHOULD apply drop using a drop probability appropriate to Classic congestion control and appropriate to the target delay in the L queue
- * If a packet that carries an ECT(1) codepoint is classified into the C queue:
 - the C AQM SHOULD apply CE-marking using the coupled AQM probability p_{CL} ($= k \cdot p'$).

The above requirements are worded as "SHOULDs", because operator-specific classifiers are for flexibility, by definition. Therefore, alternative actions might be appropriate in the operator's specific circumstances. An example would be where the operator knows that certain legacy traffic marked with one codepoint actually has a congestion response associated with another codepoint.

If the DualQ Coupled AQM has detected overload, it MUST introduce Classic drop to both types of ECN-capable traffic until the overload episode has subsided. Introducing drop if ECN marking is persistently high is recommended by Section 7 of the ECN specification [RFC3168] and Section 4.2.1 of the AQM Recommendations [RFC7567].

2.5.2. Management Requirements

2.5.2.1. Configuration

By default, a DualQ Coupled AQM SHOULD NOT need any configuration for use at a bottleneck on the public Internet [RFC7567]. The following parameters MAY be operator-configurable, e.g. to tune for non-Internet settings:

- * Optional packet classifier(s) to use in addition to the ECN field (see Section 2.3);
- * Expected typical RTT, which can be used to determine the queuing delay of the Classic AQM at its operating point, in order to prevent typical lone flows from under-utilizing capacity. For example:
 - for the PI2 algorithm (Appendix A) the queuing delay target is dependent on the typical RTT;
 - for the Curvy RED algorithm (Appendix B) the queuing delay at the desired operating point of the curvy ramp is configured to encompass a typical RTT;
 - if another Classic AQM was used, it would be likely to need an operating point for the queue based on the typical RTT, and if so it SHOULD be expressed in units of time.

An operating point that is manually calculated might be directly configurable instead, e.g. for links with large numbers of flows where under-utilization by a single flow would be unlikely.

- * Expected maximum RTT, which can be used to set the stability parameter(s) of the Classic AQM. For example:
 - for the PI2 algorithm (Appendix A), the gain parameters of the PI algorithm depend on the maximum RTT.
 - for the Curvy RED algorithm (Appendix B) the smoothing parameter is chosen to filter out transients in the queue within a maximum RTT.

Stability parameter(s) that are manually calculated assuming a maximum RTT might be directly configurable instead.

- * Coupling factor, k (see Appendix C.2);
- * A limit to the conditional priority of L4S. This is scheduler-dependent, but it SHOULD be expressed as a relation between the max delay of a C packet and an L packet. For example:

- for a WRR scheduler a weight ratio between L and C of $w:1$ means that the maximum delay to a C packet is w times that of an L packet.
 - for a time-shifted FIFO (TS-FIFO) scheduler (see Section 4.2.2) a time-shift of $tshift$ means that the maximum delay to a C packet is $tshift$ greater than that of an L packet. $tshift$ could be expressed as a multiple of the typical RTT rather than as an absolute delay.
- * The maximum Classic ECN marking probability, p_{Cmax} , before introducing drop.

2.5.2.2. Monitoring

An experimental DualQ Coupled AQM SHOULD allow the operator to monitor each of the following operational statistics on demand, per queue and per configurable sample interval, for performance monitoring and perhaps also for accounting in some cases:

- * Bits forwarded, from which utilization can be calculated;
- * Total packets in the three categories: arrived, presented to the AQM, and forwarded. The difference between the first two will measure any non-AQM tail discard. The difference between the last two will measure proactive AQM discard;
- * ECN packets marked, non-ECN packets dropped, ECN packets dropped, which can be combined with the three total packet counts above to calculate marking and dropping probabilities;
- * Queue delay (not including serialization delay of the head packet or medium acquisition delay) - see further notes below.

Unlike the other statistics, queue delay cannot be captured in a simple accumulating counter. Therefore the type of queue delay statistics produced (mean, percentiles, etc.) will depend on implementation constraints. To facilitate comparative evaluation of different implementations and approaches, an implementation SHOULD allow mean and 99th percentile queue delay to be derived (per queue per sample interval). A relatively simple way to do this would be to store a coarse-grained histogram of queue delay. This could be done with a small number of bins with configurable edges that represent contiguous ranges of queue delay. Then, over a sample interval, each bin would accumulate a count of the number of packets that had fallen within each range. The maximum queue delay per queue per interval MAY also be recorded, to aid diagnosis of faults and anomalous events.

2.5.2.3. Anomaly Detection

An experimental DualQ Coupled AQM SHOULD asynchronously report the following data about anomalous conditions:

- * Start-time and duration of overload state.

A hysteresis mechanism SHOULD be used to prevent flapping in and out of overload causing an event storm. For instance, exit from overload state could trigger one report, but also latch a timer. Then, during that time, if the AQM enters and exits overload state any number of times, the duration in overload state is accumulated but no new report is generated until the first time the AQM is out of overload once the timer has expired.

2.5.2.4. Deployment, Coexistence and Scaling

[RFC5706] suggests that deployment, coexistence and scaling should also be covered as management requirements. The raison d'être of the DualQ Coupled AQM is to enable deployment and coexistence of Scalable congestion controls - as incremental replacements for today's Reno-friendly controls that do not scale with bandwidth-delay product. Therefore there is no need to repeat these motivating issues here given they are already explained in the Introduction and detailed in the L4S architecture [I-D.ietf-tsvwg-l4s-arch].

The descriptions of specific DualQ Coupled AQM algorithms in the appendices cover scaling of their configuration parameters, e.g. with respect to RTT and sampling frequency.

3. IANA Considerations (to be removed by RFC Editor)

This specification contains no IANA considerations.

4. Security Considerations

4.1. Low Delay without Requiring Per-Flow Processing

The L4S architecture [I-D.ietf-tsvwg-l4s-arch] compares the DualQ and per-flow-queuing (FQ) approaches to L4S. The privacy considerations section in that document motivates the DualQ on the grounds that users who want to encrypt application flow identifiers, e.g. in IPSec or other encrypted VPN tunnels, don't have to sacrifice low delay ([RFC8404] encourages avoidance of such privacy compromises).

The security considerations section of the L4S architecture also includes subsections on policing of relative flow-rates (section 8.1) and on policing of flows that cause excessive queuing delay (section 8.2). It explains that the interests of users do not collide in the same way for delay as they do for bandwidth. For someone to get more of the bandwidth of a shared link, someone else necessarily gets less (a 'zero-sum game'), whereas queuing delay can be reduced for everyone, without any need for someone else to lose out. It also explains that, on the current Internet, scheduling usually enforces separation between 'sites' (e.g. households, businesses or mobile users), but it is not common to need to schedule or police individual application flows.

By the above arguments, per-flow policing might not be necessary and in trusted environments it is certainly unlikely to be needed. Therefore, because it is hard to avoid complexity and unintended side-effects with per-flow policing, it needs to be separable from a basic AQM, as an option, under policy control. On this basis, the DualQ Coupled AQM provides low delay without prejudging the question of per-flow policing.

Nonetheless, the interests of users or flows might conflict, e.g. in case of accident or malice. Then per-flow control could be necessary. If flow-rate control is needed, it can be provided as a modular addition to a DualQ. And similarly, if protection against excessive queue delay is needed, a per-flow queue protection option can be added to a DualQ (e.g. [I-D.briscoe-docsis-q-protection]).

4.2. Handling Unresponsive Flows and Overload

In the absence of any per-flow control, it is important that the basic DualQ Coupled AQM gives unresponsive flows no more throughput advantage than a single-queue AQM would, and that it at least handles overload situations. Overload means that incoming load significantly or persistently exceeds output capacity, but it is not intended to be a precise term -- significant and persistent are matters of degree.

A trade-off needs to be made between complexity and the risk of either traffic class harming the other. In overloaded conditions the higher priority L4S service will have to sacrifice some aspect of its performance. Depending on the degree of overload, alternative solutions may relax a different factor: e.g. throughput, delay, drop. These choices need to be made either by the developer or by operator policy, rather than by the IETF. Subsequent subsections discuss aspects relating to handling of different degrees of overload:

- * Unresponsive flows (L and/or C) but not overloaded, i.e. the sum of unresponsive load before adding any responsive traffic is below capacity;

This case is handled by the regular Coupled DualQ (Section 2.1) but not discussed there. So below, Section 4.2.1 explains the design goal, and how it is achieved in practice;

- * Unresponsive flows (L and/or C) causing persistent overload, i.e. the sum of unresponsive load even before adding any responsive traffic persistently exceeds capacity;

This case is not covered by the regular Coupled DualQ mechanism (Section 2.1) but the last para in Section 2.5.1.1 sets out a requirement to handle the case where ECN-capable traffic could starve non-ECN-capable traffic. Section 4.2.3 below discusses the general options and gives specific examples.

- * Short-term overload that lies between the 'not overloaded' and 'persistently overloaded' cases.

For the period before overload is deemed persistent, Section 4.2.2 discusses options for more immediate mechanisms at the scheduler timescale. These prevent short-term starvation of the C queue by making the priority of the L queue conditional, as required in Section 2.5.1.

4.2.1. Unresponsive Traffic without Overload

When one or more L flows and/or C flows are unresponsive, but their total load is within the link capacity so that they do not saturate the coupled marking (below 100%), the goal of a DualQ AQM is to behave no worse than a single-queue AQM.

Tests have shown that this is indeed the case with no additional mechanism beyond the regular Coupled DualQ of Section 2.1 (see the results of 'overload experiments' in [DCttH19]). Perhaps counter-intuitively, whether the unresponsive flow classifies itself into the L or the C queue, the DualQ system behaves as if it has subtracted from the overall link capacity. Then, the coupling shares out the remaining capacity between any competing responsive flows (in either queue). See also Section 4.2.2, which discusses scheduler-specific details.

4.2.2. Avoiding Short-Term Classic Starvation: Sacrifice L4S Throughput or Delay?

Priority of L4S is required to be conditional (see Section 2.4 & Section 2.5.1) to avoid short-term starvation of Classic. Otherwise, as explained in Section 2.4, even a lone responsive L4S flow could temporarily block a small finite set of C packets (e.g. an initial window or DNS request). The blockage would only be brief, but it could be longer for certain AQM implementations that can only increase the congestion signal coupled from the C queue when C packets are actually being dequeued. There is then the question of whether to sacrifice L4S throughput or L4S delay (or some other policy) to make the priority conditional:

Sacrifice L4S throughput: By using weighted round robin as the conditional priority scheduler, the L4S service can sacrifice some throughput during overload. This can either be thought of as guaranteeing a minimum throughput service for Classic traffic, or as guaranteeing a maximum delay for a packet at the head of the Classic queue.

Cautionary note: a WRR scheduler can only guarantee Classic throughput if Classic sources are sending enough to use it -- congestion signals can undermine scheduling because they determine how much responsive traffic of each class arrives for scheduling in the first place. This is why scheduling is only relied on to handle short-term starvation; until congestion signals build up and the sources react. Even during long-term overload (discussed more fully in Section 4.2.3), it's pragmatic to discard packets from both queues, which again thins the traffic before it reaches the scheduler. This is because a scheduler cannot be relied on to handle long-term overload since the right scheduler weight cannot be known for every scenario.

The scheduling weight of the Classic queue should be small (e.g. 1/16). In most traffic scenarios the scheduler will not interfere and it will not need to, because the coupling mechanism and the end-systems will determine the share of capacity across both queues as if it were a single pool. However, if L4S traffic is over-aggressive or unresponsive, the scheduler weight for Classic traffic will at least be large enough to ensure it does not starve in the short-term.

Although WRR scheduling is only expected to address short-term overload, there are (somewhat rare) cases when WRR has an effect on capacity shares over longer time-scales. But its effect is minor, and it certainly does no harm. Specifically, in cases where the ratio of L4S to Classic flows (e.g. 19:1) is greater

than the ratio of their scheduler weights (e.g. 15:1), the L4S flows will get less than an equal share of the capacity, but only slightly. For instance, with the example numbers given, each L4S flow will get $(15/16)/19 = 4.9\%$ when ideally each would get $1/20=5\%$. In the rather specific case of an unresponsive flow taking up just less than the capacity set aside for L4S (e.g. 14/16 in the above example), using WRR could significantly reduce the capacity left for any responsive L4S flows.

The scheduling weight of the Classic queue should not be too small, otherwise a C packet at the head of the queue could be excessively delayed by a continually busy L queue. For instance if the Classic weight is 1/16, the maximum that a Classic packet at the head of the queue can be delayed by L traffic is the serialization delay of 15 MTU-sized packets.

Sacrifice L4S Delay: The operator could choose to control overload of the Classic queue by allowing some delay to 'leak' across to the L4S queue. The scheduler can be made to behave like a single First-In First-Out (FIFO) queue with different service times by implementing a very simple conditional priority scheduler that could be called a "time-shifted FIFO" (see the Modifier Earliest Deadline First (MEDF) scheduler [MEDF]). This scheduler adds t_{shift} to the queue delay of the next L4S packet, before comparing it with the queue delay of the next Classic packet, then it selects the packet with the greater adjusted queue delay.

Under regular conditions, this time-shifted FIFO scheduler behaves just like a strict priority scheduler. But under moderate or high overload it prevents starvation of the Classic queue, because the time-shift (t_{shift}) defines the maximum extra queuing delay of Classic packets relative to L4S. This would control milder overload of responsive traffic by introducing delay to defer invoking the overload mechanisms in Section 4.2.3, particularly when close to the maximum congestion signal.

The example implementations in Appendix A and Appendix B could both be implemented with either policy.

4.2.3. L4S ECN Saturation: Introduce Drop or Delay?

This section concerns persistent overload caused by unresponsive L and/or C flows. To keep the throughput of both L4S and Classic flows roughly equal over the full load range, a different control strategy needs to be defined above the point where the L4S AQM persistently saturates to an ECN marking probability of 100% leaving no room to push back the load any harder. L4S ECN marking will saturate first (assuming the coupling factor $k>1$), even though saturation could be

caused by the sum of unresponsive traffic in either or both queues exceeding the link capacity.

The term 'unresponsive' includes cases where a flow becomes temporarily unresponsive, for instance, a real-time flow that takes a while to adapt its rate in response to congestion, or a standard Reno flow that is normally responsive, but above a certain congestion level it will not be able to reduce its congestion window below the allowed minimum of 2 segments [RFC5681], effectively becoming unresponsive. (Note that L4S traffic ought to remain responsive below a window of 2 segments (see the L4S requirements [I-D.ietf-tsvwg-ecn-l4s-id])).

Saturation raises the question of whether to relieve congestion by introducing some drop into the L4S queue or by allowing delay to grow in both queues (which could eventually lead to drop due to buffer exhaustion anyway):

Drop on Saturation: Persistent saturation can be defined by a maximum threshold for coupled L4S ECN marking (assuming $k > 1$) before saturation starts to make the flow rates of the different traffic types diverge. Above that, the drop probability of Classic traffic is applied to all packets of all traffic types. Then experiments have shown that queueing delay can be kept at the target in any overload situation, including with unresponsive traffic, and no further measures are required (Section 4.2.3.1).

Delay on Saturation: When L4S marking saturates, instead of introducing L4S drop, the drop and marking probabilities of both queues could be capped. Beyond that, delay will grow either solely in the queue with unresponsive traffic (if WRR is used), or in both queues (if time-shifted FIFO is used). In either case, the higher delay ought to control temporary high congestion. If the overload is more persistent, eventually the combined DualQ will overflow and tail drop will control congestion.

The example implementation in Appendix A solely applies the "drop on saturation" policy. The DOCSIS specification of a DualQ Coupled AQM [DOCSIS3.1] also implements the 'drop on saturation' policy with a very shallow L buffer. However, the addition of DOCSIS per-flow Queue Protection [I-D.briscoe-docsis-q-protection] turns this into 'delay on saturation' by redirecting some packets of the flow(s) most responsible for L queue overload into the C queue, which has a higher delay target. If overload continues, this again becomes 'drop on saturation' as the level of drop in the C queue rises to maintain the target delay of the C queue.

4.2.3.1. Protecting against Overload by Unresponsive ECN-Capable Traffic

Without a specific overload mechanism, unresponsive traffic would have a greater advantage if it were also ECN-capable. The advantage is undetectable at normal low levels of marking. However, it would become significant with the higher levels of marking typical during overload, when it could evade a significant degree of drop. This is an issue whether the ECN-capable traffic is L4S or Classic.

This raises the question of whether and when to introduce drop of ECN-capable traffic, as required by both Section 7 of the ECN spec [RFC3168] and Section 4.2.1 of the AQM recommendations [RFC7567].

As an example, experiments with the DualPI2 AQM (Appendix A) have shown that introducing 'drop on saturation' at 100% coupled L4S marking addresses this problem with unresponsive ECN as well as addressing the saturation problem. At saturation, DualPI2 switches into overload mode, where the base AQM is driven by the max delay of both queues and it introduces probabilistic drop to both queues equally. It leaves only a small range of congestion levels just below saturation where unresponsive traffic gains any advantage from using the ECN capability (relative to being unresponsive without ECN), and the advantage is hardly detectable (see [DualQ-Test] and section IV-E of [DCttH19]). Also overload with an unresponsive ECT(1) flow gets no more bandwidth advantage than with ECT(0).

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6. Contributors

The following contributed implementations and evaluations that validated and helped to improve this specification:

Olga Albisser <olga@albisser.org> of Simula Research Lab, Norway (Olga Bondarenko during early drafts) implemented the prototype DualPI2 AQM for Linux with Koen De Schepper and conducted extensive evaluations as well as implementing the live performance visualization GUI [L4Sdemo16].

Olivier Tilmans <olivier.tilmans@nokia-bell-labs.com> of Nokia Bell Labs, Belgium prepared and maintains the Linux implementation of DualPI2 for upstreaming.

Shravya K.S. wrote a model for the ns-3 simulator based on the -01 version of this Internet-Draft. Based on this initial work, Tom Henderson <tomh@tomh.org> updated that earlier model and created a model for the DualQ variant specified as part of the Low Latency DOCSIS specification, as well as conducting extensive evaluations.

Ing Jyh (Inton) Tsang of Nokia, Belgium built the End-to-End Data Centre to the Home broadband testbed on which DualQ Coupled AQM implementations were tested.

7. References

7.1. Normative References

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Appendix A. Example DualQ Coupled PI2 Algorithm

As a first concrete example, the pseudocode below gives the DualPI2 algorithm. DualPI2 follows the structure of the DualQ Coupled AQM framework in Figure 1. A simple ramp function (configured in units of queuing time) with unsmoothed ECN marking is used for the Native L4S AQM. The ramp can also be configured as a step function. The PI2 algorithm [PI2] is used for the Classic AQM. PI2 is an improved variant of the PIE AQM [RFC8033].

The pseudocode will be introduced in two passes. The first pass explains the core concepts, deferring handling of edge-cases like overload to the second pass. To aid comparison, line numbers are kept in step between the two passes by using letter suffixes where the longer code needs extra lines.

All variables are assumed to be floating point in their basic units (size in bytes, time in seconds, rates in bytes/second, alpha and beta in Hz, and probabilities from 0 to 1. Constants expressed in k (kilo), M (mega), G (giga), u (micro), m (milli) , %, ... are assumed to be converted to their appropriate multiple or fraction to represent the basic units. A real implementation that wants to use integer values needs to handle appropriate scaling factors and allow accordingly appropriate resolution of its integer types (including temporary internal values during calculations).

A full open source implementation for Linux is available at: https://github.com/L4STeam/sch_dualpi2_upstream and explained in [DualPI2Linux]. The specification of the DualQ Coupled AQM for DOCSIS cable modems and CMTSS is available in [DOCSIS3.1] and explained in [LLD].

A.1. Pass #1: Core Concepts

The pseudocode manipulates three main structures of variables: the packet (pkt), the L4S queue (lq) and the Classic queue (cq). The pseudocode consists of the following six functions:

- * The initialization function `dualpi2_params_init(...)` (Figure 2) that sets parameter defaults (the API for setting non-default values is omitted for brevity)
- * The enqueue function `dualpi2_enqueue(lq, cq, pkt)` (Figure 3)
- * The dequeue function `dualpi2_dequeue(lq, cq, pkt)` (Figure 4)
- * The recurrence function `recur(q, likelihood)` for de-randomized ECN marking (shown at the end of Figure 4).
- * The L4S AQM function `laqm(qdelay)` (Figure 5) used to calculate the ECN-marking probability for the L4S queue
- * The base AQM function that implements the PI algorithm `dualpi2_update(lq, cq)` (Figure 6) used to regularly update the base probability (p'), which is squared for the Classic AQM as well as being coupled across to the L4S queue.

It also uses the following functions that are not shown in full here:

- * `scheduler()`, which selects between the head packets of the two queues; the choice of scheduler technology is discussed later;
- * `cq.bytest()` or `lq.bytest()` returns the current length (aka. backlog) of the relevant queue in bytes;
- * `cq.len()` or `lq.len()` returns the current length of the relevant queue in packets;
- * `cq.time()` or `lq.time()` returns the current queuing delay (aka. sojourn time or service time) of the relevant queue in units of time (see Note a);
- * `mark(pkt)` and `drop(pkt)` for ECN-marking and dropping a packet;

In experiments so far (building on experiments with PIE) on broadband access links ranging from 4 Mb/s to 200 Mb/s with base RTTs from 5 ms to 100 ms, DualPI2 achieves good results with the default parameters in Figure 2. The parameters are categorised by whether they relate to the Base PI2 AQM, the L4S AQM or the framework coupling them together. Constants and variables derived from these parameters are also included at the end of each category. Each parameter is explained as it is encountered in the walk-through of the pseudocode below, and the rationale for the chosen defaults are given so that sensible values can be used in scenarios other than the regular public Internet.

```

1:  dualpi2_params_init(...) {           % Set input parameter defaults
2:      % DualQ Coupled framework parameters
5:      limit = MAX_LINK_RATE * 250 ms    % Dual buffer size
3:      k = 2                             % Coupling factor
4:      % NOT SHOWN % scheduler-dependent weight or equival't parameter
6:
7:      % PI2 Classic AQM parameters
8:      target = 15 ms                     % Queue delay target
9:      RTT_max = 100 ms                   % Worst case RTT expected
10:     % PI2 constants derived from above PI2 parameters
11:     p_Cmax = min(1/k^2, 1)              % Max Classic drop/mark prob
12:     Tupdate = min(target, RTT_max/3)    % PI sampling interval
13:     alpha = 0.1 * Tupdate / RTT_max^2   % PI integral gain in Hz
14:     beta = 0.3 / RTT_max                % PI proportional gain in Hz
15:
16:     % L4S ramp AQM parameters
17:     minTh = 800 us                      % L4S min marking threshold in time units
18:     range = 400 us                      % Range of L4S ramp in time units
19:     Th_len = 1 pkt                      % Min L4S marking threshold in packets
20:     % L4S constants
21:     p_Lmax = 1                          % Max L4S marking prob
22: }
```

Figure 2: Example Header Pseudocode for DualQ Coupled PI2 AQM

The overall goal of the code is to apply the marking and dropping probabilities for L4S and Classic traffic (p_L and p_C). These are derived from the underlying base probabilities p'_L and p' driven respectively by the traffic in the L and C queues. The marking probability for the L queue (p_L) depends on both the base probability in its own queue (p'_L) and a probability called p_{CL} , which is coupled across from p' in the C queue (see Section 2.4 for the derivation of the specific equations and dependencies).

The probabilities p_{CL} and p_C are derived in lines 4 and 5 of the `dualpi2_update()` function (Figure 6) then used in the `dualpi2_dequeue()` function where p_L is also derived from p_{CL} at line 6 (Figure 4). The code walk-through below builds up to explaining that part of the code eventually, but it starts from packet arrival.

```

1: dualpi2_enqueue(lq, cq, pkt) { % Test limit and classify lq or cq
2:   if ( lq.bytt() + cq.bytt() + MTU > limit)
3:     drop(pkt) % drop packet if buffer is full
4:   timestamp(pkt) % attach arrival time to packet
5:   % Packet classifier
6:   if ( ecn(pkt) modulo 2 == 1 ) % ECN bits = ECT(1) or CE
7:     lq.enqueue(pkt)
8:   else % ECN bits = not-ECT or ECT(0)
9:     cq.enqueue(pkt)
10: }

```

Figure 3: Example Enqueue Pseudocode for DualQ Coupled PI2 AQM

```

1: dualpi2_dequeue(lq, cq, pkt) { % Couples L4S & Classic queues
2:   while ( lq.bytt() + cq.bytt() > 0 ) {
3:     if ( scheduler() == lq ) {
4:       lq.dequeue(pkt) % Scheduler chooses lq
5:       p'_L = laqm(lq.time()) % Native LAQM
6:       p_L = max(p'_L, p_CL) % Combining function
7:       if ( recur(lq, p_L) ) % Linear marking
8:         mark(pkt)
9:     } else {
10:      cq.dequeue(pkt) % Scheduler chooses cq
11:      if ( recur(cq, p_C) ) { % probability p_C = p'^2
12:        if ( ecn(pkt) == 0 ) { % if ECN field = not-ECT
13:          drop(pkt) % squared drop
14:          continue % continue to the top of the while loop
15:        }
16:        mark(pkt) % squared mark
17:      }
18:    }
19:    return(pkt) % return the packet and stop
20:  }
21:  return(NULL) % no packet to dequeue
22: }

23: recur(q, likelihood) { % Returns TRUE with a certain likelihood
24:   q.count += likelihood
25:   if (q.count > 1) {
26:     q.count -= 1
27:     return TRUE
28:   }
29:   return FALSE
30: }

```

Figure 4: Example Dequeue Pseudocode for DualQ Coupled PI2 AQM

When packets arrive, first a common queue limit is checked as shown in line 2 of the enqueueing pseudocode in Figure 3. This assumes a shared buffer for the two queues (Note b discusses the merits of separate buffers). In order to avoid any bias against larger packets, 1 MTU of space is always allowed and the limit is deliberately tested before enqueue.

If limit is not exceeded, the packet is timestamped in line 4. This assumes that queue delay is measured using the sojourn time technique (see Note a for alternatives).

At lines 5-9, the packet is classified and enqueued to the Classic or L4S queue dependent on the least significant bit of the ECN field in the IP header (line 6). Packets with a codepoint having an LSB of 0 (Not-ECT and ECT(0)) will be enqueued in the Classic queue. Otherwise, ECT(1) and CE packets will be enqueued in the L4S queue. Optional additional packet classification flexibility is omitted for brevity (see the L4S ECN protocol [I-D.ietf-tsvwg-ecn-l4s-id]).

The dequeue pseudocode (Figure 4) is repeatedly called whenever the lower layer is ready to forward a packet. It schedules one packet for dequeuing (or zero if the queue is empty) then returns control to the caller, so that it does not block while that packet is being forwarded. While making this dequeue decision, it also makes the necessary AQM decisions on dropping or marking. The alternative of applying the AQMs at enqueue would shift some processing from the critical time when each packet is dequeued. However, it would also add a whole queue of delay to the control signals, making the control loop sloppier (for a typical RTT it would double the Classic queue's feedback delay).

All the dequeue code is contained within a large while loop so that if it decides to drop a packet, it will continue until it selects a packet to schedule. Line 3 of the dequeue pseudocode is where the scheduler chooses between the L4S queue (lq) and the Classic queue (cq). Detailed implementation of the scheduler is not shown (see discussion later).

- * If an L4S packet is scheduled, in lines 7 and 8 the packet is ECN-marked with likelihood p_L . The `recur()` function at the end of Figure 4 is used, which is preferred over random marking because it avoids delay due to randomization when interpreting congestion signals, but it still desynchronizes the saw-teeth of the flows. Line 6 calculates p_L as the maximum of the coupled L4S probability p_{CL} and the probability from the native L4S AQM p'_L . This implements the `max()` function shown in Figure 1 to couple the outputs of the two AQMs together. Of the two probabilities input to p_L in line 6:

- p'_L is calculated per packet in line 5 by the `laqm()` function (see Figure 5),
 - Whereas p_{CL} is maintained by the `dualpi2_update()` function which runs every `Tupdate` (`Tupdate` is set in line 12 of Figure 2).
- * If a Classic packet is scheduled, lines 10 to 17 drop or mark the packet with probability p_C .

The Native L4S AQM algorithm (Figure 5) is a ramp function, similar to the RED algorithm, but simplified as follows:

- * The extent of the ramp is defined in units of queuing delay, not bytes, so that configuration remains invariant as the queue departure rate varies.
- * It uses instantaneous queueing delay, which avoids the complexity of smoothing, but also avoids embedding a worst-case RTT of smoothing delay in the network (see Section 2.1).
- * The ramp rises linearly directly from 0 to 1, not to an intermediate value of p'_L as RED would, because there is no need to keep ECN marking probability low.
- * Marking does not have to be randomized. Determinism is used instead of randomness; to reduce the delay necessary to smooth out the noise of randomness from the signal.

The ramp function requires two configuration parameters, the minimum threshold (`minTh`) and the width of the ramp (`range`), both in units of queuing time, as shown in lines 17 & 18 of the initialization function in Figure 2. The ramp function can be configured as a step (see Note c).

Although the DCTCP paper [Alizadeh-stability] recommends an ECN marking threshold of $0.17 \cdot RTT_{typ}$, it also shows that the threshold can be much shallower with hardly any worse under-utilization of the link (because the amplitude of DCTCP's sawteeth is so small). Based on extensive experiments, for the public Internet the default minimum ECN marking threshold (`target`) in Figure 2 is considered a good compromise, even though it is significantly smaller fraction of RTT_{typ} .

```

1: laqm(qdelay) {                                % Returns native L4S AQM probability
2:   if (qdelay >= maxTh)
3:     return 1
4:   else if (qdelay > minTh)
5:     return (qdelay - minTh)/range % Divide could use a bit-shift
6:   else
7:     return 0
8: }

```

Figure 5: Example Pseudocode for the Native L4S AQM

```

1: dualpi2_update(lq, cq) {                      % Update p' every Tupdate
2:   curq = cq.time() % use queuing time of first-in Classic packet
3:   p' = p' + alpha * (curq - target) + beta * (curq - prevq)
4:   p_CL = k * p' % Coupled L4S prob = base prob * coupling factor
5:   p_C = p'^2 % Classic prob = (base prob)^2
6:   prevq = curq
7: }

```

Figure 6: Example PI-Update Pseudocode for DualQ Coupled PI2 AQM

(Clamping p' within the range $[0,1]$ omitted for clarity - see text)

The coupled marking probability, p_{CL} depends on the base probability (p'), which is kept up to date by the core PI algorithm in Figure 6 executed every Tupdate.

Note that p' solely depends on the queuing time in the Classic queue. In line 2, the current queuing delay ($curq$) is evaluated from how long the head packet was in the Classic queue (cq). The function $cq.time()$ (not shown) subtracts the time stamped at enqueue from the current time (see Note a) and implicitly takes the current queuing delay as 0 if the queue is empty.

The algorithm centres on line 3, which is a classical Proportional-Integral (PI) controller that alters p' dependent on: a) the error between the current queuing delay ($curq$) and the target queuing delay, 'target'; and b) the change in queuing delay since the last sample. The name 'PI' represents the fact that the second factor (how fast the queue is growing) is $_P_roportional$ to load while the first is the $_I_ntegral$ of the load (so it removes any standing queue in excess of the target).

The target parameter can be set based on local knowledge, but the aim is for the default to be a good compromise for anywhere in the intended deployment environment -- the public Internet. According to [PI2param], the target queuing delay on line 9 of Figure 2 is related

to the typical base RTT worldwide, RTT_{typ} , by two factors: $target = RTT_{typ} * g * f$. Below we summarize the rationale behind these factors and introduce a further adjustment. The two factors ensure that, in a large proportion of cases (say 90%), the sawtooth variations in RTT of a single flow will fit within the buffer without underutilizing the link. Frankly, these factors are educated guesses, but with the emphasis closer to 'educated' than to 'guess' (see [PI2param] for full background):

- * RTT_{typ} is taken as 25 ms. This is based on an average CDN latency measured in each country weighted by the number of Internet users in that country to produce an overall weighted average for the Internet [PI2param]. Countries were ranked by number of Internet users, and once 90% of Internet users were covered, smaller countries were excluded to avoid unrepresentatively small sample sizes. Also, importantly, the data for the average CDN latency in China (with the largest number of Internet users) has been removed, because the CDN latency was a significant outlier and, on reflection, the experimental technique seemed inappropriate to the CDN market in China.
- * g is taken as 0.38. The factor g is a geometry factor that characterizes the shape of the sawteeth of prevalent Classic congestion controllers. The geometry factor is the fraction of the amplitude of the sawtooth variability in queue delay that lies below the AQM's target. For instance, at low bit rate, the geometry factor of standard Reno is 0.5, but at higher rates it tends to just under 1. According to the census of congestion controllers conducted by Mishra *et al* in Jul-Oct 2019 [CCcensus19], most Classic TCP traffic uses Cubic. And, according to the analysis in [PI2param], if running over a PI2 AQM, a large proportion of this Cubic traffic would be in its Reno-Friendly mode, which has a geometry factor of ~ 0.39 (all known implementations). The rest of the Cubic traffic would be in true Cubic mode, which has a geometry factor of ~ 0.36 . Without modelling the sawtooth profiles from all the other less prevalent congestion controllers, we estimate a 7:3 weighted average of these two, resulting in an average geometry factor of 0.38.
- * f is taken as 2. The factor f is a safety factor that increases the target queue to allow for the distribution of RTT_{typ} around its mean. Otherwise the target queue would only avoid underutilization for those users below the mean. It also provides a safety margin for the proportion of paths in use that span beyond the distance between a user and their local CDN. Currently no data is available on the variance of queue delay around the mean in each region, so there is plenty of room for this guess to become more educated.

- * [PI2param] recommends $\text{target} = \text{RTT_typ} * g * f = 25\text{ms} * 0.38 * 2 = 19\text{ ms}$. However a further adjustment is warranted, because target is moving year on year. The paper is based on data collected in 2019, and it mentions evidence from speedtest.net that suggests RTT_typ reduced by 17% (fixed) or 12% (mobile) between 2020 and 2021. Therefore we recommend a default of $\text{target} = 15\text{ ms}$ at the time of writing (2021).

Operators can always use the data and discussion in [PI2param] to configure a more appropriate target for their environment. For instance, an operator might wish to question the assumptions called out in that paper, such as the goal of no underutilization for a large majority of single flow transfers (given many large transfers use multiple flows to avoid the scaling limitations of Classic flows).

The two 'gain factors' in line 3 of Figure 6, alpha and beta, respectively weight how strongly each of the two elements (Integral and Proportional) alters p' . They are in units of 'per second of delay' or Hz, because they transform differences in queueing delay into changes in probability (assuming probability has a value from 0 to 1).

Alpha and beta determine how much p' ought to change after each update interval (Tupdate). For smaller Tupdate, p' should change by the same amount per second, but in finer more frequent steps. So alpha depends on Tupdate (see line 13 of the initialization function in Figure 2). It is best to update p' as frequently as possible, but Tupdate will probably be constrained by hardware performance. As shown in line 13, the update interval should be frequent enough to update at least once in the time taken for the target queue to drain ('target') as long as it updates at least three times per maximum RTT. Tupdate defaults to 16 ms in the reference Linux implementation because it has to be rounded to a multiple of 4 ms. For link rates from 4 to 200 Mb/s and a maximum RTT of 100ms, it has been verified through extensive testing that Tupdate=16ms (as also recommended in the PIE spec [RFC8033]) is sufficient.

The choice of alpha and beta also determines the AQM's stable operating range. The AQM ought to change p' as fast as possible in response to changes in load without over-compensating and therefore causing oscillations in the queue. Therefore, the values of alpha and beta also depend on the RTT of the expected worst-case flow (RTT_max).

The maximum RTT of a PI controller (RTT_max in line 10 of Figure 2) is not an absolute maximum, but more instability (more queue variability) sets in for long-running flows with an RTT above this

value. The propagation delay half way round the planet and back in glass fibre is 200 ms. However, hardly any traffic traverses such extreme paths and, since the significant consolidation of Internet traffic between 2007 and 2009 [Labovitz10], a high and growing proportion of all Internet traffic (roughly two-thirds at the time of writing) has been served from content distribution networks (CDNs) or 'cloud' services distributed close to end-users. The Internet might change again, but for now, designing for a maximum RTT of 100ms is a good compromise between faster queue control at low RTT and some instability on the occasions when a longer path is necessary.

Recommended derivations of the gain constants alpha and beta can be approximated for Reno over a PI2 AQM as: $\alpha = 0.1 * \text{Tupdate} / \text{RTT_max}^2$; $\beta = 0.3 / \text{RTT_max}$, as shown in lines 14 & 15 of Figure 2. These are derived from the stability analysis in [PI2]. For the default values of Tupdate=16 ms and RTT_max = 100 ms, they result in $\alpha = 0.16$; $\beta = 3.2$ (discrepancies are due to rounding). These defaults have been verified with a wide range of link rates, target delays and a range of traffic models with mixed and similar RTTs, short and long flows, etc.

In corner cases, p' can overflow the range [0,1] so the resulting value of p' has to be bounded (omitted from the pseudocode). Then, as already explained, the coupled and Classic probabilities are derived from the new p' in lines 4 and 5 of Figure 6 as $p_{\text{CL}} = k * p'$ and $p_{\text{C}} = p'^2$.

Because the coupled L4S marking probability (p_{CL}) is factored up by k , the dynamic gain parameters alpha and beta are also inherently factored up by k for the L4S queue. So, the effective gain factor for the L4S queue is $k * \alpha$ (with defaults $\alpha = 0.16 \text{ Hz}$ and $k=2$, effective L4S $\alpha = 0.32 \text{ Hz}$).

Unlike in PIE [RFC8033], alpha and beta do not need to be tuned every Tupdate dependent on p' . Instead, in PI2, alpha and beta are independent of p' because the squaring applied to Classic traffic tunes them inherently. This is explained in [PI2], which also explains why this more principled approach removes the need for most of the heuristics that had to be added to PIE.

Nonetheless, an implementer might wish to add selected details to either AQM. For instance the Linux reference DualPI2 implementation includes the following (not shown in the pseudocode above):

- * Classic and coupled marking or dropping (i.e. based on p_C and p_CL from the PI controller) is not applied to a packet if the aggregate queue length in bytes is < 2 MTU (prior to enqueueing the packet or dequeuing it, depending on whether the AQM is configured to be applied at enqueue or dequeue);
- * In the WRR scheduler, the 'credit' indicating which queue should transmit is only changed if there are packets in both queues (i.e. if there is actual resource contention). This means that a properly paced L flow might never be delayed by the WRR. The WRR credit is reset in favour of the L queue when the link is idle.

An implementer might also wish to add other heuristics, e.g. burst protection [RFC8033] or enhanced burst protection [RFC8034].

Notes:

- a. The drain rate of the queue can vary if it is scheduled relative to other queues, or to cater for fluctuations in a wireless medium. To auto-adjust to changes in drain rate, the queue needs to be measured in time, not bytes or packets [AQMetrics], [CoDel]. Queuing delay could be measured directly by storing a per-packet time-stamp as each packet is enqueued, and subtracting this from the system time when the packet is dequeued. If time-stamping is not easy to introduce with certain hardware, queuing delay could be predicted indirectly by dividing the size of the queue by the predicted departure rate, which might be known precisely for some link technologies (see for example in DOCSIS PIE [RFC8034]).
- b. Line 2 of the `dualpi2_enqueue()` function (Figure 3) assumes an implementation where lq and cq share common buffer memory. An alternative implementation could use separate buffers for each queue, in which case the arriving packet would have to be classified first to determine which buffer to check for available space. The choice is a trade off; a shared buffer can use less memory whereas separate buffers isolate the L4S queue from tail-drop due to large bursts of Classic traffic (e.g. a Classic Reno TCP during slow-start over a long RTT).
- c. There has been some concern that using the step function of DCTCP for the Native L4S AQM requires end-systems to smooth the signal for an unnecessarily large number of round trips to ensure sufficient fidelity. A ramp is no worse than a step in initial experiments with existing DCTCP. Therefore, it is recommended that a ramp is configured in place of a step, which will allow congestion control algorithms to investigate faster smoothing algorithms.

A ramp is more general than a step, because an operator can effectively turn the ramp into a step function, as used by DCTCP, by setting the range to zero. There will not be a divide by zero problem at line 5 of Figure 5 because, if minTh is equal to maxTh, the condition for this ramp calculation cannot arise.

A.2. Pass #2: Edge-Case Details

This section takes a second pass through the pseudocode adding details of two edge-cases: low link rate and overload. Figure 7 repeats the dequeue function of Figure 4, but with details of both edge-cases added. Similarly Figure 8 repeats the core PI algorithm of Figure 6, but with overload details added. The initialization, enqueue, L4S AQM and recur functions are unchanged.

The link rate can be so low that it takes a single packet queue longer to serialize than the threshold delay at which ECN marking starts to be applied in the L queue. Therefore, a minimum marking threshold parameter in units of packets rather than time is necessary (Th_len, default 1 packet in line 19 of Figure 2) to ensure that the ramp does not trigger excessive marking on slow links. Where an implementation knows the link rate, it can set up this minimum at the time it is configured. For instance, it would divide 1 MTU by the link rate to convert it into a serialization time, then if the lower threshold of the Native L AQM ramp was lower than this serialization time, it could increase the thresholds to shift the bottom of the ramp to 2 MTU. This is the approach used in DOCSIS [DOCSIS3.1], because the configured link rate is dedicated to the DualQ.

The pseudocode given here applies where the link rate is unknown, which is more common for software implementations that might be deployed in scenarios where the link is shared with other queues. In lines 5a to 5d in Figure 7 the native L4S marking probability, p'_L , is zeroed if the queue is only 1 packet (in the default configuration).

Linux implementation note:

- * In Linux, the check that the queue exceeds Th_len before marking with the native L4S AQM is actually at enqueue, not dequeue, otherwise it would exempt the last packet of a burst from being marked. The result of the check is conveyed from enqueue to the dequeue function via a boolean in the packet metadata.

Persistent overload is deemed to have occurred when Classic drop/marking probability reaches p_{Cmax} . Above this point, the Classic drop probability is applied to both L and C queues, irrespective of whether any packet is ECN-capable. ECT packets that are not dropped can still be ECN-marked.

In line 10 of the initialization function (Figure 2), the maximum Classic drop probability $p_{Cmax} = \min(1/k^2, 1)$ or $1/4$ for the default coupling factor $k=2$. In practice, 25% has been found to be a good threshold to preserve fairness between ECN capable and non ECN capable traffic. This protects the queues against both temporary overload from responsive flows and more persistent overload from any unresponsive traffic that falsely claims to be responsive to ECN.

When the Classic ECN marking probability reaches the p_{Cmax} threshold ($1/k^2$), the marking probability coupled to the L4S queue, p_{CL} will always be 100% for any k (by equation (1) in Section 2). So, for readability, the constant p_{Lmax} is defined as 1 in line 22 of the initialization function (Figure 2). This is intended to ensure that the L4S queue starts to introduce dropping once ECN-marking saturates at 100% and can rise no further. The 'Prague L4S' requirements [I-D.ietf-tsvwg-ecn-l4s-id] state that, when an L4S congestion control detects a drop, it falls back to a response that coexists with 'Classic' Reno congestion control. So it is correct that, when the L4S queue drops packets, it drops them proportional to p'^2 , as if they are Classic packets.

The two queues each test for overload in lines 4b and 12b of the dequeue function (Figure 7). Lines 8c to 8g drop L4S packets with probability p'^2 . Lines 8h to 8i mark the remaining packets with probability p_{CL} . Given $p_{Lmax} = 1$, all remaining packets will be marked because, to have reached the else block at line 8b, $p_{CL} \geq 1$.

Line 2a in the core PI algorithm (Figure 8) deals with overload of the L4S queue when there is little or no Classic traffic. This is necessary, because the core PI algorithm maintains the appropriate drop probability to regulate overload, but it depends on the length of the Classic queue. If there is little or no Classic queue the naive PI update function in Figure 6 would drop nothing, even if the L4S queue were overloaded - so tail drop would have to take over (lines 2 and 3 of Figure 3).

Instead, line 2a of the full PI update function in Figure 8 ensures that the base PI AQM in line 3 is driven by whichever of the two queue delays is greater, but line 3 still always uses the same Classic target (default 15 ms). If L queue delay is greater just because there is little or no Classic traffic, normally it will still be well below the base AQM target. This is because L4S traffic is

also governed by the shallow threshold of its own native AQM (lines 5 and 6 of the dequeue algorithm in Figure 7). So the base AQM will be driven to zero and not contribute. However, if the L queue is overloaded by traffic that is unresponsive to its marking, the max() in line 2 enables the L queue to smoothly take over driving the base AQM into overload mode even if there is little or no Classic traffic. Then the base AQM will keep the L queue to the Classic target (default 15 ms) by shedding L packets.

```

1:  dualpi2_dequeue(lq, cq, pkt) {      % Couples L4S & Classic queues
2:    while ( lq.by() + cq.by() > 0 ) {
3:      if ( scheduler() == lq ) {
4a:        lq.dequeue(pkt)                % L4S scheduled
4b:        if ( p_CL < p_Lmax ) {          % Check for overload saturation
5a:          if (lq.len()>Th_len)           % >1 packet queued
5b:            p'_L = laqm(lq.time())       % Native LAQM
5c:          else
5d:            p'_L = 0                     % Suppress marking 1 pkt queue
6:            p_L = max(p'_L, p_CL)         % Combining function
7:            if ( recur(lq, p_L)           % Linear marking
8a:              mark(pkt)
8b:            } else {                     % overload saturation
8c:              if ( recur(lq, p_C) ) {     % probability p_C = p'^2
8e:                drop(pkt)                % revert to Classic drop due to overload
8f:                continue                 % continue to the top of the while loop
8g:              }
8h:              if ( recur(lq, p_CL) )     % probability p_CL = k * p'
8i:                mark(pkt)                % linear marking of remaining packets
8j:            }
9:          } else {
10:           cq.dequeue(pkt)                % Classic scheduled
11:           if ( recur(cq, p_C) ) {         % probability p_C = p'^2
12a:            if ( (ecn(pkt) == 0)         % ECN field = not-ECT
12b:              OR (p_C >= p_Cmax) ) {     % Overload disables ECN
13:              drop(pkt)                  % squared drop, redo loop
14:              continue                   % continue to the top of the while loop
15:            }
16:            mark(pkt)                     % squared mark
17:          }
18:        }
19:      return(pkt)                         % return the packet and stop
20:    }
21:    return(NULL)                          % no packet to dequeue
22:  }

```

Figure 7: Example Dequeue Pseudocode for DualQ Coupled PI2 AQM
(Including Code for Edge-Cases)

```

1: dualpi2_update(lq, cq) {                                % Update p' every Tupdate
2a:   curq = max(cq.time(), lq.time())                    % use greatest queuing time
3:   p' = p' + alpha * (curq - target) + beta * (curq - prevq)
4:   p_CL = p' * k % Coupled L4S prob = base prob * coupling factor
5:   p_C = p'^2                                           % Classic prob = (base prob)^2
6:   prevq = curq
7: }
```

Figure 8: Example PI-Update Pseudocode for DualQ Coupled PI2 AQM
(Including Overload Code)

The choice of scheduler technology is critical to overload protection (see Section 4.2.2).

- * A well-understood weighted scheduler such as weighted round robin (WRR) is recommended. As long as the scheduler weight for Classic is small (e.g. 1/16), its exact value is unimportant because it does not normally determine capacity shares. The weight is only important to prevent unresponsive L4S traffic starving Classic traffic in the short term (see Section 4.2.2). This is because capacity sharing between the queues is normally determined by the coupled congestion signal, which overrides the scheduler, by making L4S sources leave roughly equal per-flow capacity available for Classic flows.
- * Alternatively, a time-shifted FIFO (TS-FIFO) could be used. It works by selecting the head packet that has waited the longest, biased against the Classic traffic by a time-shift of *tshift*. To implement time-shifted FIFO, the scheduler() function in line 3 of the dequeue code would simply be implemented as the scheduler() function at the bottom of Figure 10 in Appendix B. For the public Internet a good value for *tshift* is 50ms. For private networks with smaller diameter, about 4*target would be reasonable. TS-FIFO is a very simple scheduler, but complexity might need to be added to address some deficiencies (which is why it is not recommended over WRR):
 - TS-FIFO does not fully isolate latency in the L4S queue from uncontrolled bursts in the Classic queue;
 - TS-FIFO is only appropriate if time-stamping of packets is feasible;
 - Even if time-stamping is supported, the sojourn time of the head packet is always stale. For instance, if a burst arrives at an empty queue, the sojourn time only fully measures the burst's delay when its last packet is dequeued, even though the

queue knew about the burst from the start - so it could have signalled congestion earlier. To remedy this, each head packet can be marked when it is dequeued based on the expected delay of the tail packet behind it, as explained below, rather than based on the head packet's own delay due to the packets in front of it. [Heist21] identifies a specific scenario where bursty traffic significantly hits utilization of the L queue. If this effect proves to be more widely applicable, it is believed that using the delay behind the head would improve performance.

The delay behind the head can be implemented by dividing the backlog at dequeue by the link rate or equivalently multiplying the backlog by the delay per unit of backlog. The implementation details will depend on whether the link rate is known; if it is not, a moving average of the delay per unit backlog can be maintained. This delay consists of serialization as well as media acquisition for shared media. So the details will depend strongly on the specific link technology. This approach should be less sensitive to timing errors and cost less in operations and memory than the otherwise equivalent 'scaled sojourn time' metric, which is the sojourn time of a packet scaled by the ratio of the queue sizes when the packet departed and arrived [SigQ-Dyn].

- * A strict priority scheduler would be inappropriate as discussed in Section 4.2.2.

Appendix B. Example DualQ Coupled Curvy RED Algorithm

As another example of a DualQ Coupled AQM algorithm, the pseudocode below gives the Curvy RED based algorithm. Although the AQM was designed to be efficient in integer arithmetic, to aid understanding it is first given using floating point arithmetic (Figure 10). Then, one possible optimization for integer arithmetic is given, also in pseudocode (Figure 11). To aid comparison, the line numbers are kept in step between the two by using letter suffixes where the longer code needs extra lines.

B.1. Curvy RED in Pseudocode

The pseudocode manipulates three main structures of variables: the packet (pkt), the L4S queue (lq) and the Classic queue (cq) and consists of the following five functions:

- * The initialization function `cred_params_init(...)` (Figure 2) that sets parameter defaults (the API for setting non-default values is omitted for brevity);

- * The dequeue function `cred_dequeue(lq, cq, pkt)` (Figure 4);
- * The scheduling function `scheduler()`, which selects between the head packets of the two queues.

It also uses the following functions that are either shown elsewhere, or not shown in full here:

- * The enqueue function, which is identical to that used for DualPI2, `dualpi2_enqueue(lq, cq, pkt)` in Figure 3;
- * `mark(pkt)` and `drop(pkt)` for ECN-marking and dropping a packet;
- * `cq.bytest()` or `lq.bytest()` returns the current length (aka. backlog) of the relevant queue in bytes;
- * `cq.time()` or `lq.time()` returns the current queuing delay (aka. sojourn time or service time) of the relevant queue in units of time (see Note a in Appendix A.1).

Because Curvy RED was evaluated before DualPI2, certain improvements introduced for DualPI2 were not evaluated for Curvy RED. In the pseudocode below, the straightforward improvements have been added on the assumption they will provide similar benefits, but that has not been proven experimentally. They are: i) a conditional priority scheduler instead of strict priority ii) a time-based threshold for the native L4S AQM; iii) ECN support for the Classic AQM. A recent evaluation has proved that a minimum ECN-marking threshold (`minTh`) greatly improves performance, so this is also included in the pseudocode.

Overload protection has not been added to the Curvy RED pseudocode below so as not to detract from the main features. It would be added in exactly the same way as in Appendix A.2 for the DualPI2 pseudocode. The native L4S AQM uses a step threshold, but a ramp like that described for DualPI2 could be used instead. The scheduler uses the simple TS-FIFO algorithm, but it could be replaced with WRR.

The Curvy RED algorithm has not been maintained or evaluated to the same degree as the DualPI2 algorithm. In initial experiments on broadband access links ranging from 4 Mb/s to 200 Mb/s with base RTTs from 5 ms to 100 ms, Curvy RED achieved good results with the default parameters in Figure 9.

The parameters are categorised by whether they relate to the Classic AQM, the L4S AQM or the framework coupling them together. Constants and variables derived from these parameters are also included at the end of each category. These are the raw input parameters for the

algorithm. A configuration front-end could accept more meaningful parameters (e.g. `RTT_max` and `RTT_typ`) and convert them into these raw parameters, as has been done for DualPI2 in Appendix A. Where necessary, parameters are explained further in the walk-through of the pseudocode below.

```

1: cred_params_init(...) {           % Set input parameter defaults
2:   % DualQ Coupled framework parameters
3:   limit = MAX_LINK_RATE * 250 ms   % Dual buffer size
4:   k' = 1                           % Coupling factor as a power of 2
5:   tshift = 50 ms                   % Time shift of TS-FIFO scheduler
6:   % Constants derived from Classic AQM parameters
7:   k = 2^k'                         % Coupling factor from Equation (1)
6:
7:   % Classic AQM parameters
8:   g_C = 5                          % EWMA smoothing parameter as a power of 1/2
9:   S_C = -1                         % Classic ramp scaling factor as a power of 2
10:  minTh = 500 ms                   % No Classic drop/mark below this queue delay
11:  % Constants derived from Classic AQM parameters
12:  gamma = 2^(-g_C)                 % EWMA smoothing parameter
13:  range_C = 2^S_C                  % Range of Classic ramp
14:
15:  % L4S AQM parameters
16:  T = 1 ms                         % Queue delay threshold for native L4S AQM
17:  % Constants derived from above parameters
18:  S_L = S_C - k'                   % L4S ramp scaling factor as a power of 2
19:  range_L = 2^S_L                  % Range of L4S ramp
20: }
```

Figure 9: Example Header Pseudocode for DualQ Coupled Curvy RED AQM

```

1: cred_dequeue(lq, cq, pkt) {           % Couples L4S & Classic queues
2:   while ( lq.bytt() + cq.bytt() > 0 ) {
3:     if ( scheduler() == lq ) {
4:       lq.dequeue(pkt)                  % L4S scheduled
5a:      p_CL = (Q_C - minTh) / range_L
5b:      if ( ( lq.time() > T )
5c:          OR ( p_CL > maxrand(U) ) )
6:        mark(pkt)
7:      } else {
8:        cq.dequeue(pkt)                  % Classic scheduled
9a:        Q_C = gamma * cq.time() + (1-gamma) * Q_C % Classic Q EWMA
10a:       sqrt_p_C = (Q_C - minTh) / range_C
10b:       if ( sqrt_p_C > maxrand(2*U) ) {
11:         if ( (ecn(pkt) == 0) ) {        % ECN field = not-ECT
12:           drop(pkt)                    % Squared drop, redo loop
13:           continue                    % continue to the top of the while loop
14:         }
15:         mark(pkt)
16:       }
17:     }
18:     return(pkt)                        % return the packet and stop here
19:   }
20:   return(NULL)                         % no packet to dequeue
21: }

22: maxrand(u) {                          % return the max of u random numbers
23:   maxr=0
24:   while (u-- > 0)
25:     maxr = max(maxr, rand())            % 0 <= rand() < 1
26:   return(maxr)
27: }

28: scheduler() {
29:   if ( lq.time() + tshift >= cq.time() )
30:     return lq;
31:   else
32:     return cq;
33: }

```

Figure 10: Example Dequeue Pseudocode for DualQ Coupled Curvy RED AQM

The dequeue pseudocode (Figure 10) is repeatedly called whenever the lower layer is ready to forward a packet. It schedules one packet for dequeuing (or zero if the queue is empty) then returns control to the caller, so that it does not block while that packet is being forwarded. While making this dequeue decision, it also makes the necessary AQM decisions on dropping or marking. The alternative of applying the AQMs at enqueue would shift some processing from the

critical time when each packet is dequeued. However, it would also add a whole queue of delay to the control signals, making the control loop very sloppy.

The code is written assuming the AQMs are applied on dequeue (Note 1). All the dequeue code is contained within a large while loop so that if it decides to drop a packet, it will continue until it selects a packet to schedule. If both queues are empty, the routine returns NULL at line 20. Line 3 of the dequeue pseudocode is where the conditional priority scheduler chooses between the L4S queue (lq) and the Classic queue (cq). The time-shifted FIFO scheduler is shown at lines 28-33, which would be suitable if simplicity is paramount (see Note 2).

Within each queue, the decision whether to forward, drop or mark is taken as follows (to simplify the explanation, it is assumed that $U=1$):

L4S: If the test at line 3 determines there is an L4S packet to dequeue, the tests at lines 5b and 5c determine whether to mark it. The first is a simple test of whether the L4S queue delay (`lq.time()`) is greater than a step threshold T (Note 3). The second test is similar to the random ECN marking in RED, but with the following differences: i) marking depends on queuing time, not bytes, in order to scale for any link rate without being reconfigured; ii) marking of the L4S queue depends on a logical OR of two tests; one against its own queuing time and one against the queuing time of the `_other_` (Classic) queue; iii) the tests are against the instantaneous queuing time of the L4S queue, but a smoothed average of the other (Classic) queue; iv) the queue is compared with the maximum of U random numbers (but if $U=1$, this is the same as the single random number used in RED).

Specifically, in line 5a the coupled marking probability p_{CL} is set to the amount by which the averaged Classic queueing delay Q_C exceeds the minimum queuing delay threshold (`minTh`) all divided by the L4S scaling parameter `range_L`. `range_L` represents the queuing delay (in seconds) added to `minTh` at which marking probability would hit 100%. Then in line 5c (if $U=1$) the result is compared with a uniformly distributed random number between 0 and 1, which ensures that, over `range_L`, marking probability will linearly increase with queueing time.

Classic: If the scheduler at line 3 chooses to dequeue a Classic packet and jumps to line 7, the test at line 10b determines whether to drop or mark it. But before that, line 9a updates Q_C , which is an exponentially weighted moving average (Note 4) of the queuing time of the Classic queue, where `cq.time()` is the current

instantaneous queueing time of the packet at the head of the Classic queue (zero if empty) and gamma is the EWMA constant (default 1/32, see line 12 of the initialization function).

Lines 10a and 10b implement the Classic AQM. In line 10a the averaged queueing time Q_C is divided by the Classic scaling parameter $range_C$, in the same way that queueing time was scaled for L4S marking. This scaled queueing time will be squared to compute Classic drop probability so, before it is squared, it is effectively the square root of the drop probability, hence it is given the variable name $sqrt_p_C$. The squaring is done by comparing it with the maximum out of two random numbers (assuming $U=1$). Comparing it with the maximum out of two is the same as the logical 'AND' of two tests, which ensures drop probability rises with the square of queueing time.

The AQM functions in each queue (lines 5c & 10b) are two cases of a new generalization of RED called Curvy RED, motivated as follows. When the performance of this AQM was compared with FQ-CoDel and PIE, their goal of holding queueing delay to a fixed target seemed misguided [CRED_Insights]. As the number of flows increases, if the AQM does not allow host congestion controllers to increase queueing delay, it has to introduce abnormally high levels of loss. Then loss rather than queueing becomes the dominant cause of delay for short flows, due to timeouts and tail losses.

Curvy RED constrains delay with a softened target that allows some increase in delay as load increases. This is achieved by increasing drop probability on a convex curve relative to queue growth (the square curve in the Classic queue, if $U=1$). Like RED, the curve hugs the zero axis while the queue is shallow. Then, as load increases, it introduces a growing barrier to higher delay. But, unlike RED, it requires only two parameters, not three. The disadvantage of Curvy RED (compared to a PI controller for example) is that it is not adapted to a wide range of RTTs. Curvy RED can be used as is when the RTT range to be supported is limited, otherwise an adaptation mechanism is needed.

From our limited experiments with Curvy RED so far, recommended values of these parameters are: $S_C = -1$; $g_C = 5$; $T = 5 * MTU$ at the link rate (about 1ms at 60Mb/s) for the range of base RTTs typical on the public Internet. [CRED_Insights] explains why these parameters are applicable whatever rate link this AQM implementation is deployed on and how the parameters would need to be adjusted for a scenario with a different range of RTTs (e.g. a data centre). The setting of k depends on policy (see Section 2.5 and Appendix C.2 respectively for its recommended setting and guidance on alternatives).

There is also a `cUrviness` parameter, `U`, which is a small positive integer. It is likely to take the same hard-coded value for all implementations, once experiments have determined a good value. Only `U=1` has been used in experiments so far, but results might be even better with `U=2` or higher.

Notes:

1. The alternative of applying the AQMs at enqueue would shift some processing from the critical time when each packet is dequeued. However, it would also add a whole queue of delay to the control signals, making the control loop sloppier (for a typical RTT it would double the Classic queue's feedback delay). On a platform where packet timestamping is feasible, e.g. Linux, it is also easiest to apply the AQMs at dequeue because that is where queuing time is also measured.
2. WRR better isolates the L4S queue from large delay bursts in the Classic queue, but it is slightly less simple than TS-FIFO. If WRR were used, a low default Classic weight (e.g. 1/16) would need to be configured in place of the time shift in line 5 of the initialization function (Figure 9).
3. A step function is shown for simplicity. A ramp function (see Figure 5 and the discussion around it in Appendix A.1) is recommended, because it is more general than a step and has the potential to enable L4S congestion controls to converge more rapidly.
4. An EWMA is only one possible way to filter bursts; other more adaptive smoothing methods could be valid and it might be appropriate to decrease the EWMA faster than it increases, e.g. by using the minimum of the smoothed and instantaneous queue delays, `min(Q_C, qc.time())`.

B.2. Efficient Implementation of Curvy RED

Although code optimization depends on the platform, the following notes explain where the design of Curvy RED was particularly motivated by efficient implementation.

The Classic AQM at line 10b calls `maxrand(2*U)`, which gives twice as much curviness as the call to `maxrand(U)` in the marking function at line 5c. This is the trick that implements the square rule in equation (1) (Section 2.1). This is based on the fact that, given a number X from 1 to 6, the probability that two dice throws will both be less than X is the square of the probability that one throw will be less than X . So, when $U=1$, the L4S marking function is linear and the Classic dropping function is squared. If $U=2$, L4S would be a square function and Classic would be quartic. And so on.

The `maxrand(u)` function in lines 16-21 simply generates u random numbers and returns the maximum. Typically, `maxrand(u)` could be run in parallel out of band. For instance, if $U=1$, the Classic queue would require the maximum of two random numbers. So, instead of calling `maxrand(2*U)` in-band, the maximum of every pair of values from a pseudorandom number generator could be generated out-of-band, and held in a buffer ready for the Classic queue to consume.

```

1: cred_dequeue(lq, cq, pkt) {           % Couples L4S & Classic queues
2:   while ( lq.bytt() + cq.bytt() > 0 ) {
3:     if ( scheduler() == lq ) {
4:       lq.dequeue(pkt)                  % L4S scheduled
5:       if ((lq.time() > T) OR (Q_C >> (S_L-2) > maxrand(U)))
6:         mark(pkt)
7:     } else {
8:       cq.dequeue(pkt)                  % Classic scheduled
9:       Q_C += (qc.ns() - Q_C) >> g_C    % Classic Q EWMA
10:      if ( (Q_C >> (S_C-2) ) > maxrand(2*U) ) {
11:        if ( (ecn(pkt) == 0) ) {        % ECN field = not-ECT
12:          drop(pkt)                    % Squared drop, redo loop
13:          continue                    % continue to the top of the while loop
14:        }
15:        mark(pkt)
16:      }
17:    }
18:    return(pkt)                        % return the packet and stop here
19:  }
20:  return(NULL)                        % no packet to dequeue
21: }
```

Figure 11: Optimised Example Dequeue Pseudocode for Coupled DualQ AQM using Integer Arithmetic

The two ranges, `range_L` and `range_C` are expressed as powers of 2 so that division can be implemented as a right bit-shift (`>>`) in lines 5 and 10 of the integer variant of the pseudocode (Figure 11).

For the integer variant of the pseudocode, an integer version of the `rand()` function used at line 25 of the `maxrand(function)` in Figure 10 would be arranged to return an integer in the range $0 \leq \text{maxrand}() < 2^{32}$ (not shown). This would scale up all the floating point probabilities in the range $[0,1]$ by 2^{32} .

Queuing delays are also scaled up by 2^{32} , but in two stages: i) In line 9 queuing time `qc.ns()` is returned in integer nanoseconds, making the value about 2^{30} times larger than when the units were seconds, ii) then in lines 5 and 10 an adjustment of -2 to the right bit-shift multiplies the result by 2^2 , to complete the scaling by 2^{32} .

In line 8 of the initialization function, the EWMA constant `gamma` is represented as an integer power of 2, `g_C`, so that in line 9 of the integer code the division needed to weight the moving average can be implemented by a right bit-shift ($>> g_C$).

Appendix C. Choice of Coupling Factor, k

C.1. RTT-Dependence

Where Classic flows compete for the same capacity, their relative flow rates depend not only on the congestion probability, but also on their end-to-end RTT (= base RTT + queue delay). The rates of Reno [RFC5681] flows competing over an AQM are roughly inversely proportional to their RTTs. Cubic exhibits similar RTT-dependence when in Reno-compatibility mode, but it is less RTT-dependent otherwise.

Until the early experiments with the DualQ Coupled AQM, the importance of the reasonably large Classic queue in mitigating RTT-dependence when the base RTT is low had not been appreciated. Appendix A.1.6 of the L4S ECN protocol [I-D.ietf-tsvwg-ecn-l4s-id] uses numerical examples to explain why bloated buffers had concealed the RTT-dependence of Classic congestion controls before that time. Then it explains why, the more that queuing delays have reduced, the more that RTT-dependence has surfaced as a potential starvation problem for long RTT flows, when competing against very short RTT flows.

Given that congestion control on end-systems is voluntary, there is no reason why it has to be voluntarily RTT-dependent. The RTT-dependence of existing Classic traffic cannot be 'undeployed'. Therefore, [I-D.ietf-tsvwg-ecn-l4s-id] requires L4S congestion controls to be significantly less RTT-dependent than the standard Reno congestion control [RFC5681], at least at low RTT. Then RTT-

dependence ought to be no worse than it is with appropriately sized Classic buffers. Following this approach means there is no need for network devices to address RTT-dependence, although there would be no harm if they did, which per-flow queuing inherently does.

C.2. Guidance on Controlling Throughput Equivalence

The coupling factor, k , determines the balance between L4S and Classic flow rates (see Section 2.5.2.1 and equation (1)).

For the public Internet, a coupling factor of $k=2$ is recommended, and justified below. For scenarios other than the public Internet, a good coupling factor can be derived by plugging the appropriate numbers into the same working.

To summarize the maths below, from equation (7) it can be seen that choosing $k=1.64$ would theoretically make L4S throughput roughly the same as Classic, if their actual end-to-end RTTs were the same. However, even if the base RTTs are the same, the actual RTTs are unlikely to be the same, because Classic traffic needs a fairly large queue to avoid under-utilization and excess drop. Whereas L4S does not.

Therefore, to determine the appropriate coupling factor policy, the operator needs to decide at what base RTT it wants L4S and Classic flows to have roughly equal throughput, once the effect of the additional Classic queue on Classic throughput has been taken into account. With this approach, a network operator can determine a good coupling factor without knowing the precise L4S algorithm for reducing RTT-dependence - or even in the absence of any algorithm.

The following additional terminology will be used, with appropriate subscripts:

r : Packet rate [pkt/s]

R : RTT [s/round]

p : ECN marking probability []

On the Classic side, we consider Reno as the most sensitive and therefore worst-case Classic congestion control. We will also consider Cubic in its Reno-friendly mode ('CReno'), as the most prevalent congestion control, according to the references and analysis in [PI2param]. In either case, the Classic packet rate in steady state is given by the well-known square root formula for Reno congestion control:

$$r_C = 1.22 / (R_C * p_C^{0.5}) \quad (5)$$

On the L4S side, we consider the Prague congestion control [I-D.briscoe-iccrp-prague-congestion-control] as the reference for steady-state dependence on congestion. Prague conforms to the same equation as DCTCP, but we do not use the equation derived in the DCTCP paper, which is only appropriate for step marking. The coupled marking, p_{CL} , is the appropriate one when considering throughput equivalence with Classic flows. Unlike step marking, coupled markings are inherently spaced out, so we use the formula for DCTCP packet rate with probabilistic marking derived in Appendix A of [PI2]. We use the equation without RTT-independence enabled, which will be explained later.

$$r_L = 2 / (R_L * p_{CL}) \quad (6)$$

For packet rate equivalence, we equate the two packet rates and rearrange into the same form as Equation (1), so the two can be equated and simplified to produce a formula for a theoretical coupling factor, which we shall call k^* :

$$\begin{aligned} r_C &= r_L \\ \Rightarrow p_C &= (p_{CL}/1.64 * R_L/R_C)^2 \end{aligned}$$

$$p_C = (p_{CL} / k)^2 \quad (1)$$

$$k^* = 1.64 * (R_C / R_L) \quad (7)$$

We say that this coupling factor is theoretical, because it is in terms of two RTTs, which raises two practical questions: i) for multiple flows with different RTTs, the RTT for each traffic class would have to be derived from the RTTs of all the flows in that class (actually the harmonic mean would be needed); ii) a network node cannot easily know the RTT of any of the flows anyway.

RTT-dependence is caused by window-based congestion control, so it ought to be reversed there, not in the network. Therefore, we use a fixed coupling factor in the network, and reduce RTT-dependence in L4S senders. We cannot expect Classic senders to all be updated to reduce their RTT-dependence. But solely addressing the problem in L4S senders at least makes RTT-dependence no worse – not just between L4S senders, but also between L4S and Classic senders.

Traditionally, throughput equivalence has been defined for flows under comparable conditions, including with the same base RTT [RFC2914]. So if we assume the same base RTT, R_b , for comparable flows, we can put both R_C and R_L in terms of R_b .

We can approximate the L4S RTT to be hardly greater than the base RTT, i.e. $R_L \approx R_b$. And we can replace R_C with $(R_b + q_C)$, where the Classic queue, q_C , depends on the target queue delay that the operator has configured for the Classic AQM.

Taking PI2 as an example Classic AQM, it seems that we could just take $R_C = R_b + \text{target}$ (recommended 15 ms by default in Appendix A.1). However, target is roughly the queue depth reached by the tips of the sawteeth of a congestion control, not the average [PI2param]. That is $R_{\text{max}} = R_b + \text{target}$.

The position of the average in relation to the max depends on the amplitude and geometry of the sawteeth. We consider two examples: Reno [RFC5681], as the most sensitive worst-case, and Cubic [RFC8312] in its Reno-friendly mode ('Creno') as the most prevalent congestion control algorithm on the Internet according to the references in [PI2param]. Both are AIMD, so we will generalize using b as the multiplicative decrease factor ($b_r = 0.5$ for Reno, $b_c = 0.7$ for Creno). Then:

$$\begin{aligned} R_C &= (R_{\text{max}} + b \cdot R_{\text{max}}) / 2 \\ &= R_{\text{max}} * (1+b)/2 \end{aligned}$$

$$R_{\text{reno}} = 0.75 * (R_b + \text{target}); \quad R_{\text{creno}} = 0.85 * (R_b + \text{target}). \quad (8)$$

Plugging all this into equation (7) we get a fixed coupling factor for each:

$$\begin{aligned} k_{\text{reno}} &= 1.64 * 0.75 * (R_b + \text{target}) / R_b \\ &= 1.23 * (1 + \text{target} / R_b); \quad k_{\text{creno}} = 1.39 * (1 + \text{target} / R_b) \end{aligned}$$

An operator can then choose the base RTT at which it wants throughput to be equivalent. For instance, if we recommend that the operator chooses $R_b = 25$ ms, as a typical base RTT between Internet users and CDNs [PI2param], then these coupling factors become:

$$\begin{aligned} k_{\text{reno}} &= 1.23 * (1 + 15/25) & k_{\text{creno}} &= 1.39 * (1 + 15/25) \\ &= 1.97 & &= 2.22 \\ &\approx 2 & &\approx 2 \end{aligned} \quad (9)$$

The approximation is relevant to any of the above example DualQ Coupled algorithms, which use a coupling factor that is an integer power of 2 to aid efficient implementation. It also fits best to the worst case (Reno).

To check the outcome of this coupling factor, we can express the ratio of L4S to Classic throughput by substituting from their rate equations (5) and (6), then also substituting for p_C in terms of p_{CL} , using equation (1) with $k=2$ as just determined for the Internet:

$$\begin{aligned} r_L / r_C &= 2 (R_C * p_C^{0.5}) / 1.22 (R_L * p_{CL}) \\ &= (R_C * p_{CL}) / (1.22 * R_L * p_{CL}) \\ &= R_C / (1.22 * R_L) \end{aligned} \tag{10}$$

As an example, we can then consider single competing CReNO and Prague flows, by expressing both their RTTs in (10) in terms of their base RTTs, R_{bC} and R_{bL} . So R_C is replaced by equation (8) for CReNO. And R_L is replaced by the $\max()$ function below, which represents the effective RTT of the current Prague congestion control [I-D.briscoe-iccrp-prague-congestion-control] in its (default) RTT-independent mode, because it sets a floor to the effective RTT that it uses for additive increase:

$$\begin{aligned} \tilde{R} &= 0.85 * (R_{bC} + \text{target}) / (1.22 * \max(R_{bL}, R_{typ})) \\ \tilde{R} &= (R_{bC} + \text{target}) / (1.4 * \max(R_{bL}, R_{typ})) \end{aligned}$$

It can be seen that, for base RTTs below target (15 ms), both the numerator and the denominator plateau, which has the desired effect of limiting RTT-dependence.

At the start of the above derivations, an explanation was promised for why the L4S throughput equation in equation (6) did not need to model RTT-independence. This is because we only use one point - at the the typical base RTT where the operator chooses to calculate the coupling factor. Then, throughput equivalence will at least hold at that chosen point. Nonetheless, assuming Prague senders implement RTT-independence over a range of RTTs below this, the throughput equivalence will then extend over that range as well.

Congestion control designers can choose different ways to reduce RTT-dependence. And each operator can make a policy choice to decide on a different base RTT, and therefore a different k , at which it wants throughput equivalence. Nonetheless, for the Internet, it makes sense to choose what is believed to be the typical RTT most users experience, because a Classic AQM's target queuing delay is also derived from a typical RTT for the Internet.

As a non-Internet example, for localized traffic from a particular ISP's data centre, using the measured RTTs, it was calculated that a value of $k = 8$ would achieve throughput equivalence, and experiments verified the formula very closely.

But, for a typical mix of RTTs across the general Internet, a value of $k=2$ is recommended as a good workable compromise.

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Packetization Layer Path MTU Discovery for Datagram Transports
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Abstract

This document describes a robust method for Path MTU Discovery (PMTUD) for datagram Packetization Layers (PLs). It describes an extension to RFC 1191 and RFC 8201, which specifies ICMP-based Path MTU Discovery for IPv4 and IPv6. The method allows a PL, or a datagram application that uses a PL, to discover whether a network path can support the current size of datagram. This can be used to detect and reduce the message size when a sender encounters a packet black hole (where packets are discarded). The method can probe a network path with progressively larger packets to discover whether the maximum packet size can be increased. This allows a sender to determine an appropriate packet size, providing functionality for datagram transports that is equivalent to the Packetization Layer PMTUD specification for TCP, specified in RFC 4821.

This document updates RFC 4821 to specify the PLPMTUD method for datagram PLs. It also updates RFC 8085 to refer to the method specified in this document instead of the method in RFC 4821 for use with UDP datagrams. Section 7.3 of RFC 4960 recommends an endpoint apply the techniques in RFC 4821 on a per-destination-address basis. RFC 4960, RFC 6951, and RFC 8261 are updated to recommend that SCTP, SCTP encapsulated in UDP and SCTP encapsulated in DTLS use the method specified in this document instead of the method in RFC 4821.

The document also provides implementation notes for incorporating Datagram PMTUD into IETF datagram transports or applications that use datagram transports.

When published, this specification updates RFC 4960, RFC 4821, RFC 8085 and RFC 8261.

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1. Introduction

The IETF has specified datagram transport using UDP, SCTP, and DCCP, as well as protocols layered on top of these transports (e.g., SCTP/UDP, DCCP/UDP, QUIC/UDP), and direct datagram transport over the IP network layer. This document describes a robust method for Path MTU Discovery (PMTUD) that can be used with these transport protocols (or the applications that use their transport service) to discover an appropriate size of packet to use across an Internet path.

1.1. Classical Path MTU Discovery

Classical Path Maximum Transmission Unit Discovery (PMTUD) can be used with any transport that is able to process ICMP Packet Too Big (PTB) messages (e.g., [RFC1191] and [RFC8201]). In this document, the term PTB message is applied to both IPv4 ICMP Unreachable messages (type 3) that carry the error Fragmentation Needed (Type 3, Code 4) [RFC0792] and ICMPv6 Packet Too Big messages (Type 2) [RFC4443]. When a sender receives a PTB message, it reduces the effective MTU to the value reported as the Link MTU in the PTB message. A method from time-to-time increases the packet size in attempt to discover an increase in the supported PMTU. The packets sent with a size larger than the current effective PMTU are known as probe packets.

Packets not intended as probe packets are either fragmented to the current effective PMTU, or the attempt to send fails with an error code. Applications can be provided with a primitive to let them read the Maximum Packet Size (MPS), derived from the current effective PMTU.

Classical PMTUD is subject to protocol failures. One failure arises when traffic using a packet size larger than the actual PMTU is black-holed (all datagrams larger than the actual PMTU, are discarded). This could arise when the PTB messages are not delivered back to the sender for some reason (see for example [RFC2923]).

Examples where PTB messages are not delivered include:

- * The generation of ICMP messages is usually rate limited. This could result in no PTB messages being generated to the sender (see section 2.4 of [RFC4443])
- * ICMP messages can be filtered by middleboxes (including firewalls) [RFC4890]. A firewall could be configured with a policy to block incoming ICMP messages, which would prevent reception of PTB messages to a sending endpoint behind this firewall.

- * When the router issuing the ICMP message drops a tunneled packet, the resulting ICMP message will be directed to the tunnel ingress. This tunnel endpoint is responsible for forwarding the ICMP message and also processing the quoted packet within the payload field to remove the effect of the tunnel, and return a correctly formatted ICMP message to the sender [I-D.ietf-intarea-tunnels]. Failure to do this prevents the PTB message reaching the original sender.
- * Asymmetry in forwarding can result in there being no return route to the original sender, which would prevent an ICMP message being delivered to the sender. This issue can also arise when policy-based routing is used, Equal Cost Multipath (ECMP) routing is used, or a middlebox acts as an application load balancer. An example is where the path towards the server is chosen by ECMP routing depending on bytes in the IP payload. In this case, when a packet sent by the server encounters a problem after the ECMP router, then any resulting ICMP message also needs to be directed by the ECMP router towards the original sender.
- * There are additional cases where the next hop destination fails to receive a packet because of its size. This could be due to misconfiguration of the layer 2 path between nodes, for instance the MTU configured in a layer 2 switch, or misconfiguration of the Maximum Receive Unit (MRU). If a packet is dropped by the link, this will not cause a PTB message to be sent to the original sender.

Another failure could result if a node that is not on the network path sends a PTB message that attempts to force a sender to change the effective PMTU [RFC8201]. A sender can protect itself from reacting to such messages by utilizing the quoted packet within a PTB message payload to validate that the received PTB message was generated in response to a packet that had actually originated from the sender. However, there are situations where a sender would be unable to provide this validation. Examples where validation of the PTB message is not possible include:

- * When a router issuing the ICMP message implements RFC792 [RFC0792], it is only required to include the first 64 bits of the IP payload of the packet within the quoted payload. There could be insufficient bytes remaining for the sender to interpret the quoted transport information.

Note: The recommendation in RFC1812 [RFC1812] is that IPv4 routers return a quoted packet with as much of the original datagram as possible without the length of the ICMP datagram exceeding 576

bytes. IPv6 routers include as much of the invoking packet as possible without the ICMPv6 packet exceeding 1280 bytes [RFC4443].

- * The use of tunnels/encryption can reduce the size of the quoted packet returned to the original source address, increasing the risk that there could be insufficient bytes remaining for the sender to interpret the quoted transport information.
- * Even when the PTB message includes sufficient bytes of the quoted packet, the network layer could lack sufficient context to validate the message, because validation depends on information about the active transport flows at an endpoint node (e.g., the socket/address pairs being used, and other protocol header information).
- * When a packet is encapsulated/tunneled over an encrypted transport, the tunnel/encapsulation ingress might have insufficient context, or computational power, to reconstruct the transport header that would be needed to perform validation.
- * When an ICMP message is generated by a router in a network segment that has inserted a header into a packet, the quoted packet could contain additional protocol header information that was not included in the original sent packet, and which the PL sender does not process or may not know how to process. This could disrupt the ability of the sender to validate this PTB message.
- * A Network Address Translation (NAT) device that translates a packet header, ought to also translate ICMP messages and update the ICMP quoted packet [RFC5508] in that message. If this is not correctly translated then the sender would not be able to associate the message with the PL that originated the packet, and hence this ICMP message cannot be validated.

1.2. Packetization Layer Path MTU Discovery

The term Packetization Layer (PL) has been introduced to describe the layer that is responsible for placing data blocks into the payload of IP packets and selecting an appropriate MPS. This function is often performed by a transport protocol (e.g., DCCP, RTP, SCTP, QUIC), but can also be performed by other encapsulation methods working above the transport layer.

In contrast to PMTUD, Packetization Layer Path MTU Discovery (PLPMTUD) [RFC4821] introduced a method that does not rely upon reception and validation of PTB messages. It is therefore more robust than Classical PMTUD. This has become the recommended approach for implementing discovery of the PMTU [BCP145].

It uses a general strategy where the PL sends probe packets to search for the largest size of unfragmented datagram that can be sent over a network path. Probe packets are sent to explore using a larger packet size. If a probe packet is successfully delivered (as determined by the PL), then the PLPMTU is raised to the size of the successful probe. If a black hole is detected (e.g., where packets of size PLPMTU are consistently not received), the method reduces the PLPMTU.

Datagram PLPMTUD introduces flexibility in implementation. At one extreme, it can be configured to only perform Black Hole Detection and recovery with increased robustness compared to Classical PMTUD. At the other extreme, all PTB processing can be disabled, and PLPMTUD replaces Classical PMTUD.

PLPMTUD can also include additional consistency checks without increasing the risk that data is lost when probing to discover the Path MTU. For example, information available at the PL, or higher layers, enables received PTB messages to be validated before being utilized.

1.3. Path MTU Discovery for Datagram Services

Section 5 of this document presents a set of algorithms for datagram protocols to discover the largest size of unfragmented datagram that can be sent over a network path. The method relies upon features of the PL described in Section 3 and applies to transport protocols operating over IPv4 and IPv6. It does not require cooperation from the lower layers, although it can utilize PTB messages when these received messages are made available to the PL.

The message size guidelines in section 3.2 of the UDP Usage Guidelines [BCP145] state "an application SHOULD either use the Path MTU information provided by the IP layer or implement Path MTU Discovery (PMTUD)", but does not provide a mechanism for discovering the largest size of unfragmented datagram that can be used on a network path. The present document updates RFC 8085 to specify this method in place of PLPMTUD [RFC4821] and provides a mechanism for sharing the discovered largest size as the MPS (see Section 4.4).

Section 10.2 of [RFC4821] recommended a PLPMTUD probing method for the Stream Control Transport Protocol (SCTP). SCTP utilizes probe packets consisting of a minimal sized HEARTBEAT chunk bundled with a PAD chunk as defined in [RFC4820]. However, RFC 4821 did not provide a complete specification. The present document replaces that description by providing a complete specification.

The Datagram Congestion Control Protocol (DCCP) [RFC4340] requires implementations to support Classical PMTUD and states that a DCCP sender "MUST maintain the MPS allowed for each active DCCP session". It also defines the current congestion control MPS (CCMPS) supported by a network path. This recommends use of PMTUD, and suggests use of control packets (DCCP-Sync) as path probe packets, because they do not risk application data loss. The method defined in this specification can be used with DCCP.

Section 4 and Section 5 define the protocol mechanisms and specification for Datagram Packetization Layer Path MTU Discovery (DPLPMTUD).

Section 6 specifies the method for datagram transports and provides information to enable the implementation of PLPMTUD with other datagram transports and applications that use datagram transports.

Section 6 also provides updated recommendations for [RFC6951] and [RFC8261].

2. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

The following terminology is defined. Relevant terms are directly copied from [RFC4821], and the definitions in [RFC1122].

Acknowledged PL: A PL that includes a mechanism that can confirm successful delivery of datagrams to the remote PL endpoint (e.g., SCTP). Typically, the PL receiver returns acknowledgments corresponding to the received datagrams, which can be utilised to detect black-holing of packets (c.f., Unacknowledged PL).

Actual PMTU: The Actual PMTU is the PMTU of a network path between a sender PL and a destination PL, which the DPLPMTUD algorithm seeks to determine.

Black Hole: A Black Hole is encountered when a sender is unaware that packets are not being delivered to the destination end point. Two types of Black Hole are relevant to DPLPMTUD:

- * Packets encounter a packet Black Hole when packets are not delivered to the destination endpoint (e.g., when the sender transmits packets of a particular size with a previously known

effective PMTU and they are discarded by the network).

- * An ICMP Black Hole is encountered when the sender is unaware that packets are not delivered to the destination endpoint because PTB messages are not received by the originating PL sender.

Classical Path MTU Discovery: Classical PMTUD is a process described in [RFC1191] and [RFC8201], in which nodes rely on PTB messages to learn the largest size of unfragmented packet that can be used across a network path.

Datagram: A datagram is a transport-layer protocol data unit, transmitted in the payload of an IP packet.

Effective PMTU: The Effective PMTU is the current estimated value for PMTU that is used by a PMTUD. This is equivalent to the PLPMTU derived by PLPMTUD plus the size of any headers added below the PL, including the IP layer headers.

EMTU_S: The Effective MTU for sending (EMTU_S) is defined in [RFC1122] as "the maximum IP datagram size that may be sent, for a particular combination of IP source and destination addresses...".

EMTU_R: The Effective MTU for receiving (EMTU_R) is designated in [RFC1122] as "the largest datagram size that can be reassembled".

Link: A Link is a communication facility or medium over which nodes can communicate at the link layer, i.e., a layer below the IP layer. Examples are Ethernet LANs and Internet (or higher) layer tunnels.

Link MTU: The Link Maximum Transmission Unit (MTU) is the size in bytes of the largest IP packet, including the IP header and payload, that can be transmitted over a link. Note that this could more properly be called the IP MTU, to be consistent with how other standards organizations use the acronym. This includes the IP header, but excludes link layer headers and other framing that is not part of IP or the IP payload. Other standards organizations generally define the link MTU to include the link layer headers. This specification continues the requirement in [RFC4821], that states "All links MUST enforce their MTU: links that might non-deterministically deliver packets that are larger than their rated MTU MUST consistently discard such packets."

MAX_PLPMTU: The MAX_PLPMTU is the largest size of PLPMTU that DPLPMTUD will attempt to use (see the constants defined in Section 5.1.2).

MIN_PLPMTU: The MIN_PLPMTU is the smallest size of PLPMTU that DPLPMTUD will attempt to use (see the constants defined in Section 5.1.2).

MPS: The Maximum Packet Size (MPS) is the largest size of application data block that can be sent across a network path by a PL using a single Datagram (see Section 4.4).

MSL: Maximum Segment Lifetime (MSL) The maximum delay a packet is expected to experience across a path, taken as 2 minutes [BCP145].

Packet: A Packet is the IP header(s) and any extension headers/options plus the IP payload.

Packetization Layer (PL): The PL is a layer of the network stack that places data into packets and performs transport protocol functions. Examples of a PL include: TCP, SCTP, SCTP over UDP, SCTP over DTLS, or QUIC.

Path: The Path is the set of links and routers traversed by a packet between a source node and a destination node by a particular flow.

Path MTU (PMTU): The Path MTU (PMTU) is the minimum of the Link MTU of all the links forming a network path between a source node and a destination node, as used by PMTUD.

PTB: In this document, the term PTB message is applied to both IPv4 ICMP Unreachable messages (type 3) that carry the error Fragmentation Needed (Type 3, Code 4) [RFC0792] and ICMPv6 Packet Too Big messages (Type 2) [RFC4443].

PTB_SIZE: The PTB_SIZE is a value reported in a validated PTB message that indicates next hop link MTU of a router along the path.

PL_PTBSIZE: The size reported in a validated PTB message, reduced by the size of all headers added by layers below the PL.

PLPMTU: The Packetization Layer PMTU is an estimate of the largest size of PL datagram that can be sent by a path, controlled by PLPMTUD.

PLPMTUD: Packetization Layer Path MTU Discovery (PLPMTUD), the method described in this document for datagram PLs, which is an extension to Classical PMTU Discovery.

Probe packet: A probe packet is a datagram sent with a purposely chosen size (typically the current PLPMTU or larger) to detect if

packets of this size can be successfully sent end-to-end across the network path.

Unacknowledged PL: A PL that does not itself provide a mechanism to confirm delivery of datagrams to the remote PL endpoint (e.g., UDP), and therefore requires DPLPMTUD to provide a mechanism to detect black-holing of packets (c.f., Acknowledged PL).

3. Features Required to Provide Datagram PLPMTUD

The principles expressed in [RFC4821] apply to the use of the technique with any PL. TCP PLPMTUD has been defined using standard TCP protocol mechanisms. Unlike TCP, a datagram PL requires additional mechanisms and considerations to implement PLPMTUD.

The requirements for datagram PLPMTUD are:

1. Managing the PLPMTU: For datagram PLs, the PLPMTU is managed by DPLPMTUD. A PL MUST NOT send a datagram (other than a probe packet) with a size at the PL that is larger than the current PLPMTU.
2. Probe packets: The network interface below PL is REQUIRED to provide a way to transmit a probe packet that is larger than the PLPMTU. In IPv4, a probe packet MUST be sent with the Don't Fragment (DF) bit set in the IP header, and without network layer endpoint fragmentation. In IPv6, a probe packet is always sent without source fragmentation (as specified in section 5.4 of [RFC8201]).
3. Reception feedback: The destination PL endpoint is REQUIRED to provide a feedback method that indicates to the DPLPMTUD sender when a probe packet has been received by the destination PL endpoint. Section 6 provides examples of how a PL can provide this acknowledgment of received probe packets.
4. Probe loss recovery: It is RECOMMENDED to use probe packets that do not carry any user data that would require retransmission if lost. Most datagram transports permit this. If a probe packet contains user data requiring retransmission in case of loss, the PL (or layers above) are REQUIRED to arrange any retransmission/repair of any resulting loss. The PL is REQUIRED to be robust in the case where probe packets are lost due to other reasons (including link transmission error, congestion).
5. PMTU parameters: A DPLPMTUD sender is RECOMMENDED to utilize information about the maximum size of packet that can be transmitted by the sender on the local link (e.g., the local Link

MTU). A PL sender MAY utilize similar information about the maximum size of network layer packet that a receiver can accept when this is supplied (note this could be less than EMTU_R). This avoids implementations trying to send probe packets that can not be transferred by the local link. Too high of a value could reduce the efficiency of the search algorithm. Some applications also have a maximum transport protocol data unit (PDU) size, in which case there is no benefit from probing for a size larger than this (unless a transport allows multiplexing multiple applications PDUs into the same datagram).

6. Processing PTB messages: A DPLPMTUD sender MAY optionally utilize PTB messages received from the network layer to help identify when a network path does not support the current size of probe packet. Any received PTB message MUST be validated before it is used to update the PLPMTU discovery information [RFC8201]. This validation confirms that the PTB message was sent in response to a packet originating by the sender, and needs to be performed before the PLPMTU discovery method reacts to the PTB message. A PTB message MUST NOT be used to increase the PLPMTU [RFC8201], but could trigger a probe to test for a larger PLPMTU. A valid PTB_SIZE is converted to a PL_PTBSIZE before it is to be used in the DPLPMTUD state machine. A PL_PTBSIZE that is greater than that currently probed SHOULD be ignored. (This PTB message ought to be discarded without further processing, but could be utilized as an input that enables a resilience mode).
7. Probing and congestion control: A PL MAY use a congestion controller to decide when to send a probe packet. If transmission of probe packets is limited by the congestion controller, this could result in transmission of probe packets being delayed or suspended during congestion. When the transmission of probe packets is not controlled by the congestion controller, the interval between probe packets MUST be at least one RTT. Loss of a probe packet SHOULD NOT be treated as an indication of congestion and SHOULD NOT trigger a congestion control reaction [RFC4821], because this could result in unnecessary reduction of the sending rate. An update to the PLPMTU (or MPS) MUST NOT increase the congestion window measured in bytes [RFC4821]. Therefore, an increase in the packet size does not cause an increase in the data rate in bytes per second. A PL that maintains the congestion window in terms of a limit to the number of outstanding fixed size packets SHOULD adapt this limit to compensate for the size of the actual packets. The transmission of probe packets can interact with the operation of a PL that performs burst mitigation or pacing and could need transmission of probe packets to be regulated by these methods.

8. Probing and flow control: Flow control at the PL concerns the end-to-end flow of data using the PL service. Flow control SHOULD NOT apply to DPLPMTU when probe packets use a design that does not carry user data to the remote application.
9. Shared PLPMTU state: The PMTU value calculated from the PLPMTU MAY also be stored with the corresponding entry associated with the destination in the IP layer cache, and used by other PL instances. The specification of PLPMTUD [RFC4821] states: "If PLPMTUD updates the MTU for a particular path, all Packetization Layer sessions that share the path representation (as described in Section 5.2 of [RFC4821]) SHOULD be notified to make use of the new MTU". Such methods MUST be robust to the wide variety of underlying network forwarding behaviors. Section 5.2 of [RFC8201] provides guidance on the caching of PMTU information and also the relation to IPv6 flow labels.

In addition, the following principles are stated for design of a DPLPMTUD method:

- * A PL MAY be designed to segment data blocks larger than the MPS into multiple datagrams. However, not all datagram PLs support segmentation of data blocks. It is RECOMMENDED that methods avoid forcing an application to use an arbitrary small MPS for transmission while the method is searching for the currently supported PLPMTU. A reduced MPS can adversely impact the performance of an application.
- * To assist applications in choosing a suitable data block size, the PL is RECOMMENDED to provide a primitive that returns the MPS derived from the PLPMTU to the higher layer using the PL. The value of the MPS can change following a change in the path, or loss of probe packets.
- * Path validation: It is RECOMMENDED that methods are robust to path changes that could have occurred since the path characteristics were last confirmed, and to the possibility of inconsistent path information being received.
- * Datagram reordering: A method is REQUIRED to be robust to the possibility that a flow encounters reordering, or the traffic (including probe packets) is divided over more than one network path.
- * Datagram delay and duplication: The feedback mechanism is REQUIRED to be robust to the possibility that packets could be significantly delayed or duplicated along a network path.

- * When to probe: It is RECOMMENDED that methods determine whether the path has changed since it last measured the path. This can help determine when to probe the path again.

4. DPLPMTUD Mechanisms

This section lists the protocol mechanisms used in this specification.

4.1. PLPMTU Probe Packets

The DPLPMTUD method relies upon the PL sender being able to generate probe packets with a specific size. TCP is able to generate these probe packets by choosing to appropriately segment data being sent [RFC4821]. In contrast, a datagram PL that constructs a probe packet has to either request an application to send a data block that is larger than that generated by an application, or to utilize padding functions to extend a datagram beyond the size of the application data block. Protocols that permit exchange of control messages (without an application data block) can generate a probe packet by extending a control message with padding data. The total size of a probe packet includes all headers and padding added to the payload data being sent (e.g., including protocol option fields, security-related fields such as an Authenticated Encryption with Associated Data (AEAD) tag and TLS record layer padding).

A receiver is REQUIRED to be able to distinguish an in-band data block from any added padding. This is needed to ensure that any added padding is not passed on to an application at the receiver.

This results in three possible ways that a sender can create a probe packet:

Probing using padding data: A probe packet that contains only control information together with any padding, which is needed to be inflated to the size of the probe packet. Since these probe packets do not carry an application-supplied data block, they do not typically require retransmission, although they do still consume network capacity and incur endpoint processing.

Probing using application data and padding data: A probe packet that contains a data block supplied by an application that is combined with padding to inflate the length of the datagram to the size of the probe packet.

Probing using application data: A probe packet that contains a data block supplied by an application that matches the size of the

probe packet. This method requests the application to issue a data block of the desired probe size.

A PL that uses a probe packet carrying application data and needs protection from the loss of this probe packet could perform transport-layer retransmission/repair of the data block (e.g., by retransmission after loss is detected or by duplicating the data block in a datagram without the padding data). This retransmitted data block might possibly need to be sent using a smaller PLPMTU, which could force the PL to use a smaller packet size to traverse the end-to-end path. (This could utilize endpoint network-layer fragmentation or a PL that can re-segment the data block into multiple datagrams).

DPLPMTUD MAY choose to use only one of these methods to simplify the implementation.

Probe messages sent by a PL MUST contain enough information to uniquely identify the probe within Maximum Segment Lifetime (e.g., including a unique identifier from the PL or the DPLPMTUD implementation), while being robust to reordering and replay of probe response and PTB messages.

4.2. Confirmation of Probed Packet Size

The PL needs a method to determine (confirm) when probe packets have been successfully received end-to-end across a network path.

Transport protocols can include end-to-end methods that detect and report reception of specific datagrams that they send (e.g., DCCP, SCTP, and QUIC provide keep-alive/heartbeat features). When supported, this mechanism MAY also be used by DPLPMTUD to acknowledge reception of a probe packet.

A PL that does not acknowledge data reception (e.g., UDP and UDP-Lite) is unable itself to detect when the packets that it sends are discarded because their size is greater than the actual PMTU. These PLs need to rely on an application protocol to detect this loss.

Section 6 specifies this function for a set of IETF-specified protocols.

4.3. Black Hole Detection and Reducing the PLPMTU

The description that follows uses the set of constants defined in Section 5.1.2 and variables defined in Section 5.1.3.

Black Hole Detection is triggered by an indication that the network path could be unable to support the current PLPMTU size.

There are three indicators that can detect black holes:

- * A validated PTB message can be received that indicates a PL_PTB_SIZE less than the current PLPMTU. A DPLPMTUD method MUST NOT rely solely on this method.
- * A PL can use the DPLPMTUD probing mechanism to periodically generate probe packets of the size of the current PLPMTU (e.g., using the confirmation timer Section 5.1.1). A timer tracks whether acknowledgments are received. Successive loss of probes is an indication that the current path no longer supports the PLPMTU (e.g., when the number of probe packets sent without receiving an acknowledgment, PROBE_COUNT, becomes greater than MAX_PROBES).
- * A PL can utilize an event that indicates the network path no longer sustains the sender's PLPMTU size. This could use a mechanism implemented within the PL to detect excessive loss of data sent with a specific packet size and then conclude that this excessive loss could be a result of an invalid PLPMTU (as in PLPMTUD for TCP [RFC4821]).

The three methods can result in different transmission patterns for packet probes and are expected to result in different responsiveness following a change in the actual PMTU.

A PL MAY inhibit sending probe packets when no application data has been sent since the previous probe packet. A PL that resumes sending user data MAY continue PLPMTU discovery for each path. This allows it to use an up-to-date PLPMTU. However, this could result in additional packets being sent.

When the method detects the current PLPMTU is not supported, DPLPMTUD sets a lower PLPMTU, and sets a lower MPS. The PL then confirms that the new PLPMTU can be successfully used across the path. A probe packet could need to have a size less than the size of the data block generated by the application.

4.4. The Maximum Packet Size (MPS)

The result of probing determines a usable PLPMTU, which is used to set the MPS used by the application. The MPS is smaller than the PLPMTU because it is reduced by the size of PL headers (including the overhead of security-related fields such as an AEAD tag and TLS record layer padding). The relationship between the MPS and the PLPMTU is illustrated in Figure 1.

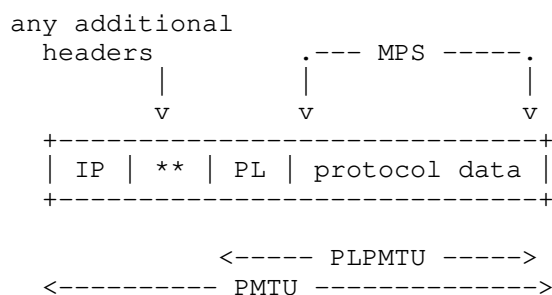


Figure 1: Relationship between MPS and PLPMTU

A PL is unable to send a packet (other than a probe packet) with a size larger than the current PLPMTU at the network layer. To avoid this, a PL MAY be designed to segment data blocks larger than the MPS into multiple datagrams.

DPLPMTUD seeks to avoid IP fragmentation. An attempt to send a data block larger than the MPS will therefore fail if a PL is unable to segment data. To determine the largest data block that can be sent, a PL SHOULD provide applications with a primitive that returns the MPS, derived from the current PLPMTU.

If DPLPMTUD results in a change to the MPS, the application needs to adapt to the new MPS. A particular case can arise when packets have been sent with a size less than the MPS and the PLPMTU was subsequently reduced. If these packets are lost, the PL MAY segment the data using the new MPS. If a PL is unable to re-segment a previously sent datagram (e.g., [RFC4960]), then the sender either discards the datagram or could perform retransmission using network-layer fragmentation to form multiple IP packets not larger than the PLPMTU. For IPv4, the use of endpoint fragmentation by the sender is preferred over clearing the DF bit in the IPv4 header. Operational experience reveals that IP fragmentation can reduce the reliability of Internet communication [I-D.ietf-intarea-frag-fragile], which may reduce the probability of successful retransmission.

4.5. Disabling the Effect of PMTUD

A PL implementing this specification MUST suspend network layer processing of outgoing packets that enforces a PMTU [RFC1191][RFC8201] for each flow utilizing DPLPMTUD, and instead use DPLPMTUD to control the size of packets that are sent by a flow. This removes the need for the network layer to drop or fragment sent packets that have a size greater than the PMTU.

4.6. Response to PTB Messages

This method requires the DPLPMTUD sender to validate any received PTB message before using the PTB information. The response to a PTB message depends on the PL_PTB_SIZE calculated from the PTB_SIZE in the PTB message, the state of the PLPMTUD state machine, and the IP protocol being used.

Section 4.6.1 first describes validation for both IPv4 ICMP Unreachable messages (type 3) and ICMPv6 Packet Too Big messages, both of which are referred to as PTB messages in this document.

4.6.1. Validation of PTB Messages

This section specifies utilization and validation of PTB messages.

- * A simple implementation MAY ignore received PTB messages and in this case the PLPMTU is not updated when a PTB message is received.
- * A PL that supports PTB messages MUST validate these messages before they are further processed.

A PL that receives a PTB message from a router or middlebox performs ICMP validation (see Section 4 of [RFC8201] and Section 5.2 of [BCP145]). Because DPLPMTUD operates at the PL, the PL needs to check that each received PTB message is received in response to a packet transmitted by the endpoint PL performing DPLPMTUD.

The PL MUST check the protocol information in the quoted packet carried in an ICMP PTB message payload to validate the message originated from the sending node. This validation includes determining that the combination of the IP addresses, the protocol, the source port and destination port match those returned in the quoted packet - this is also necessary for the PTB message to be passed to the corresponding PL.

The validation SHOULD utilize information that it is not simple for an off-path attacker to determine [BCP145]. For example, it could

check the value of a protocol header field known only to the two PL endpoints. A datagram application that uses well-known source and destination ports ought to also rely on other information to complete this validation.

These checks are intended to provide protection from packets that originate from a node that is not on the network path. A PTB message that does not complete the validation MUST NOT be further utilized by the DPLPMTUD method, as discussed in the Security Considerations section.

Section 4.6.2 describes this processing of PTB messages.

4.6.2. Use of PTB Messages

PTB messages that have been validated MAY be utilized by the DPLPMTUD algorithm, but MUST NOT be used directly to set the PLPMTU.

Before using the size reported in the PTB message it must first be converted to a PL_PTB_SIZE. The PL_PTB_SIZE is smaller than the PTB_SIZE because it is reduced by headers below the PL including any IP options or extensions added to the PL packet.

A method that utilizes these PTB messages can improve the speed at which the algorithm detects an appropriate PLPMTU by triggering an immediate probe for the PL_PTB_SIZE (resulting in a network-layer packet of size PTB_SIZE), compared to one that relies solely on probing using a timer-based search algorithm.

A set of checks are intended to provide protection from a router that reports an unexpected PTB_SIZE. The PL also needs to check that the indicated PL_PTB_SIZE is less than the size used by probe packets and at least the minimum size accepted.

This section provides a summary of how PTB messages can be utilized. (This uses the set of constants defined in Section 5.1.2). This processing depends on the PL_PTB_SIZE and the current value of a set of variables:

PL_PTB_SIZE < MIN_PLPMTU

- * Invalid PL_PTB_SIZE see Section 4.6.1.
- * PTB message ought to be discarded without further processing (i.e., PLPMTU is not modified).
- * The information could be utilized as an input that triggers enabling a resilience mode (see Section 5.3.3).

MIN_PLPMTU < PL_PTB_SIZE < BASE_PLPMTU

- * A robust PL MAY enter an error state (see Section 5.2) for an IPv4 path when the PL_PTB_SIZE reported in the PTB message is larger than or equal to 68 bytes [RFC0791] and when this is less than the BASE_PLPMTU.
- * A robust PL MAY enter an error state (see Section 5.2) for an IPv6 path when the PL_PTB_SIZE reported in the PTB message is larger than or equal to 1280 bytes [RFC8200] and when this is less than the BASE_PLPMTU.

BASE_PLPMTU <= PL_PTB_SIZE < PLPMTU

- * This could be an indication of a black hole. The PLPMTU SHOULD be set to BASE_PLPMTU (the PLPMTU is reduced to the BASE_PLPMTU to avoid unnecessary packet loss when a black hole is encountered).
- * The PL ought to start a search to quickly discover the new PLPMTU. The PL_PTB_SIZE reported in the PTB message can be used to initialize a search algorithm.

PLPMTU < PL_PTB_SIZE < PROBED_SIZE

- * The PLPMTU continues to be valid, but the size of a packet used to search (PROBED_SIZE) was larger than the actual PMTU.
- * The PLPMTU is not updated.
- * The PL can use the reported PL_PTB_SIZE from the PTB message as the next search point when it resumes the search algorithm.

PL_PTB_SIZE >= PROBED_SIZE

- * Inconsistent network signal.
- * PTB message ought to be discarded without further processing (i.e., PLPMTU is not modified).
- * The information could be utilized as an input to trigger enabling a resilience mode.

5. Datagram Packetization Layer PMTUD

This section specifies Datagram PLPMTUD (DPLPMTUD). The method can be introduced at various points (as indicated with * in the figure below) in the IP protocol stack to discover the PLPMTU so that an application can utilize an appropriate MPS for the current network path.

DPLPMTUD SHOULD only be performed at one layer between a pair of endpoints. Therefore, an upper PL or application should avoid using DPLPMTUD when this is already enabled in a lower layer. A PL MUST adjust the MPS indicated by DPLPMTUD to account for any additional overhead introduced by the PL.

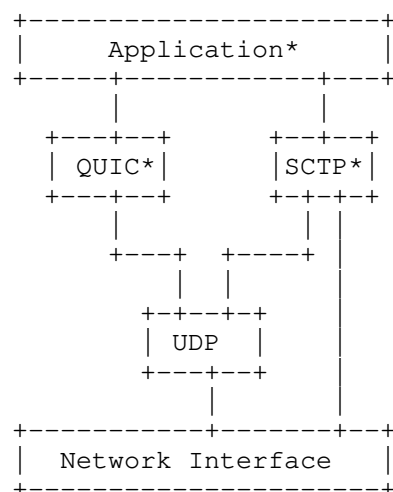


Figure 2: Examples where DPLPMTUD can be implemented

The central idea of DPLPMTUD is probing by a sender. Probe packets are sent to find the maximum size of user message that can be completely transferred across the network path from the sender to the destination.

The following sections identify the components needed for implementation, provides an overview of the phases of operation, and specifies the state machine and search algorithm.

5.1. DPLPMTUD Components

This section describes the timers, constants, and variables of DPLPMTUD.

5.1.1. Timers

The method utilizes up to three timers:

PROBE_TIMER: The PROBE_TIMER is configured to expire after a period longer than the maximum time to receive an acknowledgment to a probe packet. This value MUST NOT be smaller than 1 second, and SHOULD be larger than 15 seconds. Guidance on selection of the

timer value are provided in Section 3.1.1 of the UDP Usage Guidelines [BCP145].

PMTU_RAISE_TIMER: The PMTU_RAISE_TIMER is configured to the period a sender will continue to use the current PLPMTU, after which it re-enters the Search phase. This timer has a period of 600 seconds, as recommended by PLPMTUD [RFC4821].

DPLPMTUD MAY inhibit sending probe packets when no application data has been sent since the previous probe packet. A PL preferring to use an up-to-date PMTU once user data is sent again, can choose to continue PMTU discovery for each path. However, this will result in sending additional packets.

CONFIRMATION_TIMER: When an acknowledged PL is used, this timer MUST NOT be used. For other PLs, the CONFIRMATION_TIMER is configured to the period a PL sender waits before confirming the current PLPMTU is still supported. This is less than the PMTU_RAISE_TIMER and used to decrease the PLPMTU (e.g., when a black hole is encountered). Confirmation needs to be frequent enough when data is flowing that the sending PL does not black hole extensive amounts of traffic. Guidance on selection of the timer value are provided in Section 3.1.1 of the UDP Usage Guidelines [BCP145].

DPLPMTUD MAY inhibit sending probe packets when no application data has been sent since the previous probe packet. A PL preferring to use an up-to-date PMTU once user data is sent again, can choose to continue PMTU discovery for each path. However, this could result in sending additional packets.

DPLPMTD specifies various timers, however an implementation could choose to realise these timer functions using a single timer.

5.1.2. Constants

The following constants are defined:

MAX_PROBES: The MAX_PROBES is the maximum value of the PROBE_COUNT counter (see Section 5.1.3). MAX_PROBES represents the limit for the number of consecutive probe attempts of any size. Search algorithms benefit from a MAX_PROBES value greater than 1 because this can provide robustness to isolated packet loss. The default value of MAX_PROBES is 3.

MIN_PLPMTU: The MIN_PLPMTU is the smallest size of PLPMTU that DPLPMTUD will attempt to use. An endpoint could need to be configure the MIN_PLPMTU to provide space for extension headers and other encapsulations at layers below the PL. This value can

be interface and path dependent. For IPv6, this size is greater than or equal to the size at the PL that results in an 1280 byte IPv6 packet, as specified in [RFC8200]. For IPv4, this size is greater than or equal to the size at the PL that results in an 68 byte IPv4 packet. Note: An IPv4 router is required to be able to forward a datagram of 68 bytes without further fragmentation. This is the combined size of an IPv4 header and the minimum fragment size of 8 bytes. In addition, receivers are required to be able to reassemble fragmented datagrams at least up to 576 bytes, as stated in section 3.3.3 of [RFC1122].

MAX_PLPMTU: The MAX_PLPMTU is the largest size of PLPMTU. This has to be less than or equal to the maximum size of the PL packet that can be sent on the outgoing interface (constrained by the local interface MTU). When known, this also ought to be less than the maximum size of PL packet that can be received by the remote endpoint (constrained by EMTU_R). It can be limited by the design or configuration of the PL being used. An application, or PL, MAY choose a smaller MAX_PLPMTU when there is no need to send packets larger than a specific size.

BASE_PLPMTU: The BASE_PLPMTU is a configured size expected to work for most paths. The size is equal to or larger than the MIN_PLPMTU and smaller than the MAX_PLPMTU. For most PLs a suitable BASE_PLPMTU will be larger than 1200 bytes. When using IPv4, there is no currently equivalent size specified and a default BASE_PLPMTU of 1200 bytes is RECOMMENDED.

5.1.3. Variables

This method utilizes a set of variables:

PROBED_SIZE: The PROBED_SIZE is the size of the current probe packet as determined at the PL. This is a tentative value for the PLPMTU, which is awaiting confirmation by an acknowledgment.

PROBE_COUNT: The PROBE_COUNT is a count of the number of successive unsuccessful probe packets that have been sent. Each time a probe packet is acknowledged, the value is set to zero. (Some probe loss is expected while searching, therefore loss of a single probe is not an indication of a PMTU problem.)

The figure below illustrates the relationship between the packet size constants and variables at a point of time when the DPLPMTUD algorithm performs path probing to increase the size of the PLPMTU. A probe packet has been sent of size PROBED_SIZE. Once this is acknowledged, the PLPMTU will raise to PROBED_SIZE allowing the

DPLPMTUD algorithm to further increase PROBED_SIZE toward sending a probe with the size of the actual PMTU.

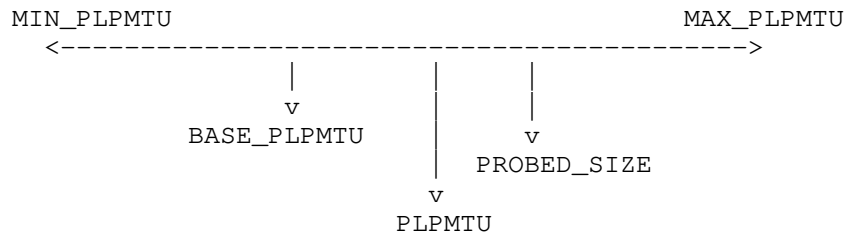


Figure 3: Relationships between packet size constants and variables

5.1.4. Overview of DPLPMTUD Phases

This section provides a high-level informative view of the DPLPMTUD method, by describing the movement of the method through several phases of operation. More detail is available in the state machine Section 5.2.

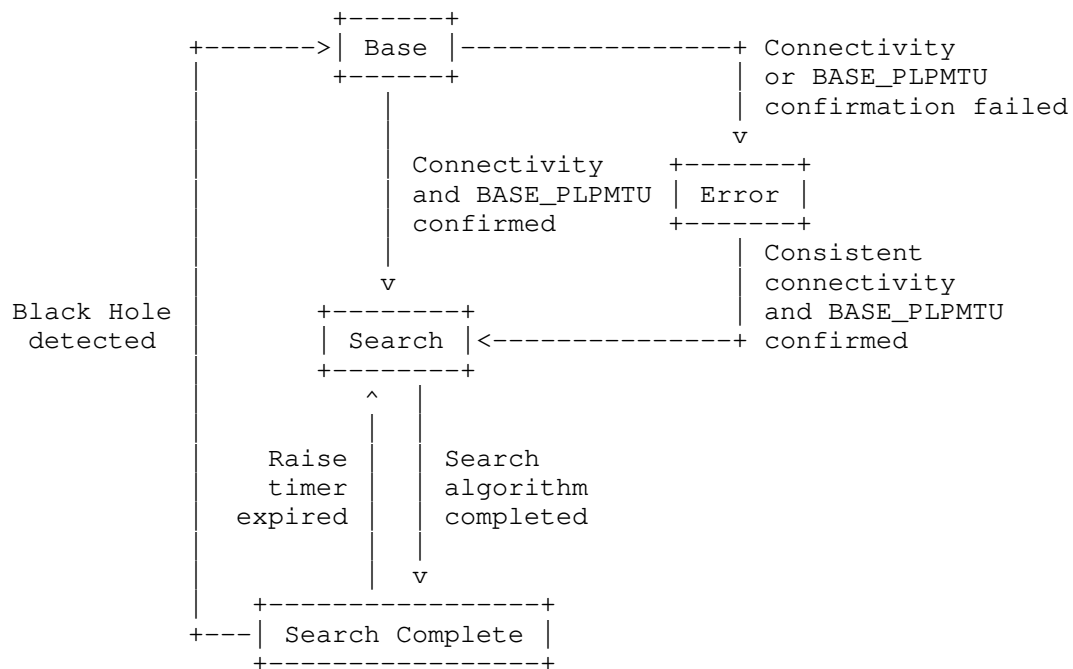


Figure 4: DPLPMTUD Phases

Base: The Base Phase confirms connectivity to the remote peer using packets of the `BASE_PLPMTU`. The confirmation of connectivity is implicit for a connection-oriented PL (where it can be performed in a PL connection handshake). A connectionless PL sends a probe packet and uses acknowledgment of this probe packet to confirm that the remote peer is reachable.

The sender also confirms that `BASE_PLPMTU` is supported across the network path. This may be achieved using a PL mechanism (e.g., using a handshake packet of size `BASE_PLPMTU`), or by sending a probe packet of size `BASE_PLPMTU` and confirming that this is received.

A probe packet of size `BASE_PLPMTU` can be sent immediately on the initial entry to the Base Phase (following a connectivity check). A PL that does not wish to support a path with a PLPMTU less than `BASE_PLPMTU` can simplify the phase into a single step by performing the connectivity checks with a probe of the `BASE_PLPMTU` size.

Once confirmed, DPLPMTUD enters the Search Phase. If the Base Phase fails to confirm the `BASE_PLPMTU`, DPLPMTUD enters the Error Phase.

Search: The Search Phase utilizes a search algorithm to send probe packets to seek to increase the PLPMTU. The algorithm concludes when it has found a suitable PLPMTU, by entering the Search Complete Phase.

A PL could respond to PTB messages using the PTB to advance or terminate the search, see Section 4.6.

Search Complete: The Search Complete Phase is entered when the PLPMTU is supported across the network path. A PL can use a `CONFIRMATION_TIMER` to periodically repeat a probe packet for the current PLPMTU size. If the sender is unable to confirm reachability (e.g., if the `CONFIRMATION_TIMER` expires) or the PL signals a lack of reachability, a black hole has been detected and DPLPMTUD enters the Base phase.

The `PMTU_RAISE_TIMER` is used to periodically resume the search phase to discover if the PLPMTU can be raised. Black Hole Detection causes the sender to enter the Base Phase.

Error: The Error Phase is entered when there is conflicting or invalid PLPMTU information for the path (e.g., a failure to support the `BASE_PLPMTU`) that cause DPLPMTUD to be unable to progress and the PLPMTU is lowered.

DPLPMTUD remains in the Error Phase until a consistent view of the path can be discovered and it has also been confirmed that the path supports the BASE_PLPMTU (or DPLPMTUD is suspended).

A method that only reduces the PLPMTU to a suitable size would be sufficient to ensure reliable operation, but can be very inefficient when the actual PMTU changes or when the method (for whatever reason) makes a suboptimal choice for the PLPMTU.

A full implementation of DPLPMTUD provides an algorithm enabling the DPLPMTUD sender to increase the PLPMTU following a change in the characteristics of the path, such as when a link is reconfigured with a larger MTU, or when there is a change in the set of links traversed by an end-to-end flow (e.g., after a routing or path fail-over decision).

5.2. State Machine

A state machine for DPLPMTUD is depicted in Figure 5. If multipath or multihoming is supported, a state machine is needed for each path.

Note: Not all changes are shown to simplify the diagram.

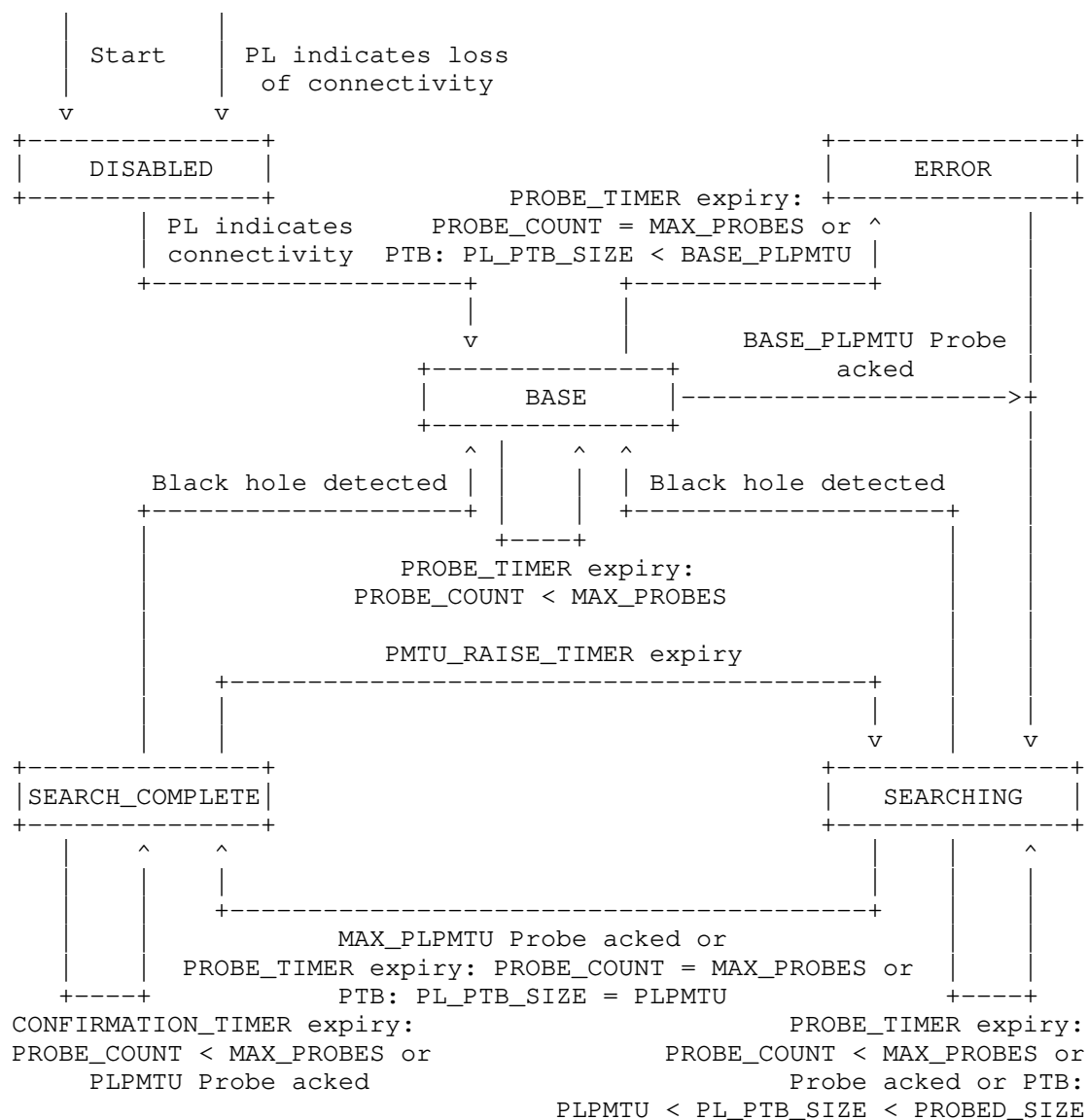


Figure 5: State machine for Datagram PLPMTUD

The following states are defined:

DISABLED: The DISABLED state is the initial state before probing has started. It is also entered from any other state, when the PL indicates loss of connectivity. This state is left once the PL

indicates connectivity to the remote PL. When transitioning to the BASE state, a probe packet of size BASE_PLPMTU can be sent immediately.

BASE: The BASE state is used to confirm that the BASE_PLPMTU size is supported by the network path and is designed to allow an application to continue working when there are transient reductions in the actual PMTU. It also seeks to avoid long periods when a sender searching for a larger PLPMTU is unaware that packets are not being delivered due to a packet or ICMP Black Hole.

On entry, the PROBED_SIZE is set to the BASE_PLPMTU size and the PROBE_COUNT is set to zero.

Each time a probe packet is sent, the PROBE_TIMER is started. The state is exited when the probe packet is acknowledged, and the PL sender enters the SEARCHING state.

The state is also left when the PROBE_COUNT reaches MAX_PROBES or a received PTB message is validated. This causes the PL sender to enter the ERROR state.

SEARCHING: The SEARCHING state is the main probing state. This state is entered when probing for the BASE_PLPMTU completes.

Each time a probe packet is acknowledged, the PROBE_COUNT is set to zero, the PLPMTU is set to the PROBED_SIZE and then the PROBED_SIZE is increased using the search algorithm (as described in Section 5.3).

When a probe packet is sent and not acknowledged within the period of the PROBE_TIMER, the PROBE_COUNT is incremented and a new probe packet is transmitted.

The state is exited to enter SEARCH_COMPLETE when the PROBE_COUNT reaches MAX_PROBES, a validated PTB is received that corresponds to the last successfully probed size (PL_PTB_SIZE = PLPMTU), or a probe of size MAX_PLPMTU is acknowledged (PLPMTU = MAX_PLPMTU).

When a black hole is detected in the SEARCHING state, this causes the PL sender to enter the BASE state.

SEARCH_COMPLETE: The SEARCH_COMPLETE state indicates that a search has completed. This is the normal maintenance state, where the PL is not probing to update the PLPMTU. DPLPMTUD remains in this state until either the PMTU_RAISE_TIMER expires or a black hole is detected.

When DPLPMTUD uses an unacknowledged PL and is in the SEARCH_COMPLETE state, a CONFIRMATION_TIMER periodically resets the PROBE_COUNT and schedules a probe packet with the size of the PLPMTU. If MAX_PROBES successive PLPMTUD sized probes fail to be acknowledged the method enters the BASE state. When used with an acknowledged PL (e.g., SCTP), DPLPMTUD SHOULD NOT continue to generate PLPMTU probes in this state.

ERROR: The ERROR state represents the case where either the network path is not known to support a PLPMTU of at least the BASE_PLPMTU size or when there is contradictory information about the network path that would otherwise result in excessive variation in the MPS signaled to the higher layer. The state implements a method to mitigate oscillation in the state-event engine. It signals a conservative value of the MPS to the higher layer by the PL. The state is exited when packet probes no longer detect the error. The PL sender then enters the SEARCHING state.

Implementations are permitted to enable endpoint fragmentation if the DPLPMTUD is unable to validate MIN_PLPMTU within PROBE_COUNT probes. If DPLPMTUD is unable to validate MIN_PLPMTU the implementation will transition to the DISABLED state.

Note: MIN_PLPMTU could be identical to BASE_PLPMTU, simplifying the actions in this state.

5.3. Search to Increase the PLPMTU

This section describes the algorithms used by DPLPMTUD to search for a larger PLPMTU.

5.3.1. Probing for a larger PLPMTU

Implementations use a search algorithm across the search range to determine whether a larger PLPMTU can be supported across a network path.

The method discovers the search range by confirming the minimum PLPMTU and then using the probe method to select a PROBED_SIZE less than or equal to MAX_PLPMTU. MAX_PLPMTU is the minimum of the local MTU and EMTU_R (when this is learned from the remote endpoint). The MAX_PLPMTU MAY be reduced by an application that sets a maximum to the size of datagrams it will send.

The PROBE_COUNT is initialized to zero when the first probe with a size greater than or equal to PLPMTUD is sent. Each probe packet successfully sent to the remote peer is confirmed by acknowledgment at the PL, see Section 4.1.

Each time a probe packet is sent to the destination, the PROBE_TIMER is started. The timer is canceled when the PL receives acknowledgment that the probe packet has been successfully sent across the path Section 4.1. This confirms that the PROBED_SIZE is supported, and the PROBED_SIZE value is then assigned to the PLPMTU. The search algorithm can continue to send subsequent probe packets of an increasing size.

If the timer expires before a probe packet is acknowledged, the probe has failed to confirm the PROBED_SIZE. Each time the PROBE_TIMER expires, the PROBE_COUNT is incremented, the PROBE_TIMER is reinitialized, and a new probe of the same size or any other size (determined by the search algorithm) can be sent. The maximum number of consecutive failed probes is configured (MAX_PROBES). If the value of the PROBE_COUNT reaches MAX_PROBES, probing will stop, and the PL sender enters the SEARCH_COMPLETE state.

5.3.2. Selection of Probe Sizes

The search algorithm determines a minimum useful gain in PLPMTU. It would not be constructive for a PL sender to attempt to probe for all sizes. This would incur unnecessary load on the path. Implementations SHOULD select the set of probe packet sizes to maximize the gain in PLPMTU from each search step.

Implementations could optimize the search procedure by selecting step sizes from a table of common PMTU sizes. When selecting the appropriate next size to search, an implementer ought to also consider that there can be common sizes of MPS that applications seek to use, and their could be common sizes of MTU used within the network.

5.3.3. Resilience to Inconsistent Path Information

A decision to increase the PLPMTU needs to be resilient to the possibility that information learned about the network path is inconsistent. A path is inconsistent when, for example, probe packets are lost due to other reasons (i.e., not packet size) or due to frequent path changes. Frequent path changes could occur by unexpected "flapping" - where some packets from a flow pass along one path, but other packets follow a different path with different properties.

A PL sender is able to detect inconsistency from the sequence of PLPMTU probes that are acknowledged or the sequence of PTB messages that it receives. When inconsistent path information is detected, a PL sender could use an alternate search mode that clamps the offered

MPS to a smaller value for a period of time. This avoids unnecessary loss of packets.

5.4. Robustness to Inconsistent Paths

Some paths could be unable to sustain packets of the BASE_PLPMTU size. The Error State could be implemented to provide robustness to such paths. This allows fallback to a smaller than desired PLPMTU, rather than suffer connectivity failure. This could utilize methods such as endpoint IP fragmentation to enable the PL sender to communicate using packets smaller than the BASE_PLPMTU.

6. Specification of Protocol-Specific Methods

DPLPMTUD requires protocol-specific details to be specified for each PL that is used.

The first subsection provides guidance on how to implement the DPLPMTUD method as a part of an application using UDP or UDP-Lite. The guidance also applies to other datagram services that do not include a specific transport protocol (such as a tunnel encapsulation). The following subsections describe how DPLPMTUD can be implemented as a part of the transport service, allowing applications using the service to benefit from discovery of the PLPMTU without themselves needing to implement this method when using SCTP and QUIC.

6.1. Application support for DPLPMTUD with UDP or UDP-Lite

The current specifications of UDP [RFC0768] and UDP-Lite [RFC3828] do not define a method in the RFC-series that supports PLPMTUD. In particular, the UDP transport does not provide the transport features needed to implement datagram PLPMTUD.

The DPLPMTUD method can be implemented as a part of an application built directly or indirectly on UDP or UDP-Lite, but relies on higher-layer protocol features to implement the method [BCP145].

Some primitives used by DPLPMTUD might not be available via the Datagram API (e.g., the ability to access the PLPMTU from the IP layer cache, or interpret received PTB messages).

In addition, it is recommended that PMTU discovery is not performed by multiple protocol layers. An application SHOULD avoid using DPLPMTUD when the underlying transport system provides this capability. A common method for managing the PLPMTU has benefits, both in the ability to share state between different processes and opportunities to coordinate probing for different PL instances.

6.1.1. Application Request

An application needs an application-layer protocol mechanism (such as a message acknowledgment method) that solicits a response from a destination endpoint. The method SHOULD allow the sender to check the value returned in the response to provide additional protection from off-path insertion of data [BCP145]. Suitable methods include a parameter known only to the two endpoints, such as a session ID or initialized sequence number.

6.1.2. Application Response

An application needs an application-layer protocol mechanism to communicate the response from the destination endpoint. This response could indicate successful reception of the probe across the path, but could also indicate that some (or all packets) have failed to reach the destination.

6.1.3. Sending Application Probe Packets

A probe packet can carry an application data block, but the successful transmission of this data is at risk when used for probing. Some applications might prefer to use a probe packet that does not carry an application data block to avoid disruption to data transfer.

6.1.4. Initial Connectivity

An application that does not have other higher-layer information confirming connectivity with the remote peer SHOULD implement a connectivity mechanism using acknowledged probe packets before entering the BASE state.

6.1.5. Validating the Path

An application that does not have other higher-layer information confirming correct delivery of datagrams SHOULD implement the CONFIRMATION_TIMER to periodically send probe packets while in the SEARCH_COMPLETE state.

6.1.6. Handling of PTB Messages

An application that is able and wishes to receive PTB messages MUST perform ICMP validation as specified in Section 5.2 of [BCP145]. This requires that the application checks each received PTB message to validate that it was received in response to transmitted traffic and that the reported PL_PTB_SIZE is less than the current probed size (see Section 4.6.2). A validated PTB message MAY be used

as input to the DPLPMTUD algorithm, but MUST NOT be used directly to set the PLPMTU.

6.2. DPLPMTUD for SCTP

Section 10.2 of [RFC4821] specified a recommended PLPMTUD probing method for SCTP and Section 7.3 of [RFC4960] recommended an endpoint apply the techniques in RFC4821 on a per-destination-address basis. The specification for DPLPMTUD continues the practice of using the PL to discover the PMTU, but updates, RFC4960 with a recommendation to use the method specified in this document: The RECOMMENDED method for generating probes is to add a chunk consisting only of padding to an SCTP message. The PAD chunk defined in [RFC4820] SHOULD be attached to a minimum length HEARTBEAT (HB) chunk to build a probe packet. This enables probing without affecting the transfer of user messages and without being limited by congestion control or flow control. This is preferred to using DATA chunks (with padding as required) as path probes.

Section 6.9 of [RFC4960] describes dividing the user messages into data chunks sent by the PL when using SCTP. This notes that once an SCTP message has been sent, it cannot be re-segmented. [RFC4960] describes the method to retransmit data chunks when the MPS has reduced, and the use of IP fragmentation for this case. This is unchanged by this document.

6.2.1. SCTP/IPv4 and SCTP/IPv6

6.2.1.1. Initial Connectivity

The base protocol is specified in [RFC4960]. This provides an acknowledged PL. A sender can therefore enter the BASE state as soon as connectivity has been confirmed.

6.2.1.2. Sending SCTP Probe Packets

Probe packets consist of an SCTP common header followed by a HEARTBEAT chunk and a PAD chunk. The PAD chunk is used to control the length of the probe packet. The HEARTBEAT chunk is used to trigger the sending of a HEARTBEAT ACK chunk. The reception of the HEARTBEAT ACK chunk acknowledges reception of a successful probe. A successful probe updates the association and path counters, but an unsuccessful probe is discounted (assumed to be a result of choosing too large a PLPMTU).

The SCTP sender needs to be able to determine the total size of a probe packet. The HEARTBEAT chunk could carry a Heartbeat Information parameter that includes, besides the information

suggested in [RFC4960], the probe size to help an implementation associate a HEARTBEAT-ACK with the size of probe that was sent. The sender could also use other methods, such as sending a nonce and verifying the information returned also contains the corresponding nonce. The length of the PAD chunk is computed by reducing the probing size by the size of the SCTP common header and the HEARTBEAT chunk. The payload of the PAD chunk contains arbitrary data. When transmitted at the IP layer, the PMTU size also includes the IPv4 or IPv6 header(s).

Probing can start directly after the PL handshake, this can be done before data is sent. Assuming this behavior (i.e., the PMTU is smaller than or equal to the interface MTU), this process will take several round trip time periods, dependent on the number of DPLPMTUD probes sent. The Heartbeat timer can be used to implement the PROBE_TIMER.

6.2.1.3. Validating the Path with SCTP

Since SCTP provides an acknowledged PL, a sender MUST NOT implement the CONFIRMATION_TIMER while in the SEARCH_COMPLETE state.

6.2.1.4. PTB Message Handling by SCTP

Normal ICMP validation MUST be performed as specified in Appendix C of [RFC4960]. This requires that the first 8 bytes of the SCTP common header are quoted in the payload of the PTB message, which can be the case for ICMPv4 and is normally the case for ICMPv6.

When a PTB message has been validated, the PL_PTB_SIZE calculated from the PTB_SIZE reported in the PTB message SHOULD be used with the DPLPMTUD algorithm, providing that the reported PL_PTB_SIZE is less than the current probe size (see Section 4.6).

6.2.2. DPLPMTUD for SCTP/UDP

The UDP encapsulation of SCTP is specified in [RFC6951].

This specification updates the reference to RFC 4821 in section 5.6 of RFC 6951 to refer to XXXTHISRFCXXX. RFC 6951 is updated by addition of the following sentence at the end of section 5.6: "The RECOMMENDED method for determining the MTU of the path is specified in XXXTHISRFCXXX".

XXX RFC EDITOR - please replace XXXTHISRFCXXX when published XXX

6.2.2.1. Initial Connectivity

A sender can enter the BASE state as soon as SCTP connectivity has been confirmed.

6.2.2.2. Sending SCTP/UDP Probe Packets

Packet probing can be performed as specified in Section 6.2.1.2. The size of the probe packet includes the 8 bytes of UDP Header. This has to be considered when filling the probe packet with the PAD chunk.

6.2.2.3. Validating the Path with SCTP/UDP

SCTP provides an acknowledged PL, therefore a sender does not implement the CONFIRMATION_TIMER while in the SEARCH_COMPLETE state.

6.2.2.4. Handling of PTB Messages by SCTP/UDP

ICMP validation MUST be performed for PTB messages as specified in Appendix C of [RFC4960]. This requires that the first 8 bytes of the SCTP common header are contained in the PTB message, which can be the case for ICMPv4 (but note the UDP header also consumes a part of the quoted packet header) and is normally the case for ICMPv6. When the validation is completed, the PL_PTB_SIZE calculated from the PTB_SIZE in the PTB message SHOULD be used with the DPLPMTUD providing that the reported PL_PTB_SIZE is less than the current probe size.

6.2.3. DPLPMTUD for SCTP/DTLS

The Datagram Transport Layer Security (DTLS) encapsulation of SCTP is specified in [RFC8261]. This is used for data channels in WebRTC implementations. This specification updates the reference to RFC 4821 in section 5 of RFC 8261 to refer to XXXTHISRFCXXX.

XXX RFC EDITOR - please replace XXXTHISRFCXXX when published XXX

6.2.3.1. Initial Connectivity

A sender can enter the BASE state as soon as SCTP connectivity has been confirmed.

6.2.3.2. Sending SCTP/DTLS Probe Packets

Packet probing can be done, as specified in Section 6.2.1.2. The maximum payload is reduced by the size of the DTLS headers, which has to be considered when filling the PAD chunk. The size of the probe packet includes the DTLS PL headers. This has to be considered when filling the probe packet with the PAD chunk.

6.2.3.3. Validating the Path with SCTP/DTLS

Since SCTP provides an acknowledged PL, a sender MUST NOT implement the CONFIRMATION_TIMER while in the SEARCH_COMPLETE state.

6.2.3.4. Handling of PTB Messages by SCTP/DTLS

[RFC4960] does not specify a way to validate SCTP/DTLS ICMP message payload and neither does this document. This can prevent processing of PTB messages at the PL.

6.3. DPLPMTUD for QUIC

QUIC [I-D.ietf-quic-transport] is a UDP-based PL that provides reception feedback. The UDP payload includes a QUIC packet header, a protected payload, and any authentication fields. It supports padding and packet coalescence that can be used to construct probe packets. From the perspective of DPLPMTUD, QUIC can function as an acknowledged PL. [I-D.ietf-quic-transport] describes the method for using DPLPMTUD with QUIC packets.

7. Acknowledgments

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Thanks to all that have commented or contributed, the TSVWG and QUIC working groups, and Mathew Calder and Julius Flohr for providing early implementations.

8. IANA Considerations

This memo includes no request to IANA.

If there are no requirements for IANA, the section will be removed during conversion into an RFC by the RFC Editor.

9. Security Considerations

The security considerations for the use of UDP and SCTP are provided in the referenced RFCs.

To avoid excessive load, the interval between individual probe packets **MUST** be at least one RTT, and the interval between rounds of probing is determined by the `PMTU_RAISE_TIMER`.

A PL sender needs to ensure that the method used to confirm reception of probe packets protects from off-path attackers injecting packets into the path. This protection is provided in IETF-defined protocols (e.g., TCP, SCTP) using a randomly-initialized sequence number. A description of one way to do this when using UDP is provided in section 5.1 of [BCP145]).

There are cases where ICMP Packet Too Big (PTB) messages are not delivered due to policy, configuration or equipment design (see Section 1.1). This method therefore does not rely upon PTB messages being received, but is able to utilize these when they are received by the sender. PTB messages could potentially be used to cause a node to inappropriately reduce the PLPMTU. A node supporting DPLPMTUD **MUST** therefore appropriately validate the payload of PTB messages to ensure these are received in response to transmitted traffic (i.e., a reported error condition that corresponds to a datagram actually sent by the path layer, see Section 4.6.1).

An on-path attacker able to create a PTB message could forge PTB messages that include a valid quoted IP packet. Such an attack could be used to drive down the PLPMTU. An on-path device could similarly force a reduction of the PLPMTU by implementing a policy that drops packets larger than a configured size. There are two ways this method can be mitigated against such attacks: First, by ensuring that a PL sender never reduces the PLPMTU below the base size, solely in response to receiving a PTB message. This is achieved by first entering the BASE state when such a message is received. Second, the design does not require processing of PTB messages, a PL sender could therefore suspend processing of PTB messages (e.g., in a robustness mode after detecting that subsequent probes actually confirm that a size larger than the `PTB_SIZE` is supported by a path).

Parsing the quoted packet inside a PTB message can introduce additional per-packet processing at the PL sender. This processing **SHOULD** be limited to avoid a denial of service attack when arbitrary headers are included. Rate-limiting the processing could result in PTB messages not being received by a PL, however the DPLPMTUD method is robust to such loss.

The successful processing of an ICMP message can trigger a probe when the reported PTB size is valid, but this does not directly update the PLPMTU for the path. This prevents a message attempting to black hole data by indicating a size larger than supported by the path.

It is possible that the information about a path is not stable. This could be a result of forwarding across more than one path that has a different actual PMTU or a single path presents a varying PMTU. The design of a PLPMTUD implementation SHOULD consider how to mitigate the effects of varying path information. One possible mitigation is to provide robustness (see Section 5.4) in the method that avoids oscillation in the MPS.

DPLPMTUD methods can introduce padding data to inflate the length of the datagram to the total size required for a probe packet. The total size of a probe packet includes all headers and padding added to the payload data being sent (e.g., including security-related fields such as an AEAD tag and TLS record layer padding). The value of the padding data does not influence the DPLPMTUD search algorithm, and therefore needs to be set consistent with the policy of the PL.

If a PL can make use of cryptographic confidentiality or data-integrity mechanisms, then the design ought to avoid adding anything (e.g., padding) to DPLPMTUD probe packets that is not also protected by those cryptographic mechanisms.

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Appendix A. Revision Notes

Note to RFC-Editor: please remove this entire section prior to publication.

Individual draft -00:

- * Comments and corrections are welcome directly to the authors or via the IETF TSVWG working group mailing list.
- * This update is proposed for WG comments.

Individual draft -01:

- * Contains the first representation of the algorithm, showing the states and timers
- * This update is proposed for WG comments.

Individual draft -02:

- * Contains updated representation of the algorithm, and textual corrections.
- * The text describing when to set the effective PMTU has not yet been validated by the authors
- * To determine security to off-path-attacks: We need to decide whether a received PTB message SHOULD/MUST be validated? The text on how to handle a PTB message indicating a link MTU larger than the probe has yet not been validated by the authors
- * No text currently describes how to handle inconsistent results from arbitrary re-routing along different parallel paths

- * This update is proposed for WG comments.

Working Group draft -00:

- * This draft follows a successful adoption call for TSVWG
- * There is still work to complete, please comment on this draft.

Working Group draft -01:

- * This draft includes improved introduction.
- * The draft is updated to require ICMP validation prior to accepting PTB messages - this to be confirmed by WG
- * Section added to discuss Selection of Probe Size - methods to be evaluated and recommendations to be considered
- * Section added to align with work proposed in the QUIC WG.

Working Group draft -02:

- * The draft was updated based on feedback from the WG, and a detailed review by Magnus Westerlund.
- * The document updates RFC 4821.
- * Requirements list updated.
- * Added more explicit discussion of a simpler black-hole detection mode.
- * This draft includes reorganisation of the section on IETF protocols.
- * Added more discussion of implementation within an application.
- * Added text on flapping paths.
- * Replaced 'effective MTU' with new term PLPMTU.

Working Group draft -03:

- * Updated figures
- * Added more discussion on blackhole detection
- * Added figure describing just blackhole detection

- * Added figure relating MPS sizes

Working Group draft -04:

- * Described phases and named these consistently.
- * Corrected transition from confirmation directly to the search phase (Base has been checked).
- * Redrawn state diagrams.
- * Renamed BASE_MTU to BASE_PMTU (because it is a base for the PMTU).
- * Clarified Error state.
- * Clarified suspending DPLPMTUD.
- * Verified normative text in requirements section.
- * Removed duplicate text.
- * Changed all text to refer to /packet probe/probe packet/
/validation/verification/ added term /Probe Confirmation/ and
clarified BlackHole detection.

Working Group draft -05:

- * Updated security considerations.
- * Feedback after speaking with Joe Touch helped improve UDP-Options description.

Working Group draft -06:

- * Updated description of ICMP issues in section 1.1
- * Update to description of QUIC.

Working group draft -07:

- * Moved description of the PTB processing method from the PTB requirements section.
- * Clarified what is performed in the PTB validation check.
- * Updated security consideration to explain PTB security without needing to read the rest of the document.

- * Reformatted state machine diagram

Working group draft -08:

- * Moved to rfcxml v3+
- * Rendered diagrams to svg in html version.
- * Removed Appendix A. Event-driven state changes.
- * Removed section on DPLPMTUD with UDP Options.
- * Shortened the description of phases.

Working group draft -09:

- * Remove final mention of UDP Options
- * Add Initial Connectivity sections to each PL
- * Add to disable outgoing pmtu enforcement of packets

Working group draft -10:

- * Address comments from Lars Eggert
- * Reinforce that PROBE_COUNT is successive attempts to probe for any size
- * Redefine MAX_PROBES to 3
- * Address PTB_SIZE of 0 or less than MIN_PLPMTU

Working group draft -11:

- * Restore a sentence removed in previous rev
- * De-acronymise QUIC
- * Address some nits

Working group draft -12:

- * Add TSVWG, QUIC and implementers to acknowledgments
- * Shorten a diagram line.
- * Address nits from Julius and Wes.

- * Be clearer when talking about IP layer caches

Working group draft -13, -14:

- * Updated after WGLC.

Working group draft -15:

- * Updated after AD evaluation and prepared for IETF-LC.

Working group draft -16:

- * Updated text after SECDIR review.

Working group draft -17:

- * Updated text after GENART and IETF-LC.

- * Renamed BASE_MTU to BASE_PLPMTU, and MIN and MAX PMTU to PLPMTU (because these are about a base for the PLPMTU), and ensured consistent separation of PMTU and PLPMTU.

- * Adopted US-style English throughout.

Working group draft -18:

- * Updated text and address nits from OPSDIR, ART and IESG reviews.

- * Order PTB processing based on PL_PTB_SIZE

Working group draft -19:

- * Updated text and address nits based on comments from Tim Chown and Murray S. Kucherawy.

Working group draft -20:

- * Address nits and comments from IESG

- * Refer to BCP 145 rather than RFC 8085 in most places.

- * Update probing method text for SCTP and QUIC.

Working group draft -21:

- * Update QUIC text for skipping into BASE state.

Working group draft -22:

- * Add a section reference to MPS
- * Clarify MIN_PLPMTU text
- * Remove most QUIC text
- * Make QUIC reference informative.

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Guidelines for Adding Congestion Notification to Protocols that
Encapsulate IP
draft-ietf-tsvwg-ecn-encap-guidelines-16

Abstract

The purpose of this document is to guide the design of congestion notification in any lower layer or tunnelling protocol that encapsulates IP. The aim is for explicit congestion signals to propagate consistently from lower layer protocols into IP. Then the IP internetwork layer can act as a portability layer to carry congestion notification from non-IP-aware congested nodes up to the transport layer (L4). Following these guidelines should assure interworking among IP layer and lower layer congestion notification mechanisms, whether specified by the IETF or other standards bodies. This document updates the advice to subnetwork designers about ECN in RFC 3819.

Status of This Memo

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1. Introduction

The benefits of Explicit Congestion Notification (ECN) described in [RFC8087] and summarized below can only be fully realized if support for ECN is added to the relevant subnetwork technology, as well as to IP. When a lower layer buffer drops a packet obviously it does not just drop at that layer; the packet disappears from all layers. In contrast, when active queue management (AQM) at a lower layer marks a packet with ECN, the marking needs to be explicitly propagated up the layers. The same is true if AQM marks the outer header of a packet that encapsulates inner tunnelled headers. Forwarding ECN is not as straightforward as other headers because it has to be assumed ECN may be only partially deployed. If a lower layer header that contains ECN congestion indications is stripped off by a subnet egress that is not ECN-aware, or if the ultimate receiver or sender is not ECN-aware, congestion needs to be indicated by dropping a packet, not marking it.

The purpose of this document is to guide the addition of congestion notification to any subnet technology or tunnelling protocol, so that lower layer AQM algorithms can signal congestion explicitly and it will propagate consistently into encapsulated (higher layer) headers, otherwise the signals will not reach their ultimate destination.

ECN is defined in the IP header (v4 and v6) [RFC3168] to allow a resource to notify the onset of queue build-up without having to drop packets, by explicitly marking a proportion of packets with the congestion experienced (CE) codepoint.

Given a suitable marking scheme, ECN removes nearly all congestion loss and it cuts delays for two main reasons:

- o It avoids the delay when recovering from congestion losses, which particularly benefits small flows or real-time flows, making their delivery time predictably short [RFC2884];
- o As ECN is used more widely by end-systems, it will gradually remove the need to configure a degree of delay into buffers before they start to notify congestion (the cause of bufferbloat). This is because drop involves a trade-off between sending a timely signal and trying to avoid impairment, whereas ECN is solely a signal not an impairment, so there is no harm triggering it earlier.

Some lower layer technologies (e.g. MPLS, Ethernet) are used to form subnetworks with IP-aware nodes only at the edges. These networks are often sized so that it is rare for interior queues to overflow. However, until recently this was more due to the inability of TCP to

saturate the links. For many years, fixes such as window scaling [RFC7323] proved hard to deploy. And the Reno variant of TCP has remained in widespread use despite its inability to scale to high flow rates. However, now that modern operating systems are finally capable of saturating interior links, even the buffers of well-provisioned interior switches will need to signal episodes of queuing.

Propagation of ECN is defined for MPLS [RFC5129], and is being defined for TRILL [RFC7780], [I-D.ietf-trill-ecn-support], but it remains to be defined for a number of other subnetwork technologies.

Similarly, ECN propagation is yet to be defined for many tunnelling protocols. [RFC6040] defines how ECN should be propagated for IP-in-IPv4 [RFC2003], IP-in-IPv6 [RFC2473] and IPsec [RFC4301] tunnels, but there are numerous other tunnelling protocols with a shim and/or a layer 2 header between two IP headers (v4 or v6). Some address ECN propagation between the IP headers, but many do not. This document gives guidance on how to address ECN propagation for future tunnelling protocols, and a companion standards track specification [I-D.ietf-tsvwg-rfc6040update-shim] updates those existing IP-shim-(L2)-IP protocols that are under IETF change control and still widely used.

Incremental deployment is the most delicate aspect when adding support for ECN. The original ECN protocol in IP [RFC3168] was carefully designed so that a congested buffer would not mark a packet (rather than drop it) unless both source and destination hosts were ECN-capable. Otherwise its congestion markings would never be detected and congestion would just build up further. However, to support congestion marking below the IP layer or within tunnels, it is not sufficient to only check that the two layer 4 transport endpoints support ECN; correct operation also depends on the decapsulator at each subnet or tunnel egress faithfully propagating congestion notifications to the higher layer. Otherwise, a legacy decapsulator might silently fail to propagate any ECN signals from the outer to the forwarded header. Then the lost signals would never be detected and again congestion would build up further. The guidelines given later require protocol designers to carefully consider incremental deployment, and suggest various safe approaches for different circumstances.

Of course, the IETF does not have standards authority over every link layer protocol. So this document gives guidelines for designing propagation of congestion notification across the interface between IP and protocols that may encapsulate IP (i.e. that can be layered beneath IP). Each lower layer technology will exhibit different issues and compromises, so the IETF or the relevant standards body

must be free to define the specifics of each lower layer congestion notification scheme. Nonetheless, if the guidelines are followed, congestion notification should interwork between different technologies, using IP in its role as a 'portability layer'.

Therefore, the capitalized terms 'SHOULD' or 'SHOULD NOT' are often used in preference to 'MUST' or 'MUST NOT', because it is difficult to know the compromises that will be necessary in each protocol design. If a particular protocol design chooses not to follow a 'SHOULD (NOT)' given in the advice below, it MUST include a sound justification.

It has not been possible to give common guidelines for all lower layer technologies, because they do not all fit a common pattern. Instead they have been divided into a few distinct modes of operation: feed-forward-and-upward; feed-upward-and-forward; feed-backward; and null mode. These modes are described in Section 3, then in the subsequent sections separate guidelines are given for each mode.

1.1. Update to RFC 3819

This document updates the brief advice to subnetwork designers about ECN in [RFC3819], by replacing the last two paragraphs of Section 13 with the following sentence:

By following the guidelines in [this document], subnetwork designers can enable a layer-2 protocol to participate in congestion control without dropping packets via propagation of explicit congestion notification (ECN [RFC3168]) to receivers.

and adding [this document] as an informative reference. {RFC Editor: Please replace both instances of [this document] above with the number of the present RFC when published.}

1.2. Scope

This document only concerns wire protocol processing of explicit notification of congestion. It makes no changes or recommendations concerning algorithms for congestion marking or for congestion response, because algorithm issues should be independent of the layer the algorithm operates in.

The default ECN semantics are described in [RFC3168] and updated by [RFC8311]. Also the guidelines for AQM designers [RFC7567] clarify the semantics of both drop and ECN signals from AQM algorithms. [RFC4774] is the appropriate best current practice specification of how algorithms with alternative semantics for the ECN field can be

partitioned from Internet traffic that uses the default ECN semantics. There are two main examples for how alternative ECN semantics have been defined in practice:

- o RFC 4774 suggests using the ECN field in combination with a Diffserv codepoint such as in PCN [RFC6660], Voice over 3G [UTRAN] or Voice over LTE (VoLTE) [LTE-RA];
- o RFC 8311 suggests using the ECT(1) codepoint of the ECN field to indicate alternative semantics such as for the experimental Low Latency Low Loss Scalable throughput (L4S) service [I-D.ietf-tsvwg-ecn-l4s-id]).

The aim is that the default rules for encapsulating and decapsulating the ECN field are sufficiently generic that tunnels and subnets will encapsulate and decapsulate packets without regard to how algorithms elsewhere are setting or interpreting the semantics of the ECN field. [RFC6040] updates RFC 4774 to allow alternative encapsulation and decapsulation behaviours to be defined for alternative ECN semantics. However it reinforces the same point - that it is far preferable to try to fit within the common ECN encapsulation and decapsulation behaviours, because expecting all lower layer technologies and tunnels to be updated is likely to be completely impractical.

Alternative semantics for the ECN field can be defined to depend on the traffic class indicated by the DSCP. Therefore correct propagation of congestion signals could depend on correct propagation of the DSCP between the layers and along the path. For instance, if the meaning of the ECN field depends on the DSCP (as in PCN or VoLTE) and if the outer DSCP is stripped on decapsulation, as in the pipe model of [RFC2983], the special semantics of the ECN field would be lost. Similarly, if the DSCP is changed at the boundary between Diffserv domains, the special ECN semantics would also be lost. This is an important implication of the localized scope of most Diffserv arrangements. In this document, correct propagation of traffic class information is assumed, while what 'correct' means and how it is achieved is covered elsewhere (e.g. RFC 2983) and is outside the scope of the present document.

The guidelines in this document do ensure that common encapsulation and decapsulation rules are sufficiently generic to cover cases where ECT(1) is used instead of ECT(0) to identify alternative ECN semantics (as in L4S [I-D.ietf-tsvwg-ecn-l4s-id]) and where ECN marking algorithms use ECT(1) to encode 3 severity levels into the ECN field (e.g. PCN [RFC6660]) rather than the default of 2. All these different semantics for the ECN field work because it has been possible to define common default decapsulation rules that allow for all cases.

Note that the guidelines in this document do not necessarily require the subnet wire protocol to be changed to add support for congestion notification. For instance, the Feed-Up-and-Forward Mode (Section 3.2) and the Null Mode (Section 3.4) do not. Another way to add congestion notification without consuming header space in the subnet protocol might be to use a parallel control plane protocol.

This document focuses on the congestion notification interface between IP and lower layer or tunnel protocols that can encapsulate IP, where the term 'IP' includes v4 or v6, unicast, multicast or anycast. However, it is likely that the guidelines will also be useful when a lower layer protocol or tunnel encapsulates itself, e.g. Ethernet MAC in MAC ([IEEE802.1Q]; previously 802.1ah) or when it encapsulates other protocols. In the feed-backward mode, propagation of congestion signals for multicast and anycast packets is out-of-scope (because the complexity would make it unlikely to be attempted).

2. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

Further terminology used within this document:

Protocol data unit (PDU): Information that is delivered as a unit among peer entities of a layered network consisting of protocol control information (typically a header) and possibly user data (payload) of that layer. The scope of this document includes layer 2 and layer 3 networks, where the PDU is respectively termed a frame or a packet (or a cell in ATM). PDU is a general term for any of these. This definition also includes a payload with a shim header lying somewhere between layer 2 and 3.

Transport: The end-to-end transmission control function, conventionally considered at layer-4 in the OSI reference model. Given the audience for this document will often use the word transport to mean low level bit carriage, whenever the term is used it will be qualified, e.g. 'L4 transport'.

Encapsulator: The link or tunnel endpoint function that adds an outer header to a PDU (also termed the 'link ingress', the 'subnet ingress', the 'ingress tunnel endpoint' or just the 'ingress' where the context is clear).

Decapsulator: The link or tunnel endpoint function that removes an outer header from a PDU (also termed the 'link egress', the 'subnet egress', the 'egress tunnel endpoint' or just the 'egress' where the context is clear).

Incoming header: The header of an arriving PDU before encapsulation.

Outer header: The header added to encapsulate a PDU.

Inner header: The header encapsulated by the outer header.

Outgoing header: The header forwarded by the decapsulator.

CE: Congestion Experienced [RFC3168]

ECT: ECN-Capable (L4) Transport [RFC3168]

Not-ECT: Not ECN-Capable (L4) Transport [RFC3168]

Load Regulator: For each flow of PDUs, the transport function that is capable of controlling the data rate. Typically located at the data source, but in-path nodes can regulate load in some congestion control arrangements (e.g. admission control, policing nodes or transport circuit-breakers [RFC8084]). Note the term "a function capable of controlling the load" deliberately includes a transport that does not actually control the load responsively but ideally it ought to (e.g. a sending application without congestion control that uses UDP).

ECN-PDU: A PDU at the IP layer or below with a capacity to signal congestion that is part of a congestion control feedback loop within which all the nodes necessary to propagate the signal back to the Load Regulator are capable of doing that propagation. An IP packet with a non-zero ECN field implies that the endpoints are ECN-capable, so this would be an ECN-PDU. However, ECN-PDU is intended to be a general term for a PDU at lower layers, as well as at the IP layer.

Not-ECN-PDU: A PDU at the IP layer or below that is part of a congestion control feedback-loop within which at least one node necessary to propagate any explicit congestion notification signals back to the Load Regulator is not capable of doing that propagation.

3. Modes of Operation

This section sets down the different modes by which congestion information is passed between the lower layer and the higher one. It acts as a reference framework for the following sections, which give normative guidelines for designers of explicit congestion notification protocols, taking each mode in turn:

Feed-Forward-and-Up: Nodes feed forward congestion notification towards the egress within the lower layer then up and along the layers towards the end-to-end destination at the transport layer. The following local optimisation is possible:

Feed-Up-and-Forward: A lower layer switch feeds-up congestion notification directly into the higher layer (e.g. into the ECN field in the IP header), irrespective of whether the node is at the egress of a subnet.

Feed-Backward: Nodes feed back congestion signals towards the ingress of the lower layer and (optionally) attempt to control congestion within their own layer.

Null: Nodes cannot experience congestion at the lower layer except at ingress nodes (which are IP-aware or equivalently higher-layer-aware).

3.1. Feed-Forward-and-Up Mode

Like IP and MPLS, many subnet technologies are based on self-contained protocol data units (PDUs) or frames sent unreliably. They provide no feedback channel at the subnetwork layer, instead relying on higher layers (e.g. TCP) to feed back loss signals.

In these cases, ECN may best be supported by standardising explicit notification of congestion into the lower layer protocol that carries the data forwards. Then a specification is needed for how the egress of the lower layer subnet propagates this explicit signal into the forwarded upper layer (IP) header. This signal continues forwards until it finally reaches the destination transport (at L4). Then typically the destination will feed this congestion notification back to the source transport using an end-to-end protocol (e.g. TCP). This is the arrangement that has already been used to add ECN to IP-in-IP tunnels [RFC6040], IP-in-MPLS and MPLS-in-MPLS [RFC5129].

This mode is illustrated in Figure 1. Along the middle of the figure, layers 2, 3 and 4 of the protocol stack are shown, and one packet is shown along the bottom as it progresses across the network from source to destination, crossing two subnets connected by a

router, and crossing two switches on the path across each subnet. Congestion at the output of the first switch (shown as *) leads to a congestion marking in the L2 header (shown as C in the illustration of the packet). The chevrons show the progress of the resulting congestion indication. It is propagated from link to link across the subnet in the L2 header, then when the router removes the marked L2 header, it propagates the marking up into the L3 (IP) header. The router forwards the marked L3 header into subnet 2, and when it adds a new L2 header it copies the L3 marking into the L2 header as well, as shown by the 'C's in both layers (assuming the technology of subnet 2 also supports explicit congestion marking).

Note that there is no implication that each 'C' marking is encoded the same; a different encoding might be used for the 'C' marking in each protocol.

Finally, for completeness, we show the L3 marking arriving at the destination, where the host transport protocol (e.g. TCP) feeds it back to the source in the L4 acknowledgement (the 'C' at L4 in the packet at the top of the diagram).

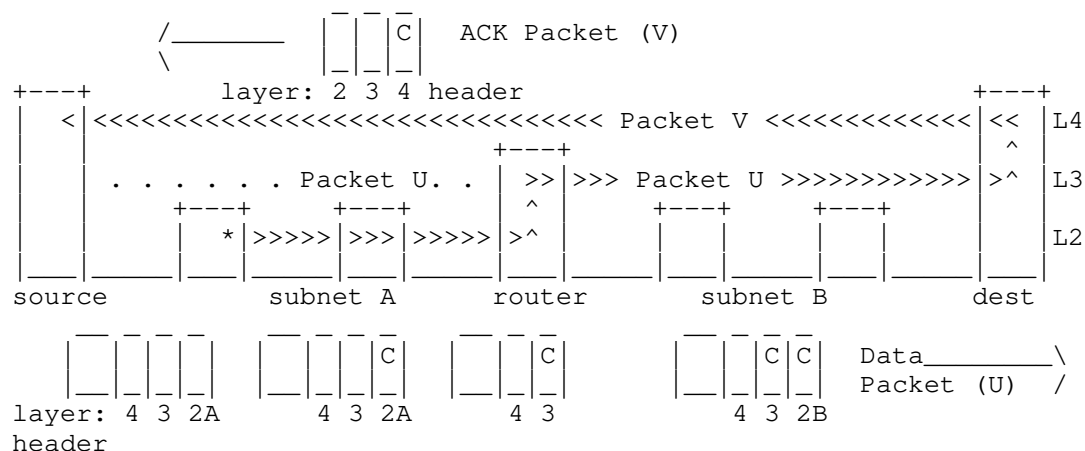


Figure 1: Feed-Forward-and-Up Mode

Of course, modern networks are rarely as simple as this text-book example, often involving multiple nested layers. For example, a 3GPP mobile network may have two IP-in-IP (GTP [GTPv1]) tunnels in series and an MPLS backhaul between the base station and the first router. Nonetheless, the example illustrates the general idea of feeding congestion notification forward then upward whenever a header is removed at the egress of a subnet.

Note that the FECN (forward ECN) bit in Frame Relay [Buck00] and the explicit forward congestion indication (EFCI [ITU-T.I.371]) bit in ATM user data cells follow a feed-forward pattern. However, in ATM, this arrangement is only part of a feed-forward-and-backward pattern at the lower layer, not feed-forward-and-up out of the lower layer--the intention was never to interface to IP ECN at the subnet egress. To our knowledge, Frame Relay FECN is solely used to detect where more capacity should be provisioned.

3.2. Feed-Up-and-Forward Mode

Ethernet is particularly difficult to extend incrementally to support explicit congestion notification. One way to support ECN in such cases has been to use so called 'layer-3 switches'. These are Ethernet switches that dig into the Ethernet payload to find an IP header and manipulate or act on certain IP fields (specifically Diffserv & ECN). For instance, in Data Center TCP [RFC8257], layer-3 switches are configured to mark the ECN field of the IP header within the Ethernet payload when their output buffer becomes congested. With respect to switching, a layer-3 switch acts solely on the addresses in the Ethernet header; it does not use IP addresses, and it does not decrement the TTL field in the IP header.

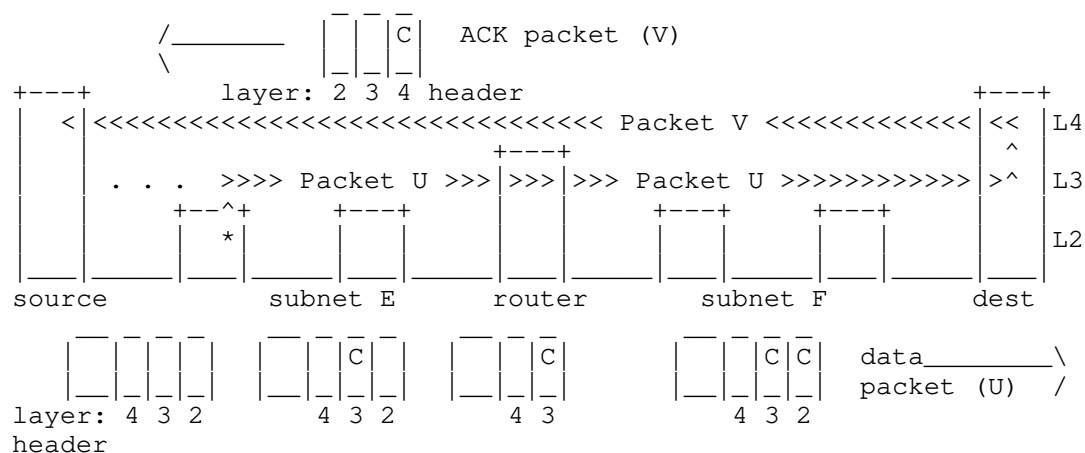


Figure 2: Feed-Up-and-Forward Mode

By comparing Figure 2 with Figure 1, it can be seen that subnet E (perhaps a subnet of layer-3 Ethernet switches) works in feed-up-and-forward mode by notifying congestion directly into L3 at the point of congestion, even though the congested switch does not otherwise act at L3. In this example, the technology in subnet F (e.g. MPLS) does

support ECN natively, so when the router adds the layer-2 header it copies the ECN marking from L3 to L2 as well.

3.3. Feed-Backward Mode

In some layer 2 technologies, explicit congestion notification has been defined for use internally within the subnet with its own feedback and load regulation, but typically the interface with IP for ECN has not been defined.

For instance, for the available bit-rate (ABR) service in ATM, the relative rate mechanism was one of the more popular mechanisms for managing traffic, tending to supersede earlier designs. In this approach ATM switches send special resource management (RM) cells in both the forward and backward directions to control the ingress rate of user data into a virtual circuit. If a switch buffer is approaching congestion or is congested it sends an RM cell back towards the ingress with respectively the No Increase (NI) or Congestion Indication (CI) bit set in its message type field [ATM-TM-ABR]. The ingress then holds or decreases its sending bit-rate accordingly.

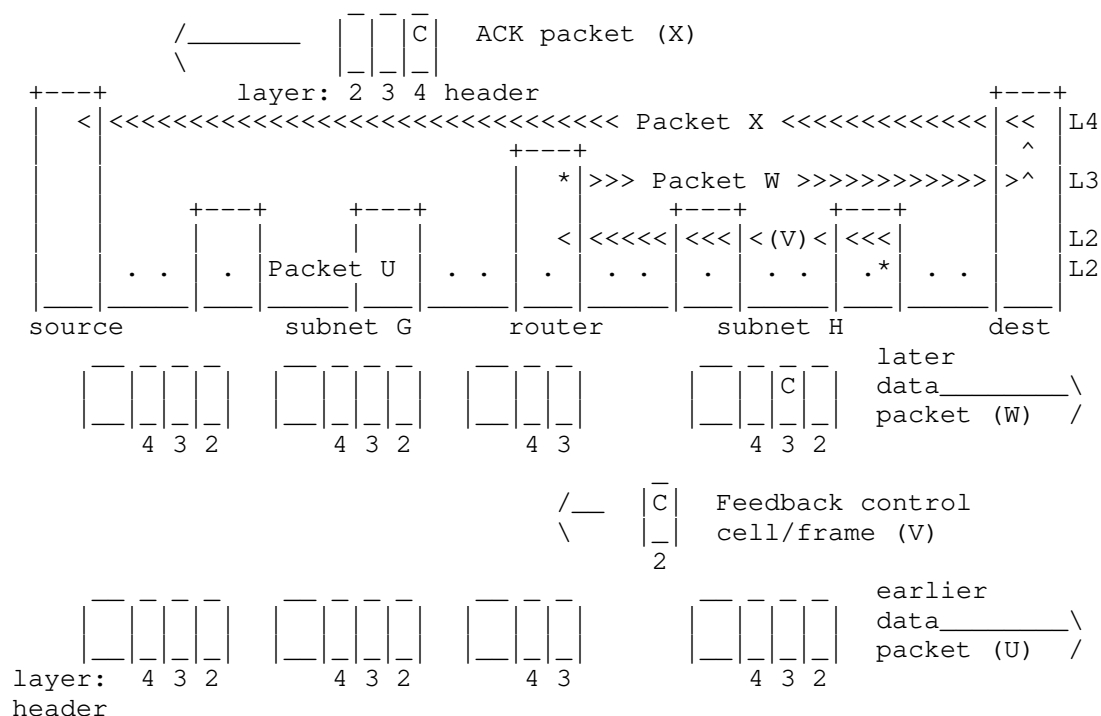


Figure 3: Feed-Backward Mode

ATM's feed-backward approach does not fit well when layered beneath IP's feed-forward approach--unless the initial data source is the same node as the ATM ingress. Figure 3 shows the feed-backward approach being used in subnet H. If the final switch on the path is congested (*), it does not feed-forward any congestion indications on packet (U). Instead it sends a control cell (V) back to the router at the ATM ingress.

However, the backward feedback does not reach the original data source directly because IP does not support backward feedback (and subnet G is independent of subnet H). Instead, the router in the middle throttles down its sending rate but the original data sources don't reduce their rates. The resulting rate mismatch causes the middle router's buffer at layer 3 to back up until it becomes congested, which it signals forwards on later data packets at layer 3 (e.g. packet W). Note that the forward signal from the middle router is not triggered directly by the backward signal. Rather, it is triggered by congestion resulting from the middle router's mismatched rate response to the backward signal.

In response to this later forward signalling, end-to-end feedback at layer-4 finally completes the tortuous path of congestion indications back to the origin data source, as before.

Quantized congestion notification (QCN [IEEE802.1Q]) would suffer from similar problems if extended to multiple subnets. However, from the start QCN was clearly characterized as solely applicable to a single subnet (see Section 6).

3.4. Null Mode

Often link and physical layer resources are 'non-blocking' by design. In these cases congestion notification may be implemented but it does not need to be deployed at the lower layer; ECN in IP would be sufficient.

A degenerate example is a point-to-point Ethernet link. Excess loading of the link merely causes the queue from the higher layer to back up, while the lower layer remains immune to congestion. Even a whole meshed subnetwork can be made immune to interior congestion by limiting ingress capacity and sufficient sizing of interior links, e.g. a non-blocking fat-tree network [Leiserson85]. An alternative to fat links near the root is numerous thin links with multi-path routing to ensure even worst-case patterns of load cannot congest any link, e.g. a Clos network [Clos53].

4. Feed-Forward-and-Up Mode: Guidelines for Adding Congestion Notification

Feed-forward-and-up is the mode already used for signalling ECN up the layers through MPLS into IP [RFC5129] and through IP-in-IP tunnels [RFC6040], whether encapsulating with IPv4 [RFC2003], IPv6 [RFC2473] or IPsec [RFC4301]. These RFCs take a consistent approach and the following guidelines are designed to ensure this consistency continues as ECN support is added to other protocols that encapsulate IP. The guidelines are also designed to ensure compliance with the more general best current practice for the design of alternate ECN schemes given in [RFC4774] and extended by [RFC8311].

The rest of this section is structured as follows:

- o Section 4.1 addresses the most straightforward cases, where [RFC6040] can be applied directly to add ECN to tunnels that are effectively IP-in-IP tunnels, but with shim header(s) between the IP headers.
- o The subsequent sections give guidelines for adding ECN to a subnet technology that uses feed-forward-and-up mode like IP, but it is

not so similar to IP that [RFC6040] rules can be applied directly. Specifically:

- * Sections 4.2, 4.3 and 4.4 respectively address how to add ECN support to the wire protocol and to the encapsulators and decapsulators at the ingress and egress of the subnet.
- * Section 4.5 deals with the special, but common, case of sequences of tunnels or subnets that all use the same technology
- * Section 4.6 deals with the question of reframing when IP packets do not map 1:1 into lower layer frames.

4.1. IP-in-IP Tunnels with Shim Headers

A common pattern for many tunnelling protocols is to encapsulate an inner IP header with shim header(s) then an outer IP header. A shim header is defined as one that is not sufficient alone to forward the packet as an outer header. Another common pattern is for a shim to encapsulate a layer 2 (L2) header, which in turn encapsulates (or might encapsulate) an IP header. [I-D.ietf-tsvwg-rfc6040update-shim] clarifies that RFC 6040 is just as applicable when there are shim(s) and possibly a L2 header between two IP headers.

However, it is not always feasible or necessary to propagate ECN between IP headers when separated by a shim. For instance, it might be too costly to dig to arbitrary depths to find an inner IP header, there may be little or no congestion within the tunnel by design (see null mode in Section 3.4 above), or a legacy implementation might not support ECN. In cases where a tunnel does not support ECN, it is important that the ingress does not copy the ECN field from an inner IP header to an outer. Therefore section 4 of [I-D.ietf-tsvwg-rfc6040update-shim] requires network operators to configure the ingress of a tunnel that does not support ECN so that it zeros the ECN field in the outer IP header.

Nonetheless, in many cases it is feasible to propagate the ECN field between IP headers separated by shim header(s) and/or a L2 header. Particularly in the typical case when the outer IP header and the shim(s) are added (or removed) as part of the same procedure. Even if the shim(s) encapsulate a L2 header, it is often possible to find an inner IP header within the L2 PDU and propagate ECN between that and the outer IP header. This can be thought of as a special case of the feed-up-and-forward mode (Section 3.2), so the guidelines for this mode apply (Section 5).

Numerous shim protocols have been defined for IP tunnelling. More recent ones e.g. Geneve [RFC8926] and Generic UDP Encapsulation (GUE) [I-D.ietf-intarea-gue] cite and follow RFC 6040. And some earlier ones, e.g. CAPWAP [RFC5415] and LISP [RFC6830], cite RFC 3168, which is compatible with RFC 6040.

However, as Section 9.3 of RFC 3168 pointed out, ECN support needs to be defined for many earlier shim-based tunnelling protocols, e.g. L2TPv2 [RFC2661], L2TPv3 [RFC3931], GRE [RFC2784], PPTP [RFC2637], GTP [GTPv1], [GTPv1-U], [GTPv2-C] and Teredo [RFC4380] as well as some recent ones, e.g. VXLAN [RFC7348], NVGRE [RFC7637] and NSH [RFC8300].

All these IP-based encapsulations can be updated in one shot by simple reference to RFC 6040. However, it would not be appropriate to update all these protocols from within the present guidance document. Instead a companion specification [I-D.ietf-tsvwg-rfc6040update-shim] has been prepared that has the appropriate standards track status to update standards track protocols. For those that are not under IETF change control [I-D.ietf-tsvwg-rfc6040update-shim] can only recommend that the relevant body updates them.

4.2. Wire Protocol Design: Indication of ECN Support

This section is intended to guide the redesign of any lower layer protocol that encapsulate IP to add native ECN support at the lower layer. It reflects the approaches used in [RFC6040] and in [RFC5129]. Therefore IP-in-IP tunnels or IP-in-MPLS or MPLS-in-MPLS encapsulations that already comply with [RFC6040] or [RFC5129] will already satisfy this guidance.

A lower layer (or subnet) congestion notification system:

1. SHOULD NOT apply explicit congestion notifications to PDUs that are destined for legacy layer-4 transport implementations that will not understand ECN, and
2. SHOULD NOT apply explicit congestion notifications to PDUs if the egress of the subnet might not propagate congestion notifications onward into the higher layer.

We use the term ECN-PDUs for a PDU on a feedback loop that will propagate congestion notification properly because it meets both the above criteria. And a Not-ECN-PDU is a PDU on a feedback loop that does not meet at least one of the criteria, and will therefore not propagate congestion notification properly. A

corollary of the above is that a lower layer congestion notification protocol:

3. SHOULD be able to distinguish ECN-PDUs from Not-ECN-PDUs.

Note that there is no need for all interior nodes within a subnet to be able to mark congestion explicitly. A mix of ECN and drop signals from different nodes is fine. However, if any interior nodes might generate ECN markings, guideline 2 above says that all relevant egress node(s) SHOULD be able to propagate those markings up to the higher layer.

In IP, if the ECN field in each PDU is cleared to the Not-ECT (not ECN-capable transport) codepoint, it indicates that the L4 transport will not understand congestion markings. A congested buffer must not mark these Not-ECT PDUs, and therefore drops them instead.

The mechanism a lower layer uses to distinguish the ECN-capability of PDUs need not mimic that of IP. The above guidelines merely say that the lower layer system, as a whole, should achieve the same outcome. For instance, ECN-capable feedback loops might use PDUs that are identified by a particular set of labels or tags. Alternatively, logical link protocols that use flow state might determine whether a PDU can be congestion marked by checking for ECN-support in the flow state. Other protocols might depend on out-of-band control signals.

The per-domain checking of ECN support in MPLS [RFC5129] is a good example of a way to avoid sending congestion markings to L4 transports that will not understand them, without using any header space in the subnet protocol.

In MPLS, header space is extremely limited, therefore RFC5129 does not provide a field in the MPLS header to indicate whether the PDU is an ECN-PDU or a Not-ECN-PDU. Instead, interior nodes in a domain are allowed to set explicit congestion indications without checking whether the PDU is destined for a L4 transport that will understand them. Nonetheless, this is made safe by requiring that the network operator upgrades all decapsulating edges of a whole domain at once, as soon as even one switch within the domain is configured to mark rather than drop during congestion. Therefore, any edge node that might decapsulate a packet will be capable of checking whether the higher layer transport is ECN-capable. When decapsulating a CE-marked packet, if the decapsulator discovers that the higher layer (inner header) indicates the transport is not ECN-capable, it drops the packet--effectively on behalf of the earlier congested node (see Decapsulation Guideline 1 in Section 4.4).

It was only appropriate to define such an incremental deployment strategy because MPLS is targeted solely at professional operators, who can be expected to ensure that a whole subnetwork is consistently configured. This strategy might not be appropriate for other link technologies targeted at zero-configuration deployment or deployment by the general public (e.g. Ethernet). For such 'plug-and-play' environments it will be necessary to invent a failsafe approach that ensures congestion markings will never fall into black holes, no matter how inconsistently a system is put together. Alternatively, congestion notification relying on correct system configuration could be confined to flavours of Ethernet intended only for professional network operators, such as Provider Backbone Bridges (PBB [IEEE802.1Q]; previously 802.1ah).

ECN support in TRILL [I-D.ietf-trill-ecn-support] provides a good example of how to add ECN to a lower layer protocol without relying on careful and consistent operator configuration. TRILL provides an extension header word with space for flags of different categories depending on whether logic to understand the extension is critical. The congestion experienced marking has been defined as a 'critical ingress-to-egress' flag. So if a transit RBridge sets this flag and an egress RBridge does not have any logic to process it, it will drop it; which is the desired default action anyway. Therefore TRILL RBridges can be updated with support for ECN in no particular order and, at the egress of the TRILL campus, congestion notification will be propagated to IP as ECN whenever ECN logic has been implemented, or as drop otherwise.

QCN [IEEE802.1Q] is not intended to extend beyond a single subnet, or to interoperate with ECN. Nonetheless, the way QCN indicates to lower layer devices that the end-points will not understand QCN provides another example that a lower layer protocol designer might be able to mimic for their scenario. An operator can define certain Priority Code Points (PCPs [IEEE802.1Q]; previously 802.1p) to indicate non-QCN frames and an ingress bridge is required to map arriving not-QCN-capable IP packets to one of these non-QCN PCPs.

4.3. Encapsulation Guidelines

This section is intended to guide the redesign of any node that encapsulates IP with a lower layer header when adding native ECN support to the lower layer protocol. It reflects the approaches used in [RFC6040] and in [RFC5129]. Therefore IP-in-IP tunnels or IP-in-MPLS or MPLS-in-MPLS encapsulations that already comply with [RFC6040] or [RFC5129] will already satisfy this guidance.

1. Egress Capability Check: A subnet ingress needs to be sure that the corresponding egress of a subnet will propagate any

congestion notification added to the outer header across the subnet. This is necessary in addition to checking that an incoming PDU indicates an ECN-capable (L4) transport. Examples of how this guarantee might be provided include:

- * by configuration (e.g. if any label switches in a domain support ECN marking, [RFC5129] requires all egress nodes to have been configured to propagate ECN)
 - * by the ingress explicitly checking that the egress propagates ECN (e.g. an early attempt to add ECN support to TRILL used IS-IS to check path capabilities before adding ECN extension flags to each frame [RFC7780]).
 - * by inherent design of the protocol (e.g. by encoding ECN marking on the outer header in such a way that a legacy egress that does not understand ECN will consider the PDU corrupt or invalid and discard it, thus at least propagating a form of congestion signal).
2. Egress Fails Capability Check: If the ingress cannot guarantee that the egress will propagate congestion notification, the ingress SHOULD disable ECN at the lower layer when it forwards the PDU. An example of how the ingress might disable ECN at the lower layer would be by setting the outer header of the PDU to identify it as a Not-ECN-PDU, assuming the subnet technology supports such a concept.
 3. Standard Congestion Monitoring Baseline: Once the ingress to a subnet has established that the egress will correctly propagate ECN, on encapsulation it SHOULD encode the same level of congestion in outer headers as is arriving in incoming headers. For example it might copy any incoming congestion notification into the outer header of the lower layer protocol.

This ensures that bulk congestion monitoring of outer headers (e.g. by a network management node monitoring ECN in passing frames) will measure congestion accumulated along the whole upstream path - since the Load Regulator not just since the ingress of the subnet. A node that is not the Load Regulator SHOULD NOT re-initialize the level of CE markings in the outer to zero.

It would still also be possible to measure congestion introduced across one subnet (or tunnel) by subtracting the level of CE markings on inner headers from that on outer headers (see Appendix C of [RFC6040]). For example:

- * If this guideline has been followed and if the level of CE markings is 0.4% on the outer and 0.1% on the inner, 0.4% congestion has been introduced across all the networks since the load regulator, and 0.3% ($= 0.4\% - 0.1\%$) has been introduced since the ingress to the current subnet (or tunnel);
- * Without this guideline, if the subnet ingress had re-initialized the outer congestion level to zero, the outer and inner would measure 0.1% and 0.3%. It would still be possible to infer that the congestion introduced since the Load Regulator was 0.4% ($= 0.1\% + 0.3\%$). But only if the monitoring system somehow knows whether the subnet ingress re-initialized the congestion level.

As long as subnet and tunnel technologies use the standard congestion monitoring baseline in this guideline, monitoring systems will know to use the former approach, rather than having to "somehow know" which approach to use.

4.4. Decapsulation Guidelines

This section is intended to guide the redesign of any node that decapsulates IP from within a lower layer header when adding native ECN support to the lower layer protocol. It reflects the approaches used in [RFC6040] and in [RFC5129]. Therefore IP-in-IP tunnels or IP-in-MPLS or MPLS-in-MPLS encapsulations that already comply with [RFC6040] or [RFC5129] will already satisfy this guidance.

A subnet egress SHOULD NOT simply copy congestion notification from outer headers to the forwarded header. It SHOULD calculate the outgoing congestion notification field from the inner and outer headers using the following guidelines. If there is any conflict, rules earlier in the list take precedence over rules later in the list:

1. If the arriving inner header is a Not-ECN-PDU it implies the L4 transport will not understand explicit congestion markings.
Then:
 - * If the outer header carries an explicit congestion marking, drop is the only indication of congestion that the L4 transport will understand. If the congestion marking is the most severe possible, the packet MUST be dropped. However, if congestion can be marked with multiple levels of severity and the packet's marking is not the most severe, this requirement can be relaxed to: the packet SHOULD be dropped.

- * If the outer is an ECN-PDU that carries no indication of congestion or a Not-ECN-PDU the PDU SHOULD be forwarded, but still as a Not-ECN-PDU.
- 2. If the outer header does not support explicit congestion notification (a Not-ECN-PDU), but the inner header does (an ECN-PDU), the inner header SHOULD be forwarded unchanged.
- 3. In some lower layer protocols congestion may be signalled as a numerical level, such as in the control frames of quantized congestion notification (QCN [IEEE802.1Q]). If such a multi-bit encoding encapsulates an ECN-capable IP data packet, a function will be needed to convert the quantized congestion level into the frequency of congestion markings in outgoing IP packets.
- 4. Congestion indications might be encoded by a severity level. For instance increasing levels of congestion might be encoded by numerically increasing indications, e.g. pre-congestion notification (PCN) can be encoded in each PDU at three severity levels in IP or MPLS [RFC6660] and the default encapsulation and decapsulation rules [RFC6040] are compatible with this interpretation of the ECN field.

If the arriving inner header is an ECN-PDU, where the inner and outer headers carry indications of congestion of different severity, the more severe indication SHOULD be forwarded in preference to the less severe.

- 5. The inner and outer headers might carry a combination of congestion notification fields that should not be possible given any currently used protocol transitions. For instance, if Encapsulation Guideline 3 in Section 4.3 had been followed, it should not be possible to have a less severe indication of congestion in the outer than in the inner. It MAY be appropriate to log unexpected combinations of headers and possibly raise an alarm.

If a safe outgoing codepoint can be defined for such a PDU, the PDU SHOULD be forwarded rather than dropped. Some implementers discard PDUs with currently unused combinations of headers just in case they represent an attack. However, an approach using alarms and policy-mediated drop is preferable to hard-coded drop, so that operators can keep track of possible attacks but currently unused combinations are not precluded from future use through new standards actions.

4.5. Sequences of Similar Tunnels or Subnets

In some deployments, particularly in 3GPP networks, an IP packet may traverse two or more IP-in-IP tunnels in sequence that all use identical technology (e.g. GTP).

In such cases, it would be sufficient for every encapsulation and decapsulation in the chain to comply with RFC 6040. Alternatively, as an optimisation, a node that decapsulates a packet and immediately re-encapsulates it for the next tunnel MAY copy the incoming outer ECN field directly to the outgoing outer and the incoming inner ECN field directly to the outgoing inner. Then the overall behavior across the sequence of tunnel segments would still be consistent with RFC 6040.

Appendix C of RFC6040 describes how a tunnel egress can monitor how much congestion has been introduced within a tunnel. A network operator might want to monitor how much congestion had been introduced within a whole sequence of tunnels. Using the technique in Appendix C of RFC6040 at the final egress, the operator could monitor the whole sequence of tunnels, but only if the above optimisation were used consistently along the sequence of tunnels, in order to make it appear as a single tunnel. Therefore, tunnel endpoint implementations SHOULD allow the operator to configure whether this optimisation is enabled.

When ECN support is added to a subnet technology, consideration SHOULD be given to a similar optimisation between subnets in sequence if they all use the same technology.

4.6. Reframing and Congestion Markings

The guidance in this section is worded in terms of framing boundaries, but it applies equally whether the protocol data units are frames, cells or packets.

Where an AQM marks the ECN field of IP packets as they queue into a layer-2 link, there will be no problem with framing boundaries, because the ECN markings would be applied directly to IP packets. The guidance in this section is only applicable where an ECN capability is being added to a layer-2 protocol so that layer-2 frames can be ECN-marked by an AQM at layer-2. This would only be necessary where AQM will be applied at pure layer-2 nodes (without IP-awareness).

When layer-2 frame headers are stripped off and IP PDUs with different boundaries are forwarded, the provisions in RFC7141 for handling congestion indications when splitting or merging packets

apply (see Section 2.4 of [RFC7141]). Those provisions include: "The general rule to follow is that the number of octets in packets with congestion indications SHOULD be equivalent before and after merging or splitting." See RFC 7141 for the complete provisions and related discussion, including an exception to that general rule.

As also recommended in RFC 7141, the mechanism for propagating congestion indications SHOULD ensure that any new incoming congestion indication is propagated immediately, and not held awaiting possible arrival of further congestion indications sufficient to indicate congestion for all of the octets of an outgoing IP PDU.

5. Feed-Up-and-Forward Mode: Guidelines for Adding Congestion Notification

The guidance in this section is applicable, for example, when IP packets:

- o are encapsulated in Ethernet headers, which have no support for ECN;
- o are forwarded by the eNode-B (base station) of a 3GPP radio access network, which is required to apply ECN marking during congestion, [LTE-RA], [UTRAN], but the Packet Data Convergence Protocol (PDCP) that encapsulates the IP header over the radio access has no support for ECN.

This guidance also generalizes to encapsulation by other subnet technologies with no native support for explicit congestion notification at the lower layer, but with support for finding and processing an IP header. It is unlikely to be applicable or necessary for IP-in-IP encapsulation, where feed-forward-and-up mode based on [RFC6040] would be more appropriate.

Marking the IP header while switching at layer-2 (by using a layer-3 switch) or while forwarding in a radio access network seems to represent a layering violation. However, it can be considered as a benign optimisation if the guidelines below are followed. Feed-up-and-forward is certainly not a general alternative to implementing feed-forward congestion notification in the lower layer, because:

- o IPv4 and IPv6 are not the only layer-3 protocols that might be encapsulated by lower layer protocols
- o Link-layer encryption might be in use, making the layer-2 payload inaccessible

- o Many Ethernet switches do not have 'layer-3 switch' capabilities so they cannot read or modify an IP payload
- o It might be costly to find an IP header (v4 or v6) when it may be encapsulated by more than one lower layer header, e.g. Ethernet MAC in MAC ([IEEE802.1Q]; previously 802.1ah).

Nonetheless, configuring lower layer equipment to look for an ECN field in an encapsulated IP header is a useful optimisation. If the implementation follows the guidelines below, this optimisation does not have to be confined to a controlled environment such as within a data centre; it could usefully be applied on any network--even if the operator is not sure whether the above issues will never apply:

1. If a native lower-layer congestion notification mechanism exists for a subnet technology, it is safe to mix feed-up-and-forward with feed-forward-and-up on other switches in the same subnet. However, it will generally be more efficient to use the native mechanism.
 2. The depth of the search for an IP header SHOULD be limited. If an IP header is not found soon enough, or an unrecognized or unreadable header is encountered, the switch SHOULD resort to an alternative means of signalling congestion (e.g. drop, or the native lower layer mechanism if available).
 3. It is sufficient to use the first IP header found in the stack; the egress of the relevant tunnel can propagate congestion notification upwards to any more deeply encapsulated IP headers later.
6. Feed-Backward Mode: Guidelines for Adding Congestion Notification

It can be seen from Section 3.3 that congestion notification in a subnet using feed-backward mode has generally not been designed to be directly coupled with IP layer congestion notification. The subnet attempts to minimize congestion internally, and if the incoming load at the ingress exceeds the capacity somewhere through the subnet, the layer 3 buffer into the ingress backs up. Thus, a feed-backward mode subnet is in some sense similar to a null mode subnet, in that there is no need for any direct interaction between the subnet and higher layer congestion notification. Therefore no detailed protocol design guidelines are appropriate. Nonetheless, a more general guideline is appropriate:

A subnetwork technology intended to eventually interface to IP SHOULD NOT be designed using only the feed-backward mode, which is certainly best for a stand-alone subnet, but would need to be

modified to work efficiently as part of the wider Internet, because IP uses feed-forward-and-up mode.

The feed-backward approach at least works beneath IP, where the term 'works' is used only in a narrow functional sense because feed-backward can result in very inefficient and sluggish congestion control--except if it is confined to the subnet directly connected to the original data source, when it is faster than feed-forward. It would be valid to design a protocol that could work in feed-backward mode for paths that only cross one subnet, and in feed-forward-and-up mode for paths that cross subnets.

In the early days of TCP/IP, a similar feed-backward approach was tried for explicit congestion signalling, using source-quench (SQ) ICMP control packets. However, SQ fell out of favour and is now formally deprecated [RFC6633]. The main problem was that it is hard for a data source to tell the difference between a spoofed SQ message and a quench request from a genuine buffer on the path. It is also hard for a lower layer buffer to address an SQ message to the original source port number, which may be buried within many layers of headers, and possibly encrypted.

QCN (also known as backward congestion notification, BCN; see Sections 30--33 of [IEEE802.1Q]; previously known as 802.1Qau) uses a feed-backward mode structurally similar to ATM's relative rate mechanism. However, QCN confines its applicability to scenarios such as some data centres where all endpoints are directly attached by the same Ethernet technology. If a QCN subnet were later connected into a wider IP-based internetwork (e.g. when attempting to interconnect multiple data centres) it would suffer the inefficiency shown in Figure 3.

7. IANA Considerations

This memo includes no request to IANA.

8. Security Considerations

If a lower layer wire protocol is redesigned to include explicit congestion signalling in-band in the protocol header, care SHOULD be taken to ensure that the field used is specified as mutable during transit. Otherwise interior nodes signalling congestion would invalidate any authentication protocol applied to the lower layer header--by altering a header field that had been assumed as immutable.

The redesign of protocols that encapsulate IP in order to propagate congestion signals between layers raises potential signal integrity

concerns. Experimental or proposed approaches exist for assuring the end-to-end integrity of in-band congestion signals, e.g.:

- o Congestion exposure (ConEx) for networks to audit that their congestion signals are not being suppressed by other networks or by receivers, and for networks to police that senders are responding sufficiently to the signals, irrespective of the L4 transport protocol used [RFC7713].
- o A test for a sender to detect whether a network or the receiver is suppressing congestion signals (for example see 2nd para of Section 20.2 of [RFC3168]).

Given these end-to-end approaches are already being specified, it would make little sense to attempt to design hop-by-hop congestion signal integrity into a new lower layer protocol, because end-to-end integrity inherently achieves hop-by-hop integrity.

Section 6 gives vulnerability to spoofing as one of the reasons for deprecating feed-backward mode.

9. Conclusions

Following the guidance in this document enables ECN support to be extended to numerous protocols that encapsulate IP (v4 & v6) in a consistent way, so that IP continues to fulfil its role as an end-to-end interoperability layer. This includes:

- o A wide range of tunnelling protocols including those with various forms of shim header between two IP headers, possibly also separated by a L2 header;
- o A wide range of subnet technologies, particularly those that work in the same 'feed-forward-and-up' mode that is used to support ECN in IP and MPLS.

Guidelines have been defined for supporting propagation of ECN between Ethernet and IP on so-called Layer-3 Ethernet switches, using a 'feed-up-and-forward' mode. This approach could enable other subnet technologies to pass ECN signals into the IP layer, even if they do not support ECN natively.

Finally, attempting to add ECN to a subnet technology in feed-backward mode is deprecated except in special cases, due to its likely sluggish response to congestion.

10. Acknowledgements

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12. Comments Solicited

Comments and questions are encouraged and very welcome. They can be addressed to the IETF Transport Area working group mailing list <tsvwg@ietf.org>, and/or to the authors.

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Appendix A. Changes in This Version (to be removed by RFC Editor)

From ietf-12 to ietf-13

* Following 3rd tsvwg WGLC:

- + Formalized update to RFC 3819 in its own subsection (1.1) and referred to it in the abstract
- + Scope: Clarified that the specification of alternative ECN semantics using ECT(1) was not in RFC 4774, but rather in RFC 8311, and that the problem with using a DSCP to indicate alternative semantics has issues at domain boundaries as well as tunnels.
- + Terminology: tightened up definitions of ECN-PDU and Not-ECN-PDU, and removed definition of Congestion Baseline, given it was only used once.
- + Mentioned QCN where feed-backward is first introduced (S.3), referring forward to where it is discussed more deeply (S.4).
- + Clarified that IS-IS solution to adding ECN support to TRILL was not pursued
- + Completely rewrote the rationale for the guideline about a Standard Congestion Monitoring Baseline, to focus on standardization of the otherwise unknown scenario used, rather than the relative usefulness of the info in each approach
- + Explained the re-framing problem better and added fragmentation as another possible cause of the problem
- + Acknowledged new reviewers
- + Updated references, replaced citations of 802.1Qau and 802.1ah with rolled up 802.1Q, and added citations of Fat trees and Clos Networks
- + Numerous other editorial improvements

From ietf-11 to ietf-12

* Updated references

From ietf-10 to ietf-11

- * Removed short section (was 3) 'Guidelines for All Cases' because it was out of scope, being covered by RFC 4774. Expanded the Scope section (1.2) to explain all this. Explained that the default encap/decap rules already support certain alternative semantics, particularly all three of the alternative semantics for ECT(1): equivalent to ECT(0) , higher severity than ECT(0), and unmarked but implying different marking semantics from ECT(0).
- * Clarified why the QCN example was being given even though not about increment deployment of ECN
- * Pointed to the spoofing issue with feed-backward mode from the Security Considerations section, to aid security review.
- * Removed any ambiguity in the word 'transport' throughout

From ietf-09 to ietf-10

- * Updated section 5.1 on "IP-in-IP tunnels with Shim Headers" to be consistent with updates to draft-ietf-tsvwg-rfc6040update-shim.
- * Removed reference to the ECN nonce, which has been made historic by RFC 8311
- * Removed "Open Issues" Appendix, given all have been addressed.

From ietf-08 to ietf-09

- * Updated para in Intro that listed all the IP-in-IP tunnelling protocols, to instead refer to draft-ietf-tsvwg-rfc6040update-shim
- * Updated section 5.1 on "IP-in-IP tunnels with Shim Headers" to summarize guidance that has evolved as rfc6040update-shim has developed.

From ietf-07 to ietf-08: Refreshed to avoid expiry. Updated references.

From ietf-06 to ietf-07:

- * Added the people involved in liaisons to the acknowledgements.

From ietf-05 to ietf-06:

- * Introduction: Added GUE and Geneve as examples of tightly coupled shims between IP headers that cite RFC 6040. And added VXLAN to list of those that do not.
- * Replaced normative text about tightly coupled shims between IP headers, with reference to new draft-ietf-tsvwg-rfc6040update-shim
- * Wire Protocol Design: Indication of ECN Support: Added TRILL as an example of a well-design protocol that does not need an indication of ECN support in the wire protocol.
- * Encapsulation Guidelines: In the case of a Not-ECN-PDU with a CE outer, replaced SHOULD be dropped, with explanations of when SHOULD or MUST are appropriate.
- * Feed-Up-and-Forward Mode: Explained examples more carefully, referred to PDCP and cited UTRAN spec as well as E-UTRAN.
- * Updated references.
- * Marked open issues as resolved, but did not delete Open Issues Appendix (yet).

From ietf-04 to ietf-05:

- * Explained why tightly coupled shim headers only "SHOULD" comply with RFC 6040, not "MUST".
- * Updated references

From ietf-03 to ietf-04:

- * Addressed Richard Scheffenegger's review comments: primarily editorial corrections, and addition of examples for clarity.

From ietf-02 to ietf-03:

- * Updated references, ad cited RFC4774.

From ietf-01 to ietf-02:

- * Added Section for guidelines that are applicable in all cases.
- * Updated references.

From ietf-00 to ietf-01: Updated references.

From briscoe-04 to ietf-00: Changed filename following tsvwg adoption.

From briscoe-03 to 04:

- * Re-arranged the introduction to describe the purpose of the document first before introducing ECN in more depth. And clarified the introduction throughout.
- * Added applicability to 3GPP TS 36.300.

From briscoe-02 to 03:

- * Scope section:
 - + Added dependence on correct propagation of traffic class information
 - + For the feed-backward mode, deemed multicast and anycast out of scope
- * Ensured all guidelines referring to subnet technologies also refer to tunnels and vice versa by adding applicability sentences at the start of sections 4.1, 4.2, 4.3, 4.4, 4.6 and 5.
- * Added Security Considerations on ensuring congestion signal fields are classed as immutable and on using end-to-end congestion signal integrity technologies rather than hop-by-hop.

From briscoe-01 to 02:

- * Added authors: JK & PT
- * Added
 - + Section 4.1 "IP-in-IP Tunnels with Tightly Coupled Shim Headers"
 - + Section 4.5 "Sequences of Similar Tunnels or Subnets"
 - + roadmap at the start of Section 4, given the subsections have become quite fragmented.
 - + Section 9 "Conclusions"

- * Clarified why transports are starting to be able to saturate interior links
- * Under Section 1.1, addressed the question of alternative signal semantics and included multicast & anycast.
- * Under Section 3.1, included a 3GPP example.
- * Section 4.2. "Wire Protocol Design":
 - + Altered guideline 2. to make it clear that it only applies to the immediate subnet egress, not later ones
 - + Added a reminder that it is only necessary to check that ECN propagates at the egress, not whether interior nodes mark ECN
 - + Added example of how QCN uses 802.1p to indicate support for QCN.
- * Added references to Appendix C of RFC6040, about monitoring the amount of congestion signals introduced within a tunnel
- * Appendix A: Added more issues to be addressed, including plan to produce a standards track update to IP-in-IP tunnel protocols.
- * Updated acks and references

From briscoe-00 to 01:

- * Intended status: BCP (was Informational) & updates 3819 added.
- * Briefer Introduction: Introductory para justifying benefits of ECN. Moved all but a brief enumeration of modes of operation to their own new section (from both Intro & Scope). Introduced incr. deployment as most tricky part.
- * Tightened & added to terminology section
- * Structured with Modes of Operation, then Guidelines section for each mode.
- * Tightened up guideline text to remove vagueness / passive voice / ambiguity and highlight main guidelines as numbered items.
- * Added Outstanding Document Issues Appendix

* Updated references

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Explicit Congestion Notification (ECN) Protocol for Very Low Queuing
Delay (L4S)
draft-ietf-tsvwg-ecn-l4s-id-25

Abstract

This specification defines the protocol to be used for a new network service called low latency, low loss and scalable throughput (L4S). L4S uses an Explicit Congestion Notification (ECN) scheme at the IP layer that is similar to the original (or 'Classic') ECN approach, except as specified within. L4S uses 'scalable' congestion control, which induces much more frequent control signals from the network and it responds to them with much more fine-grained adjustments, so that very low (typically sub-millisecond on average) and consistently low queuing delay becomes possible for L4S traffic without compromising link utilization. Thus even capacity-seeking (TCP-like) traffic can have high bandwidth and very low delay at the same time, even during periods of high traffic load.

The L4S identifier defined in this document distinguishes L4S from 'Classic' (e.g. TCP-Reno-friendly) traffic. It gives an incremental migration path so that suitably modified network bottlenecks can distinguish and isolate existing traffic that still follows the Classic behaviour, to prevent it degrading the low queuing delay and low loss of L4S traffic. This specification defines the rules that L4S transports and network elements need to follow with the intention that L4S flows neither harm each other's performance nor that of Classic traffic. Examples of new active queue management (AQM) marking algorithms and examples of new transports (whether TCP-like or real-time) are specified separately.

Status of This Memo

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

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1. Introduction

This specification defines the protocol to be used for a new network service called low latency, low loss and scalable throughput (L4S). L4S uses an Explicit Congestion Notification (ECN) scheme at the IP layer with the same set of codepoint transitions as the original (or 'Classic') Explicit Congestion Notification (ECN [RFC3168]). RFC 3168 required an ECN mark to be equivalent to a drop, both when applied in the network and when responded to by a transport. Unlike Classic ECN marking, the network applies L4S marking more immediately and more aggressively than drop, and the transport response to each mark is reduced and smoothed relative to that for drop. The two changes counterbalance each other so that the throughput of an L4S flow will be roughly the same as a comparable non-L4S flow under the same conditions. Nonetheless, the much more frequent ECN control signals and the finer responses to these signals result in very low queuing delay without compromising link utilization, and this low delay can be maintained during high load. For instance, queuing delay under heavy and highly varying load with the example DCTCP/DualQ solution cited below on a DSL or Ethernet link is sub-millisecond on average and roughly 1 to 2 milliseconds at the 99th percentile without losing link utilization [DualPI2Linux], [DCTtH19]. Note that the inherent queuing delay while waiting to acquire a discontinuous medium such as WiFi has to be minimized in its own right, so it would be additional to the above (see section 6.3 of the L4S architecture [I-D.ietf-tsvwg-l4s-arch]).

L4S relies on 'scalable' congestion controls for these delay properties and for preserving low delay as flow rate scales, hence the name. The congestion control used in Data Center TCP (DCTCP) is an example of a scalable congestion control, but DCTCP is applicable solely to controlled environments like data centres [RFC8257], because it is too aggressive to co-exist with existing TCP-Reno-friendly traffic. The DualQ Coupled AQM, which is defined in a complementary experimental specification [I-D.ietf-tsvwg-aqm-dualq-coupled], is an AQM framework that enables scalable congestion controls derived from DCTCP to co-exist with existing traffic, each getting roughly the same flow rate when they compete under similar conditions. Note that a scalable congestion control is still not safe to deploy on the Internet unless it satisfies the requirements listed in Section 4.

L4S is not only for elastic (TCP-like) traffic - there are scalable congestion controls for real-time media, such as the L4S variant of the SCReAM [RFC8298] real-time media congestion avoidance technique (RMCAT). The factor that distinguishes L4S from Classic traffic is its behaviour in response to congestion. The transport wire protocol, e.g. TCP, QUIC, SCTP, DCCP, RTP/RTCP, is orthogonal (and therefore not suitable for distinguishing L4S from Classic packets).

The L4S identifier defined in this document is the key piece that distinguishes L4S from 'Classic' (e.g. Reno-friendly) traffic. It gives an incremental migration path so that suitably modified network bottlenecks can distinguish and isolate existing Classic traffic from L4S traffic to prevent the former from degrading the very low delay and loss of the new scalable transports, without harming Classic performance at these bottlenecks. Initial implementation of the separate parts of the system has been motivated by the performance benefits.

1.1. Latency, Loss and Scaling Problems

Latency is becoming the critical performance factor for many (most?) applications on the public Internet, e.g. interactive Web, Web services, voice, conversational video, interactive video, interactive remote presence, instant messaging, online gaming, remote desktop, cloud-based applications, and video-assisted remote control of machinery and industrial processes. In the 'developed' world, further increases in access network bit-rate offer diminishing returns, whereas latency is still a multi-faceted problem. In the last decade or so, much has been done to reduce propagation time by placing caches or servers closer to users. However, queuing remains a major intermittent component of latency.

The Diffserv architecture provides Expedited Forwarding [RFC3246], so that low latency traffic can jump the queue of other traffic. If growth in high-throughput latency-sensitive applications continues, periods with solely latency-sensitive traffic will become increasingly common on links where traffic aggregation is low. For instance, on the access links dedicated to individual sites (homes, small enterprises or mobile devices). These links also tend to become the path bottleneck under load. During these periods, if all the traffic were marked for the same treatment, at these bottlenecks Diffserv would make no difference. Instead, it becomes imperative to remove the underlying causes of any unnecessary delay.

The bufferbloat project has shown that excessively-large buffering ('bufferbloat') has been introducing significantly more delay than the underlying propagation time. These delays appear only intermittently -- only when a capacity-seeking (e.g. TCP) flow is long enough for the queue to fill the buffer, making every packet in other flows sharing the buffer sit through the queue.

Active queue management (AQM) was originally developed to solve this problem (and others). Unlike Diffserv, which gives low latency to some traffic at the expense of others, AQM controls latency for all traffic in a class. In general, AQM methods introduce an increasing level of discard from the buffer the longer the queue persists above a shallow threshold. This gives sufficient signals to capacity-seeking (aka. greedy) flows to keep the buffer empty for its intended purpose: absorbing bursts. However, RED [RFC2309] and other algorithms from the 1990s were sensitive to their configuration and hard to set correctly. So, this form of AQM was not widely deployed.

More recent state-of-the-art AQM methods, e.g. FQ-CoDel [RFC8290], PIE [RFC8033], Adaptive RED [ARED01], are easier to configure, because they define the queuing threshold in time not bytes, so it is invariant for different link rates. However, no matter how good the AQM, the sawtooth sending window of a Classic congestion control will either cause queuing delay to vary or cause the link to be underutilized. Even with a perfectly tuned AQM, the additional queuing delay will be of the same order as the underlying speed-of-light delay across the network, thereby roughly doubling the total round-trip time.

If a sender's own behaviour is introducing queuing delay variation, no AQM in the network can 'un-vary' the delay without significantly compromising link utilization. Even flow-queuing (e.g. [RFC8290]), which isolates one flow from another, cannot isolate a flow from the delay variations it inflicts on itself. Therefore those applications that need to seek out high bandwidth but also need low latency will have to migrate to scalable congestion control.

Altering host behaviour is not enough on its own though. Even if hosts adopt low latency behaviour (scalable congestion controls), they need to be isolated from the behaviour of existing Classic congestion controls that induce large queue variations. L4S enables that migration by providing latency isolation in the network and distinguishing the two types of packets that need to be isolated: L4S and Classic. L4S isolation can be achieved with a queue per flow (e.g. [RFC8290]) but a DualQ [I-D.ietf-tsvwg-aqm-dualq-coupled] is sufficient, and actually gives better tail latency. Both approaches are addressed in this document.

The DualQ solution was developed to make very low latency available without requiring per-flow queues at every bottleneck. This was because per-flow-queuing (FQ) has well-known downsides - not least the need to inspect transport layer headers in the network, which makes it incompatible with privacy approaches such as IPsec VPN tunnels, and incompatible with link layer queue management, where transport layer headers can be hidden, e.g. 5G.

Latency is not the only concern addressed by L4S: It was known when TCP congestion avoidance was first developed that it would not scale to high bandwidth-delay products (footnote 6 of Jacobson and Karels [TCP-CA]). Given regular broadband bit-rates over WAN distances are already [RFC3649] beyond the scaling range of Reno congestion control, 'less unscalable' Cubic [RFC8312] and Compound [I-D.sridharan-tcpm-ctcp] variants of TCP have been successfully deployed. However, these are now approaching their scaling limits. Unfortunately, fully scalable congestion controls such as DCTCP [RFC8257] outcompete Classic ECN congestion controls sharing the same queue, which is why they have been confined to private data centres or research testbeds.

It turns out that these scalable congestion control algorithms that solve the latency problem can also solve the scalability problem of Classic congestion controls. The finer sawteeth in the congestion window have low amplitude, so they cause very little queuing delay variation and the average time to recover from one congestion signal to the next (the average duration of each sawtooth) remains invariant, which maintains constant tight control as flow-rate scales. A background paper [DCttH19] gives the full explanation of why the design solves both the latency and the scaling problems, both in plain English and in more precise mathematical form. The explanation is summarised without the maths in Section 4 of the L4S architecture [I-D.ietf-tsvwg-l4s-arch].

1.2. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119]. In this document, these words will appear with that interpretation only when in ALL CAPS. Lower case uses of these words are not to be interpreted as carrying RFC-2119 significance.

Note: The L4S architecture [I-D.ietf-tsvwg-l4s-arch] repeats the following definitions, but if there are accidental differences those below take precedence.

Classic Congestion Control: A congestion control behaviour that can

co-exist with standard Reno [RFC5681] without causing significantly negative impact on its flow rate [RFC5033]. With Classic congestion controls, such as Reno or Cubic, because flow rate has scaled since TCP congestion control was first designed in 1988, it now takes hundreds of round trips (and growing) to recover after a congestion signal (whether a loss or an ECN mark) as shown in the examples in section 5.1 of the L4S architecture [I-D.ietf-tsvwg-l4s-arch] and in [RFC3649]. Therefore control of queuing and utilization becomes very slack, and the slightest disturbances (e.g. from new flows starting) prevent a high rate from being attained.

Scalable Congestion Control: A congestion control where the average time from one congestion signal to the next (the recovery time) remains invariant as the flow rate scales, all other factors being equal. This maintains the same degree of control over queueing and utilization whatever the flow rate, as well as ensuring that high throughput is robust to disturbances. For instance, DCTCP averages 2 congestion signals per round-trip whatever the flow rate, as do other recently developed scalable congestion controls, e.g. Relentless TCP [Mathis09], TCP Prague [I-D.briscoe-iccrp-prague-congestion-control], [PragueLinux], BBRv2 [BBRv2], [I-D.cardwell-iccrp-bbr-congestion-control] and the L4S variant of SCREAM for real-time media [SCReAM], [RFC8298]). See Section 4.3 for more explanation.

Classic service: The Classic service is intended for all the congestion control behaviours that co-exist with Reno [RFC5681] (e.g. Reno itself, Cubic [RFC8312], Compound [I-D.sridharan-tcpm-ctcp], TFRC [RFC5348]). The term 'Classic queue' means a queue providing the Classic service.

Low-Latency, Low-Loss Scalable throughput (L4S) service: The 'L4S' service is intended for traffic from scalable congestion control algorithms, such as TCP Prague [I-D.briscoe-iccrp-prague-congestion-control], which was derived from DCTCP [RFC8257]. The L4S service is for more general traffic than just TCP Prague -- it allows the set of congestion controls with similar scaling properties to Prague to evolve, such as the examples listed above (Relentless, SCReAM). The term 'L4S queue' means a queue providing the L4S service.

The terms Classic or L4S can also qualify other nouns, such as 'queue', 'codepoint', 'identifier', 'classification', 'packet', 'flow'. For example: an L4S packet means a packet with an L4S identifier sent from an L4S congestion control.

Both Classic and L4S services can cope with a proportion of unresponsive or less-responsive traffic as well, but in the L4S case its rate has to be smooth enough or low enough not to build a queue (e.g. DNS, VoIP, game sync datagrams, etc).

Reno-friendly: The subset of Classic traffic that is friendly to the standard Reno congestion control defined for TCP in [RFC5681]. The TFRC spec. [RFC5348] indirectly implies that 'friendly' is defined as "generally within a factor of two of the sending rate of a TCP flow under the same conditions". Reno-friendly is used here in place of 'TCP-friendly', given the latter has become imprecise, because the TCP protocol is now used with so many different congestion control behaviours, and Reno is used in non-TCP transports such as QUIC [RFC9000].

Classic ECN: The original Explicit Congestion Notification (ECN) protocol [RFC3168], which requires ECN signals to be treated the same as drops, both when generated in the network and when responded to by the sender. For L4S, the names used for the four codepoints of the 2-bit IP-ECN field are unchanged from those defined in [RFC3168]: Not ECT, ECT(0), ECT(1) and CE, where ECT stands for ECN-Capable Transport and CE stands for Congestion Experienced. A packet marked with the CE codepoint is termed 'ECN-marked' or sometimes just 'marked' where the context makes ECN obvious.

Site: A home, mobile device, small enterprise or campus, where the network bottleneck is typically the access link to the site. Not all network arrangements fit this model but it is a useful, widely applicable generalization.

1.3. Scope

The new L4S identifier defined in this specification is applicable for IPv4 and IPv6 packets (as for Classic ECN [RFC3168]). It is applicable for the unicast, multicast and anycast forwarding modes.

The L4S identifier is an orthogonal packet classification to the Differentiated Services Code Point (DSCP) [RFC2474]. Section 5.4 explains what this means in practice.

This document is intended for experimental status, so it does not update any standards track RFCs. Therefore it depends on [RFC8311], which is a standards track specification that:

- * updates the ECN proposed standard [RFC3168] to allow experimental track RFCs to relax the requirement that an ECN mark must be equivalent to a drop (when the network applies markings and/or

when the sender responds to them). For instance, in the ABE experiment [RFC8511] this permits a sender to respond less to ECN marks than to drops;

- * changes the status of the experimental ECN nonce [RFC3540] to historic;
- * makes consequent updates to the following additional proposed standard RFCs to reflect the above two bullets:
 - ECN for RTP [RFC6679];
 - the congestion control specifications of various DCCP congestion control identifier (CCID) profiles [RFC4341], [RFC4342], [RFC5622].

This document is about identifiers that are used for interoperation between hosts and networks. So the audience is broad, covering developers of host transports and network AQMs, as well as covering how operators might wish to combine various identifiers, which would require flexibility from equipment developers.

2. Choice of L4S Packet Identifier: Requirements

This subsection briefly records the process that led to the chosen L4S identifier.

The identifier for packets using the Low Latency, Low Loss, Scalable throughput (L4S) service needs to meet the following requirements:

- * it SHOULD survive end-to-end between source and destination endpoints: across the boundary between host and network, between interconnected networks, and through middleboxes;
- * it SHOULD be visible at the IP layer;
- * it SHOULD be common to IPv4 and IPv6 and transport-agnostic;
- * it SHOULD be incrementally deployable;
- * it SHOULD enable an AQM to classify packets encapsulated by outer IP or lower-layer headers;
- * it SHOULD consume minimal extra codepoints;
- * it SHOULD be consistent on all the packets of a transport layer flow, so that some packets of a flow are not served by a different queue to others.

Whether the identifier would be recoverable if the experiment failed is a factor that could be taken into account. However, this has not been made a requirement, because that would favour schemes that would be easier to fail, rather than those more likely to succeed.

It is recognised that any choice of identifier is unlikely to satisfy all these requirements, particularly given the limited space left in the IP header. Therefore a compromise will always be necessary, which is why all the above requirements are expressed with the word 'SHOULD' not 'MUST'.

After extensive assessment of alternative schemes, "ECT(1) and CE codepoints" was chosen as the best compromise. Therefore this scheme is defined in detail in the following sections, while Appendix B records its pros and cons against the above requirements.

3. L4S Packet Identification

The L4S treatment is an experimental track alternative packet marking treatment to the Classic ECN treatment in [RFC3168], which has been updated by [RFC8311] to allow experiments such as the one defined in the present specification. [RFC4774] discusses some of the issues and evaluation criteria when defining alternative ECN semantics. Like Classic ECN, L4S ECN identifies both network and host behaviour: it identifies the marking treatment that network nodes are expected to apply to L4S packets, and it identifies packets that have been sent from hosts that are expected to comply with a broad type of sending behaviour.

For a packet to receive L4S treatment as it is forwarded, the sender sets the ECN field in the IP header to the ECT(1) codepoint. See Section 4 for full transport layer behaviour requirements, including feedback and congestion response.

A network node that implements the L4S service always classifies arriving ECT(1) packets for L4S treatment and by default classifies CE packets for L4S treatment unless the heuristics described in Section 5.3 are employed. See Section 5 for full network element behaviour requirements, including classification, ECN-marking and interaction of the L4S identifier with other identifiers and per-hop behaviours.

4. Transport Layer Behaviour (the 'Prague Requirements')

4.1. Codepoint Setting

A sender that wishes a packet to receive L4S treatment as it is forwarded, MUST set the ECN field in the IP header (v4 or v6) to the ECT(1) codepoint.

4.2. Prerequisite Transport Feedback

For a transport protocol to provide scalable congestion control (Section 4.3) it MUST provide feedback of the extent of CE marking on the forward path. When ECN was added to TCP [RFC3168], the feedback method reported no more than one CE mark per round trip. Some transport protocols derived from TCP mimic this behaviour while others report the accurate extent of ECN marking. This means that some transport protocols will need to be updated as a prerequisite for scalable congestion control. The position for a few well-known transport protocols is given below.

TCP: Support for the accurate ECN feedback requirements [RFC7560] (such as that provided by AccECN [I-D.ietf-tcpm-accurate-ecn]) by both ends is a prerequisite for scalable congestion control in TCP. Therefore, the presence of ECT(1) in the IP headers even in one direction of a TCP connection will imply that both ends support accurate ECN feedback. However, the converse does not apply. So even if both ends support AccECN, either of the two ends can choose not to use a scalable congestion control, whatever the other end's choice.

SCTP: A suitable ECN feedback mechanism for SCTP could add a chunk to report the number of received CE marks (e.g. [I-D.stewart-tsvwg-sctpecn]), and update the ECN feedback protocol sketched out in Appendix A of the original standards track specification of SCTP [RFC4960].

RTP over UDP: A prerequisite for scalable congestion control is for both (all) ends of one media-level hop to signal ECN support [RFC6679] and use the new generic RTCP feedback format of [RFC8888]. The presence of ECT(1) implies that both (all) ends of that media-level hop support ECN. However, the converse does not apply. So each end of a media-level hop can independently choose not to use a scalable congestion control, even if both ends support ECN.

QUIC: Support for sufficiently fine-grained ECN feedback is provided by the v1 IETF QUIC transport [RFC9000].

DCCP: The ACK vector in DCCP [RFC4340] is already sufficient to

report the extent of CE marking as needed by a scalable congestion control.

4.3. Prerequisite Congestion Response

As a condition for a host to send packets with the L4S identifier (ECT(1)), it SHOULD implement a congestion control behaviour that ensures that, in steady state, the average duration between induced ECN marks does not increase as flow rate scales up, all other factors being equal. This is termed a scalable congestion control. This invariant duration ensures that, as flow rate scales, the average period with no feedback information about capacity does not become excessive. It also ensures that queue variations remain small, without having to sacrifice utilization.

With a congestion control that sawtooths to probe capacity, this duration is called the recovery time, because each time the sawtooth yields, on average it takes this time to recover to its previous high point. A scalable congestion control does not have to sawtooth, but it has to coexist with scalable congestion controls that do.

For instance, for DCTCP [RFC8257], TCP Prague [I-D.briscoe-iccrp-prague-congestion-control], [PragueLinux] and the L4S variant of SCReAM [RFC8298], the average recovery time is always half a round trip (or half a reference round trip), whatever the flow rate.

As with all transport behaviours, a detailed specification (probably an experimental RFC) is expected for each congestion control, following the guidelines for specifying new congestion control algorithms in [RFC5033]. In addition it is expected to document these L4S-specific matters, specifically the timescale over which the proportionality is averaged, and control of burstiness. The recovery time requirement above is worded as a 'SHOULD' rather than a 'MUST' to allow reasonable flexibility for such implementations.

The condition 'all other factors being equal', allows the recovery time to be different for different round trip times, as long as it does not increase with flow rate for any particular RTT.

Saying that the recovery time remains roughly invariant is equivalent to saying that the number of ECN CE marks per round trip remains invariant as flow rate scales, all other factors being equal. For instance, an average recovery time of half of 1 RTT is equivalent to 2 ECN marks per round trip. For those familiar with steady-state congestion response functions, it is also equivalent to say that the congestion window is inversely proportional to the proportion of bytes in packets marked with the CE codepoint (see section 2 of [PI2]).

In order to coexist safely with other Internet traffic, a scalable congestion control MUST NOT tag its packets with the ECT(1) codepoint unless it complies with the following bulleted requirements:

1. A scalable congestion control MUST be capable of being replaced by a Classic congestion control (by application and/or by administrative control). If a Classic congestion control is activated, it will not tag its packets with the ECT(1) codepoint (see Appendix A.1.3 for rationale).
2. As well as responding to ECN markings, a scalable congestion control MUST react to packet loss in a way that will coexist safely with Classic congestion controls such as standard Reno [RFC5681], as required by [RFC5033] (see Appendix A.1.4 for rationale).
3. In uncontrolled environments, monitoring MUST be implemented to support detection of problems with an ECN-capable AQM at the path bottleneck that appears not to support L4S and might be in a shared queue. Such monitoring SHOULD be applied to live traffic that is using Scalable congestion control. Alternatively, monitoring need not be applied to live traffic, if monitoring has been arranged to cover the paths that live traffic takes through uncontrolled environments.

A function to detect the above problems with an ECN-capable AQM MUST also be implemented and used. The detection function SHOULD be capable of making the congestion control adapt its ECN-marking response in real-time to coexist safely with Classic congestion controls such as standard Reno [RFC5681], as required by [RFC5033]. This could be complemented by more detailed offline detection of potential problems. If only offline detection is used and potential problems with such an AQM are detected on certain paths, the scalable congestion control MUST be replaced by a Classic congestion control, at least for the problem paths.

See Section 4.3.1, Appendix A.1.5 and the L4S operational guidance [I-D.ietf-tsvwg-l4sops] for rationale.

Note that a scalable congestion control is not expected to change to setting ECT(0) while it transiently adapts to coexist with Classic congestion controls, whereas a replacement congestion control that solely behaves in the Classic way will set ECT(0).

4. In the range between the minimum likely RTT and typical RTTs expected in the intended deployment scenario, a scalable congestion control MUST converge towards a rate that is as independent of RTT as is possible without compromising stability or efficiency (see Appendix A.1.6 for rationale).
5. A scalable congestion control SHOULD remain responsive to congestion when typical RTTs over the public Internet are significantly smaller because they are no longer inflated by queuing delay. It would be preferable for the minimum window of a scalable congestion control to be lower than 1 segment rather than use the timeout approach described for TCP in S.6.1.2 of the ECN spec [RFC3168] (or an equivalent for other transports). However, a lower minimum is not set as a formal requirement for L4S experiments (see Appendix A.1.7 for rationale).
6. A scalable congestion control's loss detection SHOULD be resilient to reordering over an adaptive time interval that scales with throughput and adapts to reordering (as in RACK [RFC8985]), as opposed to counting only in fixed units of packets (as in the 3 DupACK rule of New Reno [RFC5681] and [RFC6675], which is not scalable). As data rates increase (e.g., due to new and/or improved technology), congestion controls that detect loss by counting in units of packets become more likely to incorrectly treat reordering events as congestion-caused loss events (see Appendix A.1.8 for further rationale). This requirement does not apply to congestion controls that are solely used in controlled environments where the network introduces hardly any reordering.
7. A scalable congestion control is expected to limit the queue caused by bursts of packets. It would not seem necessary to set the limit any lower than 10% of the minimum RTT expected in a typical deployment (e.g. additional queuing of roughly 250 us for the public Internet). This would be converted to a number of packets under the worst-case assumption that the bottleneck link capacity equals the current flow rate. No normative requirement to limit bursts is given here and, until there is more industry experience from the L4S experiment, it is not even known whether one is needed - it seems to be in an L4S sender's self-interest to limit bursts.

Each sender in a session can use a scalable congestion control independently of the congestion control used by the receiver(s) when they send data. Therefore there might be ECT(1) packets in one direction and ECT(0) or Not-ECT in the other.

Later (Section 5.4.1.1) this document discusses the conditions for mixing other "'Safe' Unresponsive Traffic" (e.g. DNS, LDAP, NTP, voice, game sync packets) with L4S traffic. To be clear, although such traffic can share the same queue as L4S traffic, it is not appropriate for the sender to tag it as ECT(1), except in the (unlikely) case that it satisfies the above conditions.

4.3.1. Guidance on Congestion Response in the RFC Series

RFC 3168 requires the congestion responses to a CE-marked packet and a dropped packet to be the same. RFC 8311 is a standards-track update to RFC 3168 intended to enable experimentation with ECN, including the L4S experiment. RFC 8311 allows an experimental congestion control's response to a CE-marked packet to differ from the response to a dropped packet, provided that the differences are documented in an experimental RFC, such as the present document.

BCP 124 [RFC4774] gives guidance to protocol designers, when specifying alternative semantics for the ECN field. RFC 8311 explained that it did not need to update the best current practice in BCP 124 in order to relax the 'equivalence with drop' requirement because, although BCP 124 quotes the same requirement from RFC 3168, the BCP does not impose requirements based on it. BCP 124 describes three options for incremental deployment, with Option 3 (in Section 4.3 of BCP 124) best matching the L4S case. Option 3's requirement for end-nodes is that they respond to CE marks "in a way that is friendly to flows using IETF-conformant congestion control." This echoes other general congestion control requirements in the RFC series, for example [RFC5033], which says "...congestion controllers that have a significantly negative impact on traffic using standard congestion control may be suspect", or [RFC8085] concerning UDP congestion control says "Bulk-transfer applications that choose not to implement TFRC or TCP-like windowing SHOULD implement a congestion control scheme that results in bandwidth (capacity) use that competes fairly with TCP within an order of magnitude."

The third normative bullet in Section 4.3 above (which concerns L4S response to congestion from a Classic ECN AQM) aims to ensure that these 'coexistence' requirements are satisfied, but it makes some compromises. This subsection highlights and justifies those compromises and Appendix A.1.5 and the L4S operational guidance [I-D.ietf-tsvwg-l4sops] give detailed analysis, examples and references (the normative text in that bullet takes precedence if any

informative elaboration leads to ambiguity). The approach is based on an assessment of the risk of harm, which is a combination of the prevalence of the conditions necessary for harm to occur, and the potential severity of the harm if they do.

Prevalence: There are three cases:

- * Drop Tail: Coexistence between L4S and Classic flows is not in doubt where the bottleneck does not support any form of ECN, which has remained by far the most prevalent case since the ECN RFC was published in 2001.
- * L4S: Coexistence is not in doubt if the bottleneck supports L4S.
- * Classic ECN [RFC3168]: The compromises centre around cases where the bottleneck supports Classic ECN but not L4S. But it depends on which sub-case:
 - Shared Queue with Classic ECN: The members of the Transport Working group are not aware of any current deployments of single-queue Classic ECN bottlenecks in the Internet. Nonetheless, at the scale of the Internet, rarity need not imply small numbers, nor that there will be rarity in future.
 - Per-Flow-queues with Classic ECN: Most AQMs with per-flow-queuing (FQ) deployed from 2012 onwards had Classic ECN enabled by default, specifically FQ-CoDel [RFC8290] and COBALT [COBALT]. But the compromises only apply to the second of two further sub-cases:
 - o With per-flow-queuing, co-existence between Classic and L4S flows is not normally a problem, because different flows are not meant to be in the same queue (BCP 124 [RFC4774] did not foresee the introduction of per-flow-queuing, which appeared as a potential isolation technique some eight years after the BCP was published).
 - o However, the isolation between L4S and Classic flows is not perfect in cases where the hashes of flow IDs collide or where multiple flows within a layer-3 VPN are encapsulated within one flow ID.

To summarize, the coexistence problem is confined to cases of imperfect flow isolation in an FQ, or in potential cases where a Classic ECN AQM has been deployed in a shared queue (see the L4S operational guidance [I-D.ietf-tsvwg-l4sops] for further details

including recent surveys attempting to quantify prevalence). Further, if one of these cases does occur, the coexistence problem does not arise unless sources of Classic and L4S flows are simultaneously sharing the same bottleneck queue (e.g. different applications in the same household) and flows of each type have to be large enough to coincide for long enough for any throughput imbalance to have developed.

Severity: Where long-running L4S and Classic flows coincide in a shared queue, testing of one L4S congestion control (TCP Prague) has found that the imbalance in average throughput between an L4S and a Classic flow can reach 25:1 in favour of L4S in the worst case [ecn-fallback]. However, when capacity is most scarce, the Classic flow gets a higher proportion of the link, for instance over a 4 Mb/s link the throughput ratio is below ~10:1 over paths with a base RTT below 100 ms, and falls below ~5:1 for base RTTs below 20ms.

These throughput ratios can clearly fall well outside current RFC guidance on coexistence. However, the tendency towards leaving a greater share for Classic flows at lower link rate and the very limited prevalence of the conditions necessary for harm to occur led to the possibility of allowing the RFC requirements to be compromised, albeit briefly:

- * The recommended approach is still to detect and adapt to a Classic ECN AQM in real-time, which is fully consistent with all the RFCs on coexistence. In other words, the "SHOULD"s in the third bullet of Section 4.3 above expect the sender to implement something similar to the proof of concept code that detects the presence of a Classic ECN AQM and falls back to a Classic congestion response within a few round trips [ecn-fallback]. However, although this code reliably detects a Classic ECN AQM, the current code can also wrongly categorize an L4S AQM as Classic, most often in cases when link rate is low or RTT is high. Although this is the safe way round, and although implementers are expected to be able to improve on this proof of concept, concerns have been raised that implementers might lose faith in such detection and disable it.
- * Therefore the third bullet in Section 4.3 above allows a compromise where coexistence could diverge from the requirements in the RFC Series briefly, but mandatory monitoring is required, in order to detect such cases and trigger remedial action. This approach tolerates a brief divergence from the RFCs given the likely low prevalence and given harm here means a flow progresses more slowly than otherwise, but it does progress. The L4S operational guidance [I-D.ietf-tsvwg-l4sops] outlines a range of example remedial actions that include alterations either to the

sender or to the network. However, the final normative requirement in the third bullet of Section 4.3 above places ultimate responsibility for remedial action on the sender. If coexistence problems with a Classic ECN AQM are detected (implying they have not been resolved by the network), it says the sender "MUST" revert to a Classic congestion control."

[I-D.ietf-tsvwg-l4sops] also gives example ways in which L4S congestion controls can be rolled out initially in lower risk scenarios.

4.4. Filtering or Smoothing of ECN Feedback

Section 5.2 below specifies that an L4S AQM is expected to signal L4S ECN immediately, to avoid introducing delay due to filtering or smoothing. This contrasts with a Classic AQM, which filters out variations in the queue before signalling ECN marking or drop. In the L4S architecture [I-D.ietf-tsvwg-l4s-arch], responsibility for smoothing out these variations shifts to the sender's congestion control.

This shift of responsibility has the advantage that each sender can smooth variations over a timescale proportionate to its own RTT. Whereas, in the Classic approach, the network doesn't know the RTTs of any of the flows, so it has to smooth out variations for a worst-case RTT to ensure stability. For all the typical flows with shorter RTT than the worst-case, this makes congestion control unnecessarily sluggish.

This also gives an L4S sender the choice not to smooth, depending on its context (start-up, congestion avoidance, etc). Therefore, this document places no requirement on an L4S congestion control to smooth out variations in any particular way. Implementers are encouraged to openly publish the approach they take to smoothing, and the results and experience they gain during the L4S experiment.

5. Network Node Behaviour

5.1. Classification and Re-Marking Behaviour

A network node that implements the L4S service:

- * MUST classify arriving ECT(1) packets for L4S treatment, unless overridden by another classifier (e.g., see Section 5.4.1.2);

- * MUST classify arriving CE packets for L4S treatment as well, unless overridden by a another classifier or unless the exception referred to next applies;

CE packets might have originated as ECT(1) or ECT(0), but the above rule to classify them as if they originated as ECT(1) is the safe choice (see Appendix B for rationale). The exception is where some flow-aware in-network mechanism happens to be available for distinguishing CE packets that originated as ECT(0), as described in Section 5.3, but there is no implication that such a mechanism is necessary.

An L4S AQM treatment follows similar codepoint transition rules to those in RFC 3168. Specifically, the ECT(1) codepoint MUST NOT be changed to any other codepoint than CE, and CE MUST NOT be changed to any other codepoint. An ECT(1) packet is classified as ECN-capable and, if congestion increases, an L4S AQM algorithm will increasingly mark the ECN field as CE, otherwise forwarding packets unchanged as ECT(1). Necessary conditions for an L4S marking treatment are defined in Section 5.2.

Under persistent overload an L4S marking treatment MUST begin applying drop to L4S traffic until the overload episode has subsided, as recommended for all AQM methods in [RFC7567] (Section 4.2.1), which follows the similar advice in RFC 3168 (Section 7). During overload, it MUST apply the same drop probability to L4S traffic as it would to Classic traffic.

Where an L4S AQM is transport-aware, this requirement could be satisfied by using drop in only the most overloaded individual per-flow AQMs. In a DualQ with flow-aware queue protection (e.g. [I-D.briscoe-docsis-q-protection]), this could be achieved by redirecting packets in those flows contributing most to the overload out of the L4S queue so that they are subjected to drop in the Classic queue.

For backward compatibility in uncontrolled environments, a network node that implements the L4S treatment MUST also implement an AQM treatment for the Classic service as defined in Section 1.2. This Classic AQM treatment need not mark ECT(0) packets, but if it does, see Section 5.2 for the strengths of the markings relative to drop. It MUST classify arriving ECT(0) and Not-ECT packets for treatment by this Classic AQM (for the DualQ Coupled AQM, see the extensive discussion on classification in Sections 2.3 and 2.5.1.1 of [I-D.ietf-tsvwg-aqm-dualq-coupled]).

In case unforeseen problems arise with the L4S experiment, it MUST be possible to configure an L4S implementation to disable the L4S treatment. Once disabled, all packets of all ECN codepoints will receive Classic treatment and ECT(1) packets MUST be treated as if they were Not-ECT.

5.2. The Strength of L4S CE Marking Relative to Drop

The relative strengths of L4S CE and drop are irrelevant where AQMs are implemented in separate queues per-application-flow, which are then explicitly scheduled (e.g. with an FQ scheduler as in FQ-CoDel [RFC8290]). Nonetheless, the relationship between them needs to be defined for the coupling between L4S and Classic congestion signals in a DualQ Coupled AQM [I-D.ietf-tsvwg-aqm-dualq-coupled], as below.

Unless an AQM node schedules application flows explicitly, the likelihood that the AQM drops a Not-ECT Classic packet (p_C) MUST be roughly proportional to the square of the likelihood that it would have marked it if it had been an L4S packet (p_L). That is

$$p_C \sim (p_L / k)^2$$

The constant of proportionality (k) does not have to be standardised for interoperability, but a value of 2 is RECOMMENDED. The term 'likelihood' is used above to allow for marking and dropping to be either probabilistic or deterministic.

This formula ensures that Scalable and Classic flows will converge to roughly equal congestion windows, for the worst case of Reno congestion control. This is because the congestion windows of Scalable and Classic congestion controls are inversely proportional to p_L and $\sqrt{p_C}$ respectively. So squaring p_C in the above formula counterbalances the square root that characterizes Reno-friendly flows.

Note that, contrary to RFC 3168, an AQM implementing the L4S and Classic treatments does not mark an ECT(1) packet under the same conditions that it would have dropped a Not-ECT packet, as allowed by [RFC8311], which updates RFC 3168. However, if it marks ECT(0) packets, it does so under the same conditions that it would have dropped a Not-ECT packet [RFC3168].

Also, In the L4S architecture [I-D.ietf-tsvwg-l4s-arch], the sender, not the network, is responsible for smoothing out variations in the queue. So, an L4S AQM MUST signal congestion as soon as possible. Then, an L4S sender generally interprets CE marking as an unsmoothed signal.

This requirement does not prevent an L4S AQM from mixing in additional congestion signals that are smoothed, such as the signals from a Classic smoothed AQM that are coupled with unsmoothed L4S signals in the coupled DualQ [I-D.ietf-tsvwg-aqm-dualq-coupled]. But only as long as the onset of congestion can be signalled immediately, and can be interpreted by the sender as if it has been signalled immediately, which is important for interoperability

5.3. Exception for L4S Packet Identification by Network Nodes with Transport-Layer Awareness

To implement L4S packet classification, a network node does not need to identify transport-layer flows. Nonetheless, if an L4S network node classifies packets by their transport-layer flow ID and their ECN field, and if all the ECT packets in a flow have been ECT(0), the node MAY classify any CE packets in the same flow as if they were Classic ECT(0) packets. In all other cases, a network node MUST classify all CE packets as if they were ECT(1) packets. Examples of such other cases are: i) if no ECT packets have yet been identified in a flow; ii) if it is not desirable for a network node to identify transport-layer flows; or iii) if some ECT packets in a flow have been ECT(1) (this advice will need to be verified as part of L4S experiments).

5.4. Interaction of the L4S Identifier with other Identifiers

The examples in this section concern how additional identifiers might complement the L4S identifier to classify packets between class-based queues. Firstly Section 5.4.1 considers two queues, L4S and Classic, as in the Coupled DualQ AQM [I-D.ietf-tsvwg-aqm-dualq-coupled], either alone (Section 5.4.1.1) or within a larger queuing hierarchy (Section 5.4.1.2). Then Section 5.4.2 considers schemes that might combine per-flow 5-tuples with other identifiers.

5.4.1. DualQ Examples of Other Identifiers Complementing L4S Identifiers

5.4.1.1. Inclusion of Additional Traffic with L4S

In a typical case for the public Internet a network element that implements L4S in a shared queue might want to classify some low-rate but unresponsive traffic (e.g. DNS, LDAP, NTP, voice, game sync packets) into the low latency queue to mix with L4S traffic. In this case it would not be appropriate to call the queue an L4S queue, because it is shared by L4S and non-L4S traffic. Instead it will be called the low latency or L queue. The L queue then offers two different treatments:

- * The L4S treatment, which is a combination of the L4S AQM treatment and a priority scheduling treatment;
- * The low latency treatment, which is solely the priority scheduling treatment, without ECN-marking by the AQM.

To identify packets for just the scheduling treatment, it would be inappropriate to use the L4S ECT(1) identifier, because such traffic is unresponsive to ECN marking. Examples of relevant non-ECN identifiers are:

- * address ranges of specific applications or hosts configured to be, or known to be, safe, e.g. hard-coded IoT devices sending low intensity traffic;
- * certain low data-volume applications or protocols (e.g. ARP, DNS);
- * specific Diffserv codepoints that indicate traffic with limited burstiness such as the EF (Expedited Forwarding [RFC3246]), Voice-Admit [RFC5865] or proposed NQB (Non-Queue-Building [I-D.ietf-tsvwg-nqb]) service classes or equivalent local-use DSCPs (see [I-D.briscoe-tsvwg-l4s-diffserv]).

In summary, a network element that implements L4S in a shared queue MAY classify additional types of packets into the L queue based on identifiers other than the ECN field, but the types SHOULD be 'safe' to mix with L4S traffic, where 'safe' is explained in Section 5.4.1.1.1.

A packet that carries one of these non-ECN identifiers to classify it into the L queue would not be subject to the L4S ECN marking treatment, unless it also carried an ECT(1) or CE codepoint. The specification of an L4S AQM MUST define the behaviour for packets with unexpected combinations of codepoints, e.g. a non-ECN-based classifier for the L queue, but ECT(0) in the ECN field (for examples see section 2.5.1.1 of the DualQ spec [I-D.ietf-tsvwg-aqm-dualq-coupled]).

For clarity, non-ECN identifiers, such as the examples itemized above, might be used by some network operators who believe they identify non-L4S traffic that would be safe to mix with L4S traffic. They are not alternative ways for a host to indicate that it is sending L4S packets. Only the ECT(1) ECN codepoint indicates to a network element that a host is sending L4S packets (and CE indicates that it could have originated as ECT(1)). Specifically ECT(1) indicates that the host claims its behaviour satisfies the prerequisite transport requirements in Section 4.

In order to include non-L4S packets in the L queue, a network node MUST NOT alter Not-ECT or ECT(0) in the IP-ECN field to an L4S identifier. This ensures that these codepoints survive for any potential use later on the network path.

5.4.1.1.1. 'Safe' Unresponsive Traffic

The above section requires unresponsive traffic to be 'safe' to mix with L4S traffic. Ideally this means that the sender never sends any sequence of packets at a rate that exceeds the available capacity of the bottleneck link. However, typically an unresponsive transport does not even know the bottleneck capacity of the path, let alone its available capacity. Nonetheless, an application can be considered safe enough if it paces packets out (not necessarily completely regularly) such that its maximum instantaneous rate from packet to packet stays well below a typical broadband access rate.

This is a vague but useful definition, because many low latency applications of interest, such as DNS, voice, game sync packets, RPC, ACKs, keep-alives, could match this description.

Low rate streams such as voice and game sync packets, might not use continuously adapting ECN-based congestion control, but they ought to at least use a 'circuit-breaker' style of congestion response [RFC8083]. If the volume of traffic from unresponsive applications is high enough to overload the link, this will at least protect the capacity available to responsive applications. However, queuing delay in the L queue will probably rise to that controlled by the Classic (drop-based) AQM. If a network operator considers that such self-restraint is not enough, it might want to police the L queue (see Section 8.2 of the L4S architecture [I-D.ietf-tsvwg-l4s-arch]).

5.4.1.2. Exclusion of Traffic From L4S Treatment

To extend the above example, an operator might want to exclude some traffic from the L4S treatment for a policy reason, e.g. security (traffic from malicious sources) or commercial (e.g. initially the operator may wish to confine the benefits of L4S to business customers).

In this exclusion case, the classifier MUST classify on the relevant locally-used identifiers (e.g. source addresses) before classifying the non-matching traffic on the end-to-end L4S ECN identifier.

A network node MUST NOT alter the end-to-end L4S ECN identifier from L4S to Classic, because an operator decision to exclude certain traffic from L4S treatment is local-only. The end-to-end L4S

identifier then survives for other operators to use, or indeed, they can apply their own policy, independently based on their own choice of locally-used identifiers. This approach also allows any operator to remove its locally-applied exclusions in future, e.g. if it wishes to widen the benefit of the L4S treatment to all its customers.

A network node that supports L4S but excludes certain packets carrying the L4S identifier from L4S treatment **MUST** still apply marking or dropping that is compatible with an L4S congestion response. For instance, it could either drop such packets with the same likelihood as Classic packets or it could ECN-mark them with a likelihood appropriate to L4S traffic (e.g. the coupled probability in a DualQ coupled AQM) but aiming for the Classic delay target. It **MUST NOT** ECN-mark such packets with a Classic marking probability, which could confuse the sender.

5.4.1.3. Generalized Combination of L4S and Other Identifiers

L4S concerns low latency, which it can provide for all traffic without differentiation and without necessarily affecting bandwidth allocation. Diffserv provides for differentiation of both bandwidth and low latency, but its control of latency depends on its control of bandwidth. The two can be combined if a network operator wants to control bandwidth allocation but it also wants to provide low latency - for any amount of traffic within one of these allocations of bandwidth (rather than only providing low latency by limiting bandwidth) [I-D.briscoe-tsvwg-l4s-diffserv].

The DualQ examples so far have been framed in the context of providing the default Best Efforts Per-Hop Behaviour (PHB) using two queues - a Low Latency (L) queue and a Classic (C) Queue. This single DualQ structure is expected to be the most common and useful arrangement. But, more generally, an operator might choose to control bandwidth allocation through a hierarchy of Diffserv PHBs at a node, and to offer one (or more) of these PHBs using a pair of queues for a low latency and a Classic variant of the PHB.

In the first case, if we assume that a network element provides no PHBs except the DualQ, if a packet carries ECT(1) or CE, the network element would classify it for the L4S treatment irrespective of its DSCP. And, if a packet carried (say) the EF DSCP, the network element could classify it into the L queue irrespective of its ECN codepoint. However, where the DualQ is in a hierarchy of other PHBs, the classifier would classify some traffic into other PHBs based on DSCP before classifying between the low latency and Classic queues (based on ECT(1), CE and perhaps also the EF DSCP or other identifiers as in the above example).

[I-D.briscoe-tsvwg-l4s-diffserv] gives a number of examples of such arrangements to address various requirements.

[I-D.briscoe-tsvwg-l4s-diffserv] describes how an operator might use L4S to offer low latency as well as using Diffserv for bandwidth differentiation. It identifies two main types of approach, which can be combined: the operator might split certain Diffserv PHBs between L4S and a corresponding Classic service. Or it might split the L4S and/or the Classic service into multiple Diffserv PHBs. In either of these cases, a packet would have to be classified on its Diffserv and ECN codepoints.

In summary, there are numerous ways in which the L4S ECN identifier (ECT(1) and CE) could be combined with other identifiers to achieve particular objectives. The following categorization articulates those that are valid, but it is not necessarily exhaustive. Those tagged 'Recommended-standard-use' could be set by the sending host or a network. Those tagged 'Local-use' would only be set by a network:

1. Identifiers Complementing the L4S Identifier
 - a. Including More Traffic in the L Queue
(Could use Recommended-standard-use or Local-use identifiers)
 - b. Excluding Certain Traffic from the L Queue
(Local-use only)
2. Identifiers to place L4S classification in a PHB Hierarchy
(Could use Recommended-standard-use or Local-use identifiers)
 - a. PHBs Before L4S ECN Classification
 - b. PHBs After L4S ECN Classification

5.4.2. Per-Flow Queuing Examples of Other Identifiers Complementing L4S Identifiers

At a node with per-flow queueing (e.g. FQ-CoDel [RFC8290]), the L4S identifier could complement the Layer-4 flow ID as a further level of flow granularity (i.e. Not-ECT and ECT(0) queued separately from ECT(1) and CE packets). "Risk of reordering Classic CE packets" in Appendix B discusses the resulting ambiguity if packets originally marked ECT(0) are marked CE by an upstream AQM before they arrive at a node that classifies CE as L4S. It argues that the risk of reordering is vanishingly small and the consequence of such a low level of reordering is minimal.

Alternatively, it could be assumed that it is not in a flow's own interest to mix Classic and L4S identifiers. Then the AQM could use the ECN field to switch itself between a Classic and an L4S AQM behaviour within one per-flow queue. For instance, for ECN-capable packets, the AQM might consist of a simple marking threshold and an L4S ECN identifier might simply select a shallower threshold than a Classic ECN identifier would.

5.5. Limiting Packet Bursts from Links

As well as senders needing to limit packet bursts (Section 4.3), links need to limit the degree of burstiness they introduce. In both cases (senders and links) this is a tradeoff, because batch-handling of packets is done for good reason, e.g. processing efficiency or to make efficient use of medium acquisition delay. Some take the attitude that there is no point reducing burst delay at the sender below that introduced by links (or vice versa). However, delay reduction proceeds by cutting down 'the longest pole in the tent', which turns the spotlight on the next longest, and so on.

This document does not set any quantified requirements for links to limit burst delay, primarily because link technologies are outside the remit of L4S specifications. Nonetheless, the following two subsections outline opportunities for addressing bursty links in the process of L4S implementation and deployment.

5.5.1. Limiting Packet Bursts from Links Fed by an L4S AQM

It would not make sense to implement an L4S AQM that feeds into a particular link technology without also reviewing opportunities to reduce any form of burst delay introduced by that link technology. This would at least limit the bursts that the link would otherwise introduce into the onward traffic, which would cause jumpy feedback to the sender as well as potential extra queuing delay downstream. This document does not presume to even give guidance on an

appropriate target for such burst delay until there is more industry experience of L4S. However, as suggested in Section 4.3 it would not seem necessary to limit bursts lower than roughly 10% of the minimum base RTT expected in the typical deployment scenario (e.g. 250 us burst duration for links within the public Internet).

5.5.2. Limiting Packet Bursts from Links Upstream of an L4S AQM

The initial scope of the L4S experiment is to deploy L4S AQMs at bottlenecks and L4S congestion controls at senders. This is expected to highlight interactions with the most bursty upstream links and lead operators to tune down the burstiness of those links in their network that are configurable, or failing that, to have to compromise on the delay target of some L4S AQMs. It might also require specific redesign work relevant to the most problematic link types. Such knock-on effects of initial L4S deployment would all be part of the learning from the L4S experiment.

The details of such link changes are beyond the scope of the present document. Nonetheless, where L4S technology is being implemented on an outgoing interface of a device, it would make sense to consider opportunities for reducing bursts arriving at other incoming interface(s). For instance, where an L4S AQM is implemented to feed into the upstream WAN interface of a home gateway, there would be opportunities to alter the WiFi profiles sent out of any WiFi interfaces from the same device, in order to mitigate incoming bursts of aggregated WiFi frames from other WiFi stations.

6. Behaviour of Tunnels and Encapsulations

6.1. No Change to ECN Tunnels and Encapsulations in General

The L4S identifier is expected to work through and within any tunnel without modification, as long as the tunnel propagates the ECN field in any of the ways that have been defined since the first variant in the year 2001 [RFC3168]. L4S will also work with (but does not rely on) any of the more recent updates to ECN propagation in [RFC4301], [RFC6040] or [I-D.ietf-tsvwg-rfc6040update-shim]. However, it is likely that some tunnels still do not implement ECN propagation at all. In these cases, L4S will work through such tunnels, but within them the outer header of L4S traffic will appear as Classic.

AQMs are typically implemented where an IP-layer buffer feeds into a lower layer, so they are agnostic to link layer encapsulations. Where a bottleneck link is not IP-aware, the L4S identifier is still expected to work within any lower layer encapsulation without modification, as long as it propagates the ECN field as defined for the link technology, for example for MPLS [RFC5129] or

TRILL [I-D.ietf-trill-ecn-support]. In some of these cases, e.g. layer-3 Ethernet switches, the AQM accesses the IP layer header within the outer encapsulation, so again the L4S identifier is expected to work without modification. Nonetheless, the programme to define ECN for other lower layers is still in progress [I-D.ietf-tsvwg-ecn-encap-guidelines].

6.2. VPN Behaviour to Avoid Limitations of Anti-Replay

If a mix of L4S and Classic packets is sent into the same security association (SA) of a virtual private network (VPN), and if the VPN egress is employing the optional anti-replay feature, it could inappropriately discard Classic packets (or discard the records in Classic packets) by mistaking their greater queuing delay for a replay attack (see "Dropped Packets for Tunnels with Replay Protection Enabled" in [Heist21] for the potential performance impact). This known problem is common to both IPsec [RFC4301] and DTLS [RFC6347] VPNs, given they use similar anti-replay window mechanisms. The mechanism used can only check for replay within its window, so if the window is smaller than the degree of reordering, it can only assume there might be a replay attack and discard all the packets behind the trailing edge of the window. The specifications of IPsec AH [RFC4302] and ESP [RFC4303] suggest that an implementer scales the size of the anti-replay window with interface speed, and DTLS 1.3 [I-D.ietf-tls-dtls13] says "The receiver SHOULD pick a window large enough to handle any plausible reordering, which depends on the data rate." However, in practice, the size of a VPN's anti-replay window is not always scaled appropriately.

If a VPN carrying traffic participating in the L4S experiment experiences inappropriate replay detection, the foremost remedy would be to ensure that the egress is configured to comply with the above window-sizing requirements.

If an implementation of a VPN egress does not support a sufficiently large anti-replay window, e.g. due to hardware limitations, one of the temporary alternatives listed in order of preference below might be feasible instead:

- * If the VPN can be configured to classify packets into different SAs indexed by DSCP, apply the appropriate locally defined DSCPs to Classic and L4S packets. The DSCPs could be applied by the network (based on the least significant bit of the ECN field), or by the sending host. Such DSCPs would only need to survive as far as the VPN ingress.
- * If the above is not possible and it is necessary to use L4S, either of the following might be appropriate as a last resort:

- disable anti-replay protection at the VPN egress, after considering the security implications (optional anti-replay is mandatory in both IPsec and DTLS);
- configure the tunnel ingress not to propagate ECN to the outer, which would lose the benefits of L4S and Classic ECN over the VPN.

Modification to VPN implementations is outside the present scope, which is why this section has so far focused on reconfiguration. Although this document does not define any requirements for VPN implementations, determining whether there is a need for such requirements could be one aspect of L4S experimentation.

7. L4S Experiments

This section describes open questions that L4S Experiments ought to focus on. This section also documents outstanding open issues that will need to be investigated as part of L4S experimentation, given they could not be fully resolved during the WG phase. It also lists metrics that will need to be monitored during experiments (summarizing text elsewhere in L4S documents) and finally lists some potential future directions that researchers might wish to investigate.

In addition to this section, the DualQ spec [I-D.ietf-tsvwg-aqm-dualq-coupled] sets operational and management requirements for experiments with DualQ Coupled AQMs; and General operational and management requirements for experiments with L4S congestion controls are given in Section 4 and Section 5 above, e.g. co-existence and scaling requirements, incremental deployment arrangements.

The specification of each scalable congestion control will need to include protocol-specific requirements for configuration and monitoring performance during experiments. Appendix A of the guidelines in [RFC5706] provides a helpful checklist.

7.1. Open Questions

L4S experiments would be expected to answer the following questions:

- * Have all the parts of L4S been deployed, and if so, what proportion of paths support it?
 - What types of L4S AQMs were deployed, e.g. FQ, coupled DualQ, uncoupled DualQ, other? And how prevalent was each?

- Are the signalling patterns emitted by the deployed AQMs in any way different from those expected when the Prague requirements for endpoints were written?
- * Does use of L4S over the Internet result in significantly improved user experience?
- * Has L4S enabled novel interactive applications?
- * Did use of L4S over the Internet result in improvements to the following metrics:
 - queue delay (mean and 99th percentile) under various loads;
 - utilization;
 - starvation / fairness;
 - scaling range of flow rates and RTTs?
- * How dependent was the performance of L4S service on the bottleneck bandwidth or the path RTT?
- * How much do bursty links in the Internet affect L4S performance (see "Underutilization with Bursty Links" in [Heist21]) and how prevalent are they? How much limitation of burstiness from upstream links was needed and/or was realized - both at senders and at links, especially radio links or how much did L4S target delay have to be increased to accommodate the bursts (see bullet #7 in Section 4.3 and Section 5.5.2)?
- * Is the initial experiment with mis-marked bursty traffic at high RTT (see "Underutilization with Bursty Traffic" in [Heist21]) indicative of similar problems at lower RTTs and, if so, how effective is the suggested remedy in Appendix A.1 of the DualQ spec [I-D.ietf-tsvwg-aqm-dualq-coupled] (or possible other remedies)?
- * Was per-flow queue protection typically (un)necessary?
 - How well did overload protection or queue protection work?
- * How well did L4S flows coexist with Classic flows when sharing a bottleneck?
 - How frequently did problems arise?

- What caused any coexistence problems, and were any problems due to single-queue Classic ECN AQMs (this assumes single-queue Classic ECN AQMs can be distinguished from FQ ones)?
- * How prevalent were problems with the L4S service due to tunnels / encapsulations that do not support ECN decapsulation?
- * How easy was it to implement a fully compliant L4S congestion control, over various different transport protocols (TCP, QUIC, RMCAT, etc)?

Monitoring for harm to other traffic, specifically bandwidth starvation or excess queuing delay, will need to be conducted alongside all early L4S experiments. It is hard, if not impossible, for an individual flow to measure its impact on other traffic. So such monitoring will need to be conducted using bespoke monitoring across flows and/or across classes of traffic.

7.2. Open Issues

- * What is the best way forward to deal with L4S over single-queue Classic ECN AQM bottlenecks, given current problems with misdetecting L4S AQMs as Classic ECN AQMs? See the L4S operational guidance [I-D.ietf-tsvwg-l4sops].
- * Fixing the poor Interaction between current L4S congestion controls and CoDel with only Classic ECN support during flow startup. Originally, this was due to a bug in the initialization of the congestion EWMA in the Linux implementation of TCP Prague. That was quickly fixed, which removed the main performance impact, but further improvement would be useful (either by modifying CoDel, Scalable congestion controls, or both).

7.3. Future Potential

Researchers might find that L4S opens up the following interesting areas for investigation:

- * Potential for faster convergence time and tracking of available capacity;
- * Potential for improvements to particular link technologies, and cross-layer interactions with them;
- * Potential for using virtual queues, e.g. to further reduce latency jitter, or to leave headroom for capacity variation in radio networks;

- * Development and specification of reverse path congestion control using L4S building blocks (e.g. AccECN, QUIC);
- * Once queuing delay is cut down, what becomes the 'second longest pole in the tent' (other than the speed of light)?
- * Novel alternatives to the existing set of L4S AQMs;
- * Novel applications enabled by L4S.

8. IANA Considerations

The 01 codepoint of the ECN Field of the IP header is specified by the present Experimental RFC. The process for an experimental RFC to assign this codepoint in the IP header (v4 and v6) is documented in Proposed Standard [RFC8311], which updates the Proposed Standard [RFC3168].

When the present document is published as an RFC, IANA is asked to update the 01 entry in the registry, "ECN Field (Bits 6-7)" to the following (see <https://www.iana.org/assignments/dscp-registry/dscp-registry.xhtml#ecn-field>):

Binary	Keyword	References
01	ECT(1) (ECN-Capable Transport (1)) [1]	[RFC8311] [RFC Errata 5399] [RFCXXXX]

Table 1

[XXXX is the number that the RFC Editor assigns to the present document (this sentence to be removed by the RFC Editor)].

9. Security Considerations

Approaches to assure the integrity of signals using the new identifier are introduced in Appendix C.1. See the security considerations in the L4S architecture [I-D.ietf-tsvwg-l4s-arch] for further discussion of mis-use of the identifier, as well as extensive discussion of policing rate and latency in regard to L4S.

If the anti-replay window of a VPN egress is too small, it will mistake deliberate delay differences as a replay attack, and discard higher delay packets (e.g. Classic) carried within the same security association (SA) as low delay packets (e.g. L4S). Section 6.2 recommends that VPNs used in L4S experiments are configured with a sufficiently large anti-replay window, as required by the relevant specifications. It also discusses other alternatives.

If a user taking part in the L4S experiment sets up a VPN without being aware of the above advice, and if the user allows anyone to send traffic into their VPN, they would open up a DoS vulnerability in which an attacker could induce the VPN's anti-replay mechanism to discard enough of the user's Classic (C) traffic (if they are receiving any) to cause a significant rate reduction. While the user is actively downloading C traffic, the attacker sends C traffic into the VPN to fill the remainder of the bottleneck link, then sends intermittent L4S packets to maximize the chance of exceeding the VPN's replay window. The user can prevent this attack by following the recommendations in Section 6.2.

The recommendation to detect loss in time units prevents the ACK-splitting attacks described in [Savage-TCP].

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Appendix A. Rationale for the 'Prague L4S Requirements'

This appendix is informative, not normative. It gives a list of modifications to current scalable congestion controls so that they can be deployed over the public Internet and coexist safely with existing traffic. The list complements the normative requirements in Section 4 that a sender has to comply with before it can set the L4S identifier in packets it sends into the Internet. As well as rationale for safety improvements (the requirements in Section 4) this appendix also includes preferable performance improvements (optimizations).

The requirements and recommendations in Section 4) have become known as the Prague L4S Requirements, because they were originally identified at an ad hoc meeting during IETF-94 in Prague [TCPPrague]. They were originally called the 'TCP Prague Requirements', but they are not solely applicable to TCP, so the name and wording has been generalized for all transport protocols, and the name 'TCP Prague' is now used for a specific implementation of the requirements.

At the time of writing, DCTCP [RFC8257] is the most widely used scalable transport protocol. In its current form, DCTCP is specified to be deployable only in controlled environments. Deploying it in the public Internet would lead to a number of issues, both from the safety and the performance perspective. The modifications and additional mechanisms listed in this section will be necessary for its deployment over the global Internet. Where an example is needed, DCTCP is used as a base, but the requirements in Section 4 apply equally to other scalable congestion controls, covering adaptive real-time media, etc., not just capacity-seeking behaviours.

A.1. Rationale for the Requirements for Scalable Transport Protocols

A.1.1. Use of L4S Packet Identifier

Description: A scalable congestion control needs to distinguish the packets it sends from those sent by Classic congestion controls (see the precise normative requirement wording in Section 4.1).

Motivation: It needs to be possible for a network node to classify L4S packets without flow state into a queue that applies an L4S ECN marking behaviour and isolates L4S packets from the queuing delay of Classic packets.

A.1.2. Accurate ECN Feedback

Description: The transport protocol for a scalable congestion control needs to provide timely, accurate feedback about the extent of ECN marking experienced by all packets (see the precise normative requirement wording in Section 4.2).

Motivation: Classic congestion controls only need feedback about the existence of a congestion episode within a round trip, not precisely how many packets were marked with ECN or dropped. Therefore, in 2001, when ECN feedback was added to TCP [RFC3168], it could not inform the sender of more than one ECN mark per RTT. Since then, requirements for more accurate ECN feedback in TCP have been defined in [RFC7560] and [I-D.ietf-tcpm-accurate-ecn] specifies a change to the TCP protocol to satisfy these requirements. Most other transport protocols already satisfy this requirement (see Section 4.2).

A.1.3. Capable of Replacement by Classic Congestion Control

Description: It needs to be possible to replace the implementation of a scalable congestion control with a Classic control (see the precise normative requirement wording in Section 4.3).

Motivation: L4S is an experimental protocol, therefore it seems prudent to be able to disable it at source in case of insurmountable problems, perhaps due to some unexpected interaction on a particular sender; over a particular path or network; with a particular receiver or even ultimately an insurmountable problem with the experiment as a whole.

A.1.4. Fall back to Classic Congestion Control on Packet Loss

Description: As well as responding to ECN markings in a scalable way, a scalable congestion control needs to react to packet loss in a way that will coexist safely with a Reno congestion control [RFC5681] (see the precise normative requirement wording in Section 4.3).

Motivation: Part of the safety conditions for deploying a scalable congestion control on the public Internet is to make sure that it behaves properly when it builds a queue at a network bottleneck that has not been upgraded to support L4S. Packet loss can have many causes, but it usually has to be conservatively assumed that it is a sign of congestion. Therefore, on detecting packet loss, a scalable congestion control will need to fall back to Classic congestion control behaviour. If it does not comply, it could starve Classic traffic.

A scalable congestion control can be used for different types of transport, e.g. for real-time media or for reliable transport like TCP. Therefore, the particular Classic congestion control behaviour to fall back on will need to be dependent on the specific congestion control implementation. In the particular case of DCTCP, the DCTCP specification [RFC8257] states that "It is RECOMMENDED that an implementation deal with loss episodes in the same way as conventional TCP." For safe deployment, Section 4.3 requires any specification of a scalable congestion control for the public Internet to define the above requirement as a "MUST".

Even though a bottleneck is L4S capable, it might still become overloaded and have to drop packets. In this case, the sender may receive a high proportion of packets marked with the CE bit set and also experience loss. Current DCTCP implementations each react differently to this situation. At least one implementation reacts only to the drop signal (e.g. by halving the CWND) and at least another DCTCP implementation reacts to both signals (e.g. by halving the CWND due to the drop and also further reducing the CWND based on the proportion of marked packet). A third approach for the public Internet has been proposed that adjusts the loss response to result in a halving when combined with the ECN response. We believe that further experimentation is needed to understand what is the best behaviour for the public Internet, which may or not be one of these existing approaches.

A.1.5. Coexistence with Classic Congestion Control at Classic ECN bottlenecks

Description: Monitoring has to be in place so that a non-L4S but ECN-capable AQM can be detected at path bottlenecks. This is in case such an AQM has been implemented in a shared queue, in which case any long-running scalable flow would predominate over any simultaneous long-running Classic flow sharing the queue. The precise requirement wording in Section 4.3 is written so that such a problem could either be resolved in real-time, or via administrative intervention.

Motivation: Similarly to the discussion in Appendix A.1.4, this requirement in Section 4.3 is a safety condition to ensure an L4S congestion control coexists well with Classic flows when it builds a queue at a shared network bottleneck that has not been upgraded to support L4S. Nonetheless, if necessary, it is considered reasonable to resolve such problems over management timescales (possibly involving human intervention) because:

- * although a Classic flow can considerably reduce its throughput in the face of a competing scalable flow, it still makes progress and does not starve;
- * implementations of a Classic ECN AQM in a queue that is intended to be shared are believed to be rare;
- * detection of such AQMs is not always clear-cut; so focused out-of-band testing (or even contacting the relevant network operator) would improve certainty.

Therefore, the relevant normative requirement (Section 4.3) is divided into three stages: monitoring, detection and action:

Monitoring: Monitoring involves collection of the measurement data to be analysed. Monitoring is expressed as a 'MUST' for uncontrolled environments, although the placement of the monitoring function is left open. Whether monitoring has to be applied in real-time is expressed as a 'SHOULD'. This allows for the possibility that the operator of an L4S sender (e.g. a CDN) might prefer to test out-of-band for signs of Classic ECN AQMs, perhaps to avoid continually consuming resources to monitor live traffic.

Detection: Detection involves analysis of the monitored data to detect the likelihood of a Classic ECN AQM. Detection can either directly detect actual coexistence problems between flows, or it can aim to identify AQM technologies that are likely to present coexistence problems, based on knowledge of AQMs deployed at the

time. The requirements recommend that detection occurs live in real-time. However, detection is allowed to be deferred (e.g. it might involve further testing targeted at candidate AQMs);

Action: This involves the act of switching the sender to a Classic congestion control. This might occur in real-time within the congestion control for the subsequent duration of a flow, or it might involve administrative action to switch to Classic congestion control for a specific interface or for a certain set of destination addresses.

Instead of the sender taking action itself, the operator of the sender (e.g. a CDN) might prefer to ask the network operator to modify the Classic AQM's treatment of L4S packets; or to ensure L4S packets bypass the AQM; or to upgrade the AQM to support L4S (see the L4S operational guidance [I-D.ietf-tsvwg-l4sops]). Once L4S flows no longer shared the Classic ECN AQM they would obviously no longer detect it, and the requirement to act on it would no longer apply.

The whole set of normative requirements concerning Classic ECN AQMs in Section 4.3 is worded so that it does not apply in controlled environments, such as private networks or data centre networks. CDN servers placed within an access ISP's network can be considered as a single controlled environment, but any onward networks served by the access network, including all the attached customer networks, would be unlikely to fall under the same degree of coordinated control. Monitoring is expressed as a 'MUST' for these uncontrolled segments of paths (e.g. beyond the access ISP in a home network), because there is a possibility that there might be a shared queue Classic ECN AQM in that segment. Nonetheless, the intent of the wording is to only require occasional monitoring of these uncontrolled regions, and not to burden CDN operators if monitoring never uncovers any potential problems.

More detailed discussion of all the above options and alternatives can be found in the L4S operational guidance [I-D.ietf-tsvwg-l4sops].

Having said all the above, the approach recommended in Section 4.3 is to monitor, detect and act in real-time on live traffic. A passive monitoring algorithm to detect a Classic ECN AQM at the bottleneck and fall back to Classic congestion control is described in an extensive technical report [ecn-fallback], which also provides a link to Linux source code, and a large online visualization of its evaluation results. Very briefly, the algorithm primarily monitors RTT variation using the same algorithm that maintains the mean deviation of TCP's smoothed RTT, but it smooths over a duration of the order of a Classic sawtooth. The outcome is also conditioned on

other metrics such as the presence of CE marking and congestion avoidance phase having stabilized. The report also identifies further work to improve the approach, for instance improvements with low capacity links and combining the measurements with a cache of what had been learned about a path in previous connections. The report also suggests alternative approaches.

Although using passive measurements within live traffic (as above) can detect a Classic ECN AQM, it is much harder (perhaps impossible) to determine whether or not the AQM is in a shared queue. Nonetheless, this is much easier using active test traffic out-of-band, because two flows can be used. Section 4 of the same report [ecn-fallback] describes a simple technique to detect a Classic ECN AQM and determine whether it is in a shared queue, summarized here.

An L4S-enabled test server could be set up so that, when a test client accesses it, it serves a script that gets the client to open two parallel long-running flows. It could serve one with a Classic congestion control (C, that sets ECT(0)) and one with a scalable CC (L, that sets ECT(1)). If neither flow induces any ECN marks, it can be presumed the path does not contain a Classic ECN AQM. If either flow induces some ECN marks, the server could measure the relative flow rates and round trip times of the two flows. Table 2 shows the AQM that can be inferred for various cases (presuming the AQM behaviours known at the time of writing).

Rate	RTT	Inferred AQM
L > C	L = C	Classic ECN AQM (FIFO)
L = C	L = C	Classic ECN AQM (FQ)
L = C	L < C	FQ-L4S AQM
L ~ C	L < C	Coupled DualQ AQM

Table 2: Out-of-band testing with two parallel flows. L:=L4S, C:=Classic.

Finally, we motivate the recommendation in Section 4.3 that a scalable congestion control is not expected to change to setting ECT(0) while it adapts its behaviour to coexist with Classic flows. This is because the sender needs to continue to check whether it made the right decision – and switch back if it was wrong, or if a different link becomes the bottleneck:

- * If, as recommended, the sender changes only its behaviour but not its codepoint to Classic, its codepoint will still be compatible with either an L4S or a Classic AQM. If the bottleneck does actually support both, it will still classify ECT(1) into the same L4S queue, where the sender can measure that switching to Classic behaviour was wrong, so that it can switch back.
- * In contrast, if the sender changes both its behaviour and its codepoint to Classic, even if the bottleneck supports both, it will classify ECT(0) into the Classic queue, reinforcing the sender's incorrect decision so that it never switches back.
- * Also, not changing codepoint avoids the risk of being flipped to a different path by a load balancer or multipath routing that hashes on the whole of the ex-ToS byte (unfortunately still a common pathology).

Note that if a flow is configured to only use a Classic congestion control, it is then entirely appropriate not to use ECT(1).

A.1.6. Reduce RTT dependence

Description: A scalable congestion control needs to reduce RTT bias as much as possible at least over the low to typical range of RTTs that will interact in the intended deployment scenario (see the precise normative requirement wording in Section 4.3).

Motivation: The throughput of Classic congestion controls is known to be inversely proportional to RTT, so one would expect flows over very low RTT paths to nearly starve flows over larger RTTs. However, Classic congestion controls have never allowed a very low RTT path to exist because they induce a large queue. For instance, consider two paths with base RTT 1 ms and 100 ms. If a Classic congestion control induces a 100 ms queue, it turns these RTTs into 101 ms and 200 ms leading to a throughput ratio of about 2:1. Whereas if a scalable congestion control induces only a 1 ms queue, the ratio is 2:101, leading to a throughput ratio of about 50:1.

Therefore, with very small queues, long RTT flows will essentially starve, unless scalable congestion controls comply with this requirement in Section 4.3.

The RTT bias in current Classic congestion controls works satisfactorily when the RTT is higher than typical, and L4S does not change that. So, there is no additional requirement in Section 4.3 for high RTT L4S flows to remove RTT bias - they can but they don't have to.

A.1.7. Scaling down to fractional congestion windows

Description: A scalable congestion control needs to remain responsive to congestion when typical RTTs over the public Internet are significantly smaller because they are no longer inflated by queuing delay (see the precise normative requirement wording in Section 4.3).

Motivation: As currently specified, the minimum congestion window of ECN-capable TCP (and its derivatives) is expected to be 2 sender maximum segment sizes (SMSS), or 1 SMSS after a retransmission timeout. Once the congestion window reaches this minimum, if there is further ECN-marking, TCP is meant to wait for a retransmission timeout before sending another segment (see section 6.1.2 of the ECN spec [RFC3168]). In practice, most known window-based congestion control algorithms become unresponsive to ECN congestion signals at this point. No matter how much ECN marking, the congestion window no longer reduces. Instead, the sender's lack of any further congestion response forces the queue to grow, overriding any AQM and increasing queuing delay (making the window large enough to become responsive again). This can result in a stable but deeper queue, or it might drive the queue to loss, then the retransmission timeout mechanism acts as a backstop.

Most window-based congestion controls for other transport protocols have a similar minimum window, albeit when measured in bytes for those that use smaller packets.

L4S mechanisms significantly reduce queuing delay so, over the same path, the RTT becomes lower. Then this problem becomes surprisingly common [sub-mss-prob]. This is because, for the same link capacity, smaller RTT implies a smaller window. For instance, consider a residential setting with an upstream broadband Internet access of 8 Mb/s, assuming a max segment size of 1500 B. Two upstream flows will each have the minimum window of 2 SMSS if the RTT is 6 ms or less, which is quite common when accessing a nearby data centre. So, any more than two such parallel TCP flows will become unresponsive to ECN and increase queuing delay.

Unless scalable congestion controls address the requirement in Section 4.3 from the start, they will frequently become unresponsive to ECN, negating the low latency benefit of L4S, for themselves and for others.

That would seem to imply that scalable congestion controllers ought to be required to be able work with a congestion window less than 1 SMSS. For instance, if an ECN-capable TCP gets an ECN-mark when it is already sitting at a window of 1 SMSS, RFC 3168 requires it to defer sending for a retransmission timeout. A less drastic but more

complex mechanism can maintain a congestion window less than 1 SMSS (significantly less if necessary), as described in [Ahmed19]. Other approaches are likely to be feasible.

However, the requirement in Section 4.3 is worded as a "SHOULD" because it is believed that the existence of a minimum window is not all bad. When competing with an unresponsive flow, a minimum window naturally protects the flow from starvation by at least keeping some data flowing.

By stating the requirement to go lower than 1 SMSS as a "SHOULD", while the requirement in RFC 3168 still stands as well, we shall be able to watch the choices of minimum window evolve in different scalable congestion controllers.

A.1.8. Measuring Reordering Tolerance in Time Units

Description: When detecting loss, a scalable congestion control needs to be tolerant to reordering over an adaptive time interval, which scales with throughput, rather than counting only in fixed units of packets, which does not scale (see the precise normative requirement wording in Section 4.3).

Motivation: A primary purpose of L4S is scalable throughput (it's in the name). Scalability in all dimensions is, of course, also a goal of all IETF technology. The inverse linear congestion response in Section 4.3 is necessary, but not sufficient, to solve the congestion control scalability problem identified in [RFC3649]. As well as maintaining frequent ECN signals as rate scales, it is also important to ensure that a potentially false perception of loss does not limit throughput scaling.

End-systems cannot know whether a missing packet is due to loss or reordering, except in hindsight - if it appears later. So they can only deem that there has been a loss if a gap in the sequence space has not been filled, either after a certain number of subsequent packets has arrived (e.g. the 3 DupACK rule of standard TCP congestion control [RFC5681]) or after a certain amount of time (e.g. the RACK approach [RFC8985]).

As we attempt to scale packet rate over the years:

- * Even if only some sending hosts still deem that loss has occurred by counting reordered packets, all networks will have to keep reducing the time over which they keep packets in order. If some link technologies keep the time within which reordering occurs roughly unchanged, then loss over these links, as perceived by these hosts, will appear to continually rise over the years.

- * In contrast, if all senders detect loss in units of time, the time over which the network has to keep packets in order stays roughly invariant.

Therefore hosts have an incentive to detect loss in time units (so as not to fool themselves too often into detecting losses when there are none). And for hosts that are changing their congestion control implementation to L4S, there is no downside to including time-based loss detection code in the change (loss recovery implemented in hardware is an exception, covered later). Therefore requiring L4S hosts to detect loss in time-based units would not be a burden.

If the requirement in Section 4.3 were not placed on L4S hosts, even though it would be no burden on hosts to comply, all networks would face unnecessary uncertainty over whether some L4S hosts might be detecting loss by counting packets. Then all link technologies will have to unnecessarily keep reducing the time within which reordering occurs. That is not a problem for some link technologies, but it becomes increasingly challenging for other link technologies to continue to scale, particularly those relying on channel bonding for scaling, such as LTE, 5G and DOCSIS.

Given Internet paths traverse many link technologies, any scaling limit for these more challenging access link technologies would become a scaling limit for the Internet as a whole.

It might be asked how it helps to place this loss detection requirement only on L4S hosts, because networks will still face uncertainty over whether non-L4S flows are detecting loss by counting DupACKs. The answer is that those link technologies for which it is challenging to keep squeezing the reordering time will only need to do so for non-L4S traffic (which they can do because the L4S identifier is visible at the IP layer). Therefore, they can focus their processing and memory resources into scaling non-L4S (Classic) traffic. Then, the higher the proportion of L4S traffic, the less of a scaling challenge they will have.

To summarize, there is no reason for L4S hosts not to be part of the solution instead of part of the problem.

Requirement ("MUST") or recommendation ("SHOULD")? As explained above, this is a subtle interoperability issue between hosts and networks, which seems to need a "MUST". Unless networks can be certain that all L4S hosts follow the time-based approach, they still have to cater for the worst case - continually squeeze reordering into a smaller and smaller duration - just for hosts that might be using the counting approach. However, it was decided to express this as a recommendation, using "SHOULD". The main justification was that networks can still be fairly certain that L4S hosts will follow this recommendation, because following it offers only gain and no pain.

Details:

The speed of loss recovery is much more significant for short flows than long, therefore a good compromise is to adapt the reordering window; from a small fraction of the RTT at the start of a flow, to a larger fraction of the RTT for flows that continue for many round trips.

This is broadly the approach adopted by TCP RACK (Recent ACKnowledgements) [RFC8985]. However, RACK starts with the 3 DupACK approach, because the RTT estimate is not necessarily stable. As long as the initial window is paced, such initial use of 3 DupACK counting would amount to time-based loss detection and therefore would satisfy the time-based loss detection recommendation of Section 4.3. This is because pacing of the initial window would ensure that 3 DupACKs early in the connection would be spread over a small fraction of the round trip.

As mentioned above, hardware implementations of loss recovery using DupACK counting exist (e.g. some implementations of RoCEv2 for RDMA). For low latency, these implementations can change their congestion control to implement L4S, because the congestion control (as distinct from loss recovery) is implemented in software. But they cannot easily satisfy this loss recovery requirement. However, it is believed they do not need to, because such implementations are believed to solely exist in controlled environments, where the network technology keeps reordering extremely low anyway. This is why controlled environments with hardly any reordering are excluded from the scope of the normative recommendation in Section 4.3.

Detecting loss in time units also prevents the ACK-splitting attacks described in [Savage-TCP].

A.2. Scalable Transport Protocol Optimizations

A.2.1. Setting ECT in Control Packets and Retransmissions

Description: This item concerns TCP and its derivatives (e.g. SCTP) as well as RTP/RTCP [RFC6679]. The original specification of ECN for TCP precluded the use of ECN on control packets and retransmissions. Similarly RFC 6679 precludes the use of ECT on RTCP datagrams, in case the path changes after it has been checked for ECN traversal. To improve performance, scalable transport protocols ought to enable ECN at the IP layer in TCP control packets (SYN, SYN-ACK, pure ACKs, etc.) and in retransmitted packets. The same is true for other transports, e.g. SCTP, RTCP.

Motivation (TCP): RFC 3168 prohibits the use of ECN on these types of TCP packet, based on a number of arguments. This means these packets are not protected from congestion loss by ECN, which considerably harms performance, particularly for short flows. ECN++ [I-D.ietf-tcpm-generalized-ecn] proposes experimental use of ECN on all types of TCP packet as long as AccECN feedback [I-D.ietf-tcpm-accurate-ecn] is available (which itself satisfies the accurate feedback requirement in Section 4.2 for using a scalable congestion control).

Motivation (RTCP): L4S experiments in general will need to observe the rule in the RTP ECN spec [RFC6679] that precludes ECT on RTCP datagrams. Nonetheless, as ECN usage becomes more widespread, it would be useful to conduct specific experiments with ECN-capable RTCP to gather data on whether such caution is necessary.

A.2.2. Faster than Additive Increase

Description: It would improve performance if scalable congestion controls did not limit their congestion window increase to the standard additive increase of 1 SMSS per round trip [RFC5681] during congestion avoidance. The same is true for derivatives of TCP congestion control, including similar approaches used for real-time media.

Motivation: As currently defined [RFC8257], DCTCP uses the traditional Reno additive increase in congestion avoidance phase. When the available capacity suddenly increases (e.g. when another flow finishes, or if radio capacity increases) it can take very many round trips to take advantage of the new capacity. TCP Cubic [RFC8312] was designed to solve this problem, but as flow rates have continued to increase, the delay accelerating into available capacity has become prohibitive. See, for instance, the examples in

Section 5.1 of the L4S architecture [I-D.ietf-tsvwg-l4s-arch]. Even when out of its Reno-compatibility mode, every 8x scaling of Cubic's flow rate leads to 2x more acceleration delay.

In the steady state, DCTCP induces about 2 ECN marks per round trip, so it is possible to quickly detect when these signals have disappeared and seek available capacity more rapidly, while minimizing the impact on other flows (Classic and scalable) [LinuxPacedChirping]. Alternatively, approaches such as Adaptive Acceleration (A2DTCP [A2DTCP]) have been proposed to address this problem in data centres, which might be deployable over the public Internet.

A.2.3. Faster Convergence at Flow Start

Description: It would improve performance if scalable congestion controls converged (reached their steady-state share of the capacity) faster than Classic congestion controls or at least no slower. This affects the flow start behaviour of any L4S congestion control derived from a Classic transport that uses TCP slow start, including those for real-time media.

Motivation: As an example, a new DCTCP flow takes longer than a Classic congestion control to obtain its share of the capacity of the bottleneck when there are already ongoing flows using the bottleneck capacity. In a data centre environment DCTCP takes about a factor of 1.5 to 2 longer to converge due to the much higher typical level of ECN marking that DCTCP background traffic induces, which causes new flows to exit slow start early [Alizadeh-stability]. In testing for use over the public Internet the convergence time of DCTCP relative to a regular loss-based TCP slow start is even less favourable [Paced-Chirping] due to the shallow ECN marking threshold needed for L4S. It is exacerbated by the typically greater mismatch between the link rate of the sending host and typical Internet access bottlenecks. This problem is detrimental in general, but would particularly harm the performance of short flows relative to Classic congestion controls.

Appendix B. Compromises in the Choice of L4S Identifier

This appendix is informative, not normative. As explained in Section 2, there is insufficient space in the IP header (v4 or v6) to fully accommodate every requirement. So the choice of L4S identifier involves tradeoffs. This appendix records the pros and cons of the choice that was made.

Non-normative recap of the chosen codepoint scheme:

Packets with ECT(1) and conditionally packets with CE signify L4S semantics as an alternative to the semantics of Classic ECN [RFC3168], specifically:

- The ECT(1) codepoint signifies that the packet was sent by an L4S-capable sender.
- Given shortage of codepoints, both L4S and Classic ECN sides of an AQM have to use the same CE codepoint to indicate that a packet has experienced congestion. If a packet that had already been marked CE in an upstream buffer arrived at a subsequent AQM, this AQM would then have to guess whether to classify CE packets as L4S or Classic ECN. Choosing the L4S treatment is a safer choice, because then a few Classic packets might arrive early, rather than a few L4S packets arriving late.
- Additional information might be available if the classifier were transport-aware. Then it could classify a CE packet for Classic ECN treatment if the most recent ECT packet in the same flow had been marked ECT(0). However, the L4S service ought not to need transport-layer awareness.

Cons:

Consumes the last ECN codepoint: The L4S service could potentially supersede the service provided by Classic ECN, therefore using ECT(1) to identify L4S packets could ultimately mean that the ECT(0) codepoint was 'wasted' purely to distinguish one form of ECN from its successor.

ECN hard in some lower layers: It is not always possible to support the equivalent of an IP-ECN field in an AQM acting in a buffer below the IP layer [I-D.ietf-tsvwg-ecn-encap-guidelines]. Then, depending on the lower layer scheme, the L4S service might have to drop rather than mark frames even though they might encapsulate an ECN-capable packet.

Risk of reordering Classic CE packets within a flow: Classifying all CE packets into the L4S queue risks any CE packets that were originally ECT(0) being incorrectly classified as L4S. If there were delay in the Classic queue, these incorrectly classified CE packets would arrive early, which is a form of reordering. Reordering within a microflow can cause TCP senders (and senders of similar transports) to retransmit spuriously. However, the risk of spurious retransmissions would be extremely low for the following reasons:

1. It is quite unusual to experience queuing at more than one bottleneck on the same path (the available capacities have to be identical).
2. In only a subset of these unusual cases would the first bottleneck support Classic ECN marking while the second supported L4S ECN marking, which would be the only scenario where some ECT(0) packets could be CE marked by an AQM supporting Classic ECN then the remainder experienced further delay through the Classic side of a subsequent L4S DualQ AQM.
3. Even then, when a few packets are delivered early, it takes very unusual conditions to cause a spurious retransmission, in contrast to when some packets are delivered late. The first bottleneck has to apply CE-marks to at least N contiguous packets and the second bottleneck has to inject an uninterrupted sequence of at least N of these packets between two packets earlier in the stream (where N is the reordering window that the transport protocol allows before it considers a packet is lost).

For example consider $N=3$, and consider the sequence of packets 100, 101, 102, 103,... and imagine that packets 150,151,152 from later in the flow are injected as follows: 100, 150, 151, 101, 152, 102, 103... If this were late reordering, even one packet arriving out of sequence would trigger a spurious retransmission, but there is no spurious retransmission here with early reordering, because packet 101 moves the cumulative ACK counter forward before 3 packets have arrived out of order. Later, when packets 148, 149, 153... arrive, even though there is a 3-packet hole, there will be no problem, because the packets to fill the hole are already in the receive buffer.

4. Even with the current TCP recommendation of $N=3$ [RFC5681] spurious retransmissions will be unlikely for all the above reasons. As RACK [RFC8985] is becoming widely deployed, it tends to adapt its reordering window to a larger value of N , which will make the chance of a contiguous sequence of N early arrivals vanishingly small.
5. Even a run of 2 CE marks within a Classic ECN flow is unlikely, given FQ-CoDel is the only known widely deployed AQM that supports Classic ECN marking and it takes great care to separate out flows and to space any markings evenly along each flow.

It is extremely unlikely that the above set of 5 eventualities that are each unusual in themselves would all happen simultaneously. But, even if they did, the consequences would hardly be dire: the odd spurious fast retransmission. Whenever the traffic source (a Classic congestion control) mistakes the reordering of a string of CE marks for a loss, one might think that it will reduce its congestion window as well as emitting a spurious retransmission. However, it would have already reduced its congestion window when the CE markings arrived early. If it is using ABE [RFC8511], it might reduce cwnd a little more for a loss than for a CE mark. But it will revert that reduction once it detects that the retransmission was spurious.

In conclusion, the impact of early reordering on spurious retransmissions due to CE being ambiguous will generally be vanishingly small.

Insufficient anti-replay window in some pre-existing VPNs: If delay is reduced for a subset of the flows within a VPN, the anti-replay feature of some VPNs is known to potentially mistake the difference in delay for a replay attack. Section 6.2 recommends that the anti-replay window at the VPN egress is sufficiently sized, as required by the relevant specifications. However, in some VPN implementations the maximum anti-replay window is insufficient to cater for a large delay difference at prevailing packet rates. Section 6.2 suggests alternative work-rounds for such cases, but end-users using L4S over a VPN will need to be able to recognize the symptoms of this problem, in order to seek out these work-rounds.

Hard to distinguish Classic ECN AQM: With this scheme, when a source receives ECN feedback, it is not explicitly clear which type of AQM generated the CE markings. This is not a problem for Classic ECN sources that send ECT(0) packets, because an L4S AQM will recognize the ECT(0) packets as Classic and apply the appropriate Classic ECN marking behaviour.

However, in the absence of explicit disambiguation of the CE markings, an L4S source needs to use heuristic techniques to work out which type of congestion response to apply (see Appendix A.1.5). Otherwise, if long-running Classic flow(s) are sharing a Classic ECN AQM bottleneck with long-running L4S flow(s), which then apply an L4S response to Classic CE signals, the L4S flows would outcompete the Classic flow(s). Experiments have shown that L4S flows can take about 20 times more capacity share than equivalent Classic flows. Nonetheless, as link capacity reduces (e.g. to 4 Mb/s), the inequality reduces. So Classic flows always make progress and are not starved.

When L4S was first proposed (in 2015, 14 years after the Classic ECN spec [RFC3168] was published), it was believed that Classic ECN AQMs had failed to be deployed, because research measurements had found little or no evidence of CE marking. In subsequent years Classic ECN was included in per-flow-queuing (FQ) deployments, however an FQ scheduler stops an L4S flow outcompeting Classic, because it enforces equality between flow rates. It is not known whether there have been any non-FQ deployments of Classic ECN AQMs in the subsequent years, or whether there will be in future.

An algorithm for detecting a Classic ECN AQM as soon as a flow stabilizes after start-up has been proposed [ecn-fallback] (see Appendix A.1.5 for a brief summary). Testbed evaluations of v2 of the algorithm have shown detection is reasonably good for Classic ECN AQMs, in a wide range of circumstances. However, although it can correctly detect an L4S ECN AQM in many circumstances, it is often incorrect at low link capacities and/or high RTTs. Although this is the safe way round, there is a danger that it will discourage use of the algorithm.

Non-L4S service for control packets: Solely for the case of TCP, the Classic ECN RFCs [RFC3168] and [RFC5562] require a sender to clear the ECN field to Not-ECT on retransmissions and on certain control packets specifically pure ACKs, window probes and SYNs. When L4S packets are classified by the ECN field, these TCP control packets would not be classified into an L4S queue, and could therefore be delayed relative to the other packets in the flow. This would not cause reordering (because retransmissions are already out of order, and these control packets typically carry no data). However, it would make critical TCP control packets more vulnerable to loss and delay. To address this problem, ECN++ [I-D.ietf-tcpm-generalized-ecn] proposes an experiment in which all TCP control packets and retransmissions are ECN-capable as long as appropriate ECN feedback is available in each case.

Pros:

Should work e2e: The ECN field generally propagates end-to-end across the Internet without being wiped or mangled, at least over fixed networks. Unlike the DSCP, the setting of the ECN field is at least meant to be forwarded unchanged by networks that do not support ECN.

Should work in tunnels: The L4S identifiers work across and within

any tunnel that propagates the ECN field in any of the variant ways it has been defined since ECN-tunneling was first specified in the year 2001 [RFC3168]. However, it is likely that some tunnels still do not implement ECN propagation at all.

Should work for many link technologies: At most, but not all, path bottlenecks there is IP-awareness, so that L4S AQMs can be located where the IP-ECN field can be manipulated. Bottlenecks at lower layer nodes without IP-awareness either have to use drop to signal congestion or a specific congestion notification facility has to be defined for that link technology, including propagation to and from IP-ECN. The programme to define these is progressing and in each case so far the scheme already defined for ECN inherently supports L4S as well (see Section 6.1).

Could migrate to one codepoint: If all Classic ECN senders eventually evolve to use the L4S service, the ECT(0) codepoint could be reused for some future purpose, but only once use of ECT(0) packets had reduced to zero, or near-zero, which might never happen.

L4 not required: Being based on the ECN field, this scheme does not need the network to access transport layer flow identifiers. Nonetheless, it does not preclude solutions that do.

Appendix C. Potential Competing Uses for the ECT(1) Codepoint

The ECT(1) codepoint of the ECN field has already been assigned once for the ECN nonce [RFC3540], which has now been categorized as historic [RFC8311]. ECN is probably the only remaining field in the Internet Protocol that is common to IPv4 and IPv6 and still has potential to work end-to-end, with tunnels and with lower layers. Therefore, ECT(1) should not be reassigned to a different experimental use (L4S) without carefully assessing competing potential uses. These fall into the following categories:

C.1. Integrity of Congestion Feedback

Receiving hosts can fool a sender into downloading faster by suppressing feedback of ECN marks (or of losses if retransmissions are not necessary or available otherwise).

The historic ECN nonce protocol [RFC3540] proposed that a TCP sender could set either of ECT(0) or ECT(1) in each packet of a flow and remember the sequence it had set. If any packet was lost or congestion marked, the receiver would miss that bit of the sequence. An ECN Nonce receiver had to feed back the least significant bit of the sum, so it could not suppress feedback of a loss or mark without a 50-50 chance of guessing the sum incorrectly.

It is highly unlikely that ECT(1) will be needed for integrity protection in future. The ECN Nonce RFC [RFC3540] has been reclassified as historic, partly because other ways have been developed to protect feedback integrity of TCP and other transports [RFC8311] that do not consume a codepoint in the IP header. For instance:

- * the sender can test the integrity of the receiver's feedback by occasionally setting the IP-ECN field to a value normally only set by the network. Then it can test whether the receiver's feedback faithfully reports what it expects (see para 2 of Section 20.2 of the ECN spec [RFC3168]). This works for loss and it will work for the accurate ECN feedback [RFC7560] intended for L4S.
- * A network can enforce a congestion response to its ECN markings (or packet losses) by auditing congestion exposure (ConEx) [RFC7713]. Whether the receiver or a downstream network is suppressing congestion feedback or the sender is unresponsive to the feedback, or both, ConEx audit can neutralise any advantage that any of these three parties would otherwise gain.
- * The TCP authentication option (TCP-AO [RFC5925]) can be used to detect any tampering with TCP congestion feedback (whether malicious or accidental). TCP's congestion feedback fields are immutable end-to-end, so they are amenable to TCP-AO protection, which covers the main TCP header and TCP options by default. However, TCP-AO is often too brittle to use on many end-to-end paths, where middleboxes can make verification fail in their attempts to improve performance or security, e.g. by resegmentation or shifting the sequence space.

C.2. Notification of Less Severe Congestion than CE

Various researchers have proposed to use ECT(1) as a less severe congestion notification than CE, particularly to enable flows to fill available capacity more quickly after an idle period, when another flow departs or when a flow starts, e.g. VCP [VCP], Queue View (QV) [QV].

Before assigning ECT(1) as an identifier for L4S, we must carefully consider whether it might be better to hold ECT(1) in reserve for future standardisation of rapid flow acceleration, which is an important and enduring problem [RFC6077].

Pre-Congestion Notification (PCN) is another scheme that assigns alternative semantics to the ECN field. It uses ECT(1) to signify a less severe level of pre-congestion notification than CE [RFC6660]. However, the ECN field only takes on the PCN semantics if packets carry a Diffserv codepoint defined to indicate PCN marking within a controlled environment. PCN is required to be applied solely to the outer header of a tunnel across the controlled region in order not to interfere with any end-to-end use of the ECN field. Therefore a PCN region on the path would not interfere with the L4S service identifier defined in Section 3.

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Forward Error Correction (FEC) Framework Extension to Sliding Window
Codes
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Abstract

RFC 6363 describes a framework for using Forward Error Correction (FEC) codes to provide protection against packet loss. The framework supports applying FEC to arbitrary packet flows over unreliable transport and is primarily intended for real-time, or streaming, media. However, FECFRAME as per RFC 6363 is restricted to block FEC codes. This document updates RFC 6363 to support FEC Codes based on a sliding encoding window, in addition to Block FEC Codes, in a backward-compatible way. During multicast/broadcast real-time content delivery, the use of sliding window codes significantly improves robustness in harsh environments, with less repair traffic and lower FEC-related added latency.

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1. Introduction

Many applications need to transport a continuous stream of packetized data from a source (sender) to one or more destinations (receivers) over networks that do not provide guaranteed packet delivery. In particular packets may be lost, which is strictly the focus of this document: we assume that transmitted packets are either lost (e.g., because of a congested router, of a poor signal-to-noise ratio in a wireless network, or because the number of bit errors exceeds the correction capabilities of the physical-layer error correcting code)

or received by the transport protocol without any corruption (i.e., the bit-errors, if any, have been fixed by the physical-layer error correcting code and therefore are hidden to the upper layers).

For these use-cases, Forward Error Correction (FEC) applied within the transport or application layer is an efficient technique to improve packet transmission robustness in presence of packet losses (or "erasures"), without going through packet retransmissions that create a delay often incompatible with real-time constraints. The FEC Building Block defined in [RFC5052] provides a framework for the definition of Content Delivery Protocols (CDPs) that make use of separately-defined FEC schemes. Any CDP defined according to the requirements of the FEC Building Block can then easily be used with any FEC Scheme that is also defined according to the requirements of the FEC Building Block.

Then FECFRAME [RFC6363] provides a framework to define Content Delivery Protocols (CDPs) that provide FEC protection for arbitrary packet flows over an unreliable datagram service transport such as UDP. It is primarily intended for real-time or streaming media applications, using broadcast, multicast, or on-demand delivery.

However, [RFC6363] only considers block FEC schemes defined in accordance with the FEC Building Block [RFC5052] (e.g., [RFC6681], [RFC6816] or [RFC6865]). These codes require the input flow(s) to be segmented into a sequence of blocks. Then FEC encoding (at a sender or an encoding middlebox) and decoding (at a receiver or a decoding middlebox) are both performed on a per-block basis. For instance, if the current block encompasses the 100's to 119's source symbols (i.e., a block of size 20 symbols) of an input flow, encoding (and decoding) will be performed on this block independently of other blocks. This approach has major impacts on FEC encoding and decoding delays. The data packets of continuous media flow(s) may be passed to the transport layer immediately, without delay. But the block creation time, that depends on the number of source symbols in this block, impacts both the FEC encoding delay (since encoding requires that all source symbols be known), and mechanically the packet loss recovery delay at a receiver (since no repair symbol for the current block can be generated and therefore received before that time). Therefore a good value for the block size is necessarily a balance between the maximum FEC decoding latency at the receivers (which must be in line with the most stringent real-time requirement of the protected flow(s), hence an incentive to reduce the block size), and the desired robustness against long loss bursts (which increases with the block size, hence an incentive to increase this size).

This document updates [RFC6363] in order to also support FEC codes based on a sliding encoding window (A.K.A. convolutional codes)

[RFC8406]. This encoding window, either of fixed or variable size, slides over the set of source symbols. FEC encoding is launched whenever needed, from the set of source symbols present in the sliding encoding window at that time. This approach significantly reduces FEC-related latency, since repair symbols can be generated and passed to the transport layer on-the-fly, at any time, and can be regularly received by receivers to quickly recover packet losses. Using sliding window FEC codes is therefore highly beneficial to real-time flows, one of the primary targets of FECFRAME. [RLC-ID] provides an example of such FEC Scheme for FECFRAME, built upon the simple sliding window Random Linear Codes (RLC).

This document is fully backward compatible with [RFC6363]. Indeed:

- o this FECFRAME update does not prevent nor compromise in any way the support of block FEC codes. Both types of codes can nicely co-exist, just like different block FEC schemes can co-exist;
- o each sliding window FEC Scheme is associated to a specific FEC Encoding ID subject to IANA registration, just like block FEC Schemes;
- o any receiver, for instance a legacy receiver that only supports block FEC schemes, can easily identify the FEC Scheme used in a FECFRAME session. Indeed, the FEC Encoding ID that identifies the FEC Scheme is carried in the FEC Framework Configuration Information (see section 5.5 of [RFC6363]). For instance, when the Session Description Protocol (SDP) is used to carry the FEC Framework Configuration Information, the FEC Encoding ID can be communicated in the "encoding-id=" parameter of a "fec-repair-flow" attribute [RFC6364]. This mechanism is the basic approach for a FECFRAME receiver to determine whether or not it supports the FEC Scheme used in a given FECFRAME session;

This document leverages on [RFC6363] and re-uses its structure. It proposes new sections specific to sliding window FEC codes whenever required. The only exception is Section 3 that provides a quick summary of FECFRAME in order to facilitate the understanding of this document to readers not familiar with the concepts and terminology.

2. Definitions and Abbreviations

The following list of definitions and abbreviations is copied from [RFC6363], adding only the Block/sliding window FEC Code and Encoding/Decoding Window definitions (tagged with "ADDED"):

Application Data Unit (ADU): The unit of source data provided as payload to the transport layer. For instance, it can be a

payload containing the result of the RTP packetization of a compressed video frame.

ADU Flow: A sequence of ADUs associated with a transport-layer flow identifier (such as the standard 5-tuple {source IP address, source port, destination IP address, destination port, transport protocol}).

AL-FEC: Application-layer Forward Error Correction.

Application Protocol: Control protocol used to establish and control the source flow being protected, e.g., the Real-Time Streaming Protocol (RTSP).

Content Delivery Protocol (CDP): A complete application protocol specification that, through the use of the framework defined in this document, is able to make use of FEC schemes to provide FEC capabilities.

FEC Code: An algorithm for encoding data such that the encoded data flow is resilient to data loss. Note that, in general, FEC codes may also be used to make a data flow resilient to corruption, but that is not considered in this document.

Block FEC Code: (ADDED) An FEC Code that operates on blocks, i.e., for which the input flow MUST be segmented into a sequence of blocks, FEC encoding and decoding being performed independently on a per-block basis.

Sliding Window FEC Code: (ADDED) An FEC Code that can generate repair symbols on-the-fly, at any time, from the set of source symbols present in the sliding encoding window at that time. These codes are also known as convolutional codes.

FEC Framework: A protocol framework for the definition of Content Delivery Protocols using FEC, such as the framework defined in this document.

FEC Framework Configuration Information: Information that controls the operation of the FEC Framework.

FEC Payload ID: Information that identifies the contents and provides positional information of a packet with respect to the FEC Scheme.

FEC Repair Packet: At a sender (respectively, at a receiver), a payload submitted to (respectively, received from) the transport

protocol containing one or more repair symbols along with a Repair FEC Payload ID and possibly an RTP header.

FEC Scheme: A specification that defines the additional protocol aspects required to use a particular FEC code with the FEC Framework.

FEC Source Packet: At a sender (respectively, at a receiver), a payload submitted to (respectively, received from) the transport protocol containing an ADU along with an optional Explicit Source FEC Payload ID.

Repair Flow: The packet flow carrying FEC data.

Repair FEC Payload ID: A FEC Payload ID specifically for use with repair packets.

Source Flow: The packet flow to which FEC protection is to be applied. A source flow consists of ADUs.

Source FEC Payload ID: A FEC Payload ID specifically for use with source packets.

Source Protocol: A protocol used for the source flow being protected, e.g., RTP.

Transport Protocol: The protocol used for the transport of the source and repair flows, using an unreliable datagram service such as UDP.

Encoding Window: (ADDED) Set of Source Symbols available at the sender/coding node that are used to generate a repair symbol, with a Sliding Window FEC Code.

Decoding Window: (ADDED) Set of received or decoded source and repair symbols available at a receiver that are used to decode erased source symbols, with a Sliding Window FEC Code.

Code Rate: The ratio between the number of source symbols and the number of encoding symbols. By definition, the code rate is such that $0 < \text{code rate} \leq 1$. A code rate close to 1 indicates that a small number of repair symbols have been produced during the encoding process.

Encoding Symbol: Unit of data generated by the encoding process. With systematic codes, source symbols are part of the encoding symbols.

Packet Erasure Channel: A communication path where packets are either lost (e.g., in our case, by a congested router, or because the number of transmission errors exceeds the correction capabilities of the physical-layer code) or received. When a packet is received, it is assumed that this packet is not corrupted (i.e., in our case, the bit-errors, if any, are fixed by the physical-layer code and therefore hidden to the upper layers).

Repair Symbol: Encoding symbol that is not a source symbol.

Source Block: Group of ADUs that are to be FEC protected as a single block. This notion is restricted to Block FEC Codes.

Source Symbol: Unit of data used during the encoding process.

Systematic Code: FEC code in which the source symbols are part of the encoding symbols.

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

3. Summary of Architecture Overview

The architecture of [RFC6363], Section 3, equally applies to this FECFRAME extension and is not repeated here. However, we provide hereafter a quick summary to facilitate the understanding of this document to readers not familiar with the concepts and terminology.

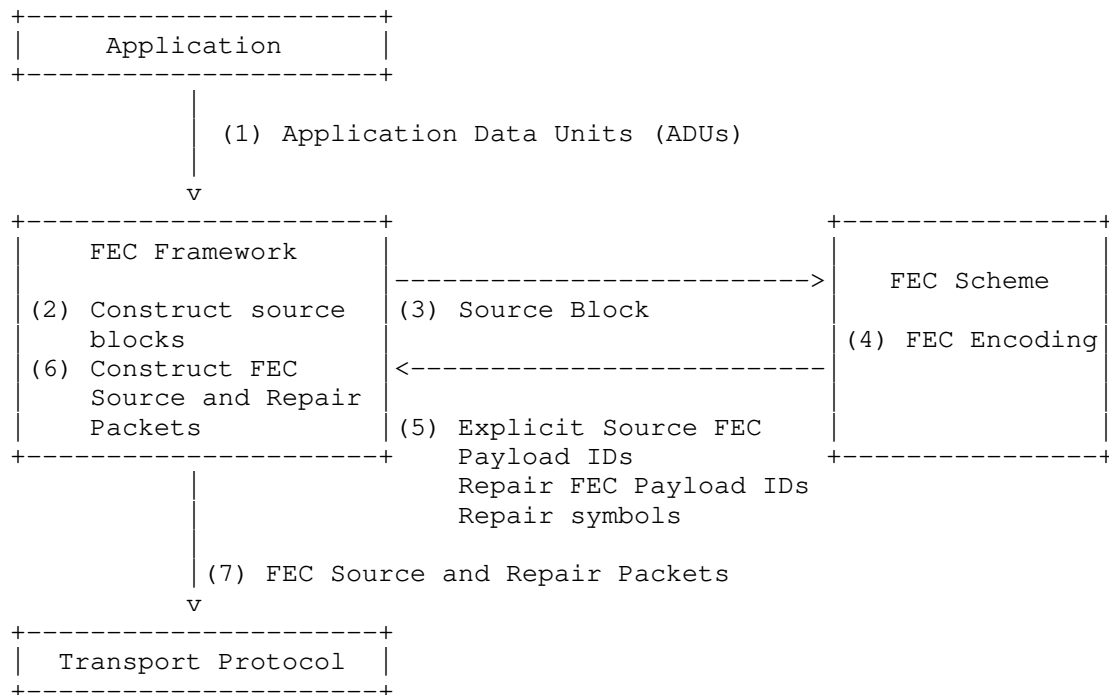


Figure 1: FECFRAME architecture at a sender.

The FECFRAME architecture is illustrated in Figure 1 from the sender's point of view, in case of a block FEC Scheme. It shows an application generating an ADU flow (other flows, from other applications, may co-exist). These ADUs, of variable size, must be somehow mapped to source symbols of fixed size (this fixed size is a requirement of all FEC Schemes that comes from the way mathematical operations are applied to symbols content). This is the goal of an ADU-to-symbols mapping process that is FEC-Scheme specific (see below). Once the source block is built, taking into account both the FEC Scheme constraints (e.g., in terms of maximum source block size) and the application's flow constraints (e.g., in terms of real-time constraints), the associated source symbols are handed to the FEC Scheme in order to produce an appropriate number of repair symbols. FEC Source Packets (containing ADUs) and FEC Repair Packets (containing one or more repair symbols each) are then generated and sent using an appropriate transport protocol (more precisely [RFC6363], Section 7, requires a transport protocol providing an unreliable datagram service, such as UDP). In practice FEC Source Packets may be passed to the transport layer as soon as available, without having to wait for FEC encoding to take place. In that case

a copy of the associated source symbols needs to be kept within FECFRAME for future FEC encoding purposes.

At a receiver (not shown), FECFRAME processing operates in a similar way, taking as input the incoming FEC Source and Repair Packets received. In case of FEC Source Packet losses, the FEC decoding of the associated block may recover all (in case of successful decoding) or a subset potentially empty (otherwise) of the missing source symbols. After source-symbol-to-ADU mapping, when lost ADUs are recovered, they are then assigned to their respective flow (see below). ADUs are returned to the application(s), either in their initial transmission order (in that case ADUs received after an erased one will be delayed until FEC decoding has taken place) or not (in that case each ADU is returned as soon as it is received or recovered), depending on the application requirements.

FECFRAME features two subtle mechanisms:

- o ADUs-to-source-symbols mapping: in order to manage variable size ADUs, FECFRAME and FEC Schemes can use small, fixed size symbols and create a mapping between ADUs and symbols. To each ADU this mechanism prepends a length field (plus a flow identifier, see below) and pads the result to a multiple of the symbol size. A small ADU may be mapped to a single source symbol while a large one may be mapped to multiple symbols. The mapping details are FEC-Scheme-dependent and must be defined in the associated document;
- o Assignment of decoded ADUs to flows in multi-flow configurations: when multiple flows are multiplexed over the same FECFRAME instance, a problem is to assign a decoded ADU to the right flow (UDP port numbers and IP addresses traditionally used to map incoming ADUs to flows are not recovered during FEC decoding). To make it possible, at the FECFRAME sending instance, each ADU is prepended with a flow identifier (1 byte) during the ADU-to-source-symbols mapping (see above). The flow identifiers are also shared between all FECFRAME instances as part of the FEC Framework Configuration Information. This (flow identifier + length + application payload + padding), called ADUI, is then FEC protected. Therefore a decoded ADUI contains enough information to assign the ADU to the right flow.

A few aspects are not covered by FECFRAME, namely:

- o [RFC6363] section 8 does not detail any congestion control mechanism, but only provides high level normative requirements;

- o the possibility of having feedbacks from receiver(s) is considered out of scope, although such a mechanism may exist within the application (e.g., through RTCP control messages);
- o flow adaptation at a FECFRAME sender (e.g., how to set the FEC code rate based on transmission conditions) is not detailed, but it needs to comply with the congestion control normative requirements (see above).

4. Procedural Overview

4.1. General

The general considerations of [RFC6363], Section 4.1, that are specific to block FEC codes are not repeated here.

With a Sliding Window FEC Code, the FEC Source Packet MUST contain information to identify the position occupied by the ADU within the source flow, in terms specific to the FEC Scheme. This information is known as the Source FEC Payload ID, and the FEC Scheme is responsible for defining and interpreting it.

With a Sliding Window FEC Code, the FEC Repair Packets MUST contain information that identifies the relationship between the contained repair payloads and the original source symbols used during encoding. This information is known as the Repair FEC Payload ID, and the FEC Scheme is responsible for defining and interpreting it.

The Sender Operation ([RFC6363], Section 4.2.) and Receiver Operation ([RFC6363], Section 4.3) are both specific to block FEC codes and therefore omitted below. The following two sections detail similar operations for Sliding Window FEC codes.

4.2. Sender Operation with Sliding Window FEC Codes

With a Sliding Window FEC Scheme, the following operations, illustrated in Figure 2 for the generic case (non-RTP repair flows), and in Figure 3 for the case of RTP repair flows, describe a possible way to generate compliant source and repair flows:

1. A new ADU is provided by the application.
2. The FEC Framework communicates this ADU to the FEC Scheme.
3. The sliding encoding window is updated by the FEC Scheme. The ADU-to-source-symbols mapping as well as the encoding window management details are both the responsibility of the FEC Scheme

and MUST be detailed there. Appendix A provides non-normative hints about what FEC Scheme designers need to consider;

4. The Source FEC Payload ID information of the source packet is determined by the FEC Scheme. If required by the FEC Scheme, the Source FEC Payload ID is encoded into the Explicit Source FEC Payload ID field and returned to the FEC Framework.
5. The FEC Framework constructs the FEC Source Packet according to [RFC6363] Figure 6, using the Explicit Source FEC Payload ID provided by the FEC Scheme if applicable.
6. The FEC Source Packet is sent using normal transport-layer procedures. This packet is sent using the same ADU flow identification information as would have been used for the original source packet if the FEC Framework were not present (e.g., the source and destination addresses and UDP port numbers on the IP datagram carrying the source packet will be the same whether or not the FEC Framework is applied).
7. When the FEC Framework needs to send one or several FEC Repair Packets (e.g., according to the target Code Rate), it asks the FEC Scheme to create one or several repair packet payloads from the current sliding encoding window along with their Repair FEC Payload ID.
8. The Repair FEC Payload IDs and repair packet payloads are provided back by the FEC Scheme to the FEC Framework.
9. The FEC Framework constructs FEC Repair Packets according to [RFC6363] Figure 7, using the FEC Payload IDs and repair packet payloads provided by the FEC Scheme.
10. The FEC Repair Packets are sent using normal transport-layer procedures. The port(s) and multicast group(s) to be used for FEC Repair Packets are defined in the FEC Framework Configuration Information.

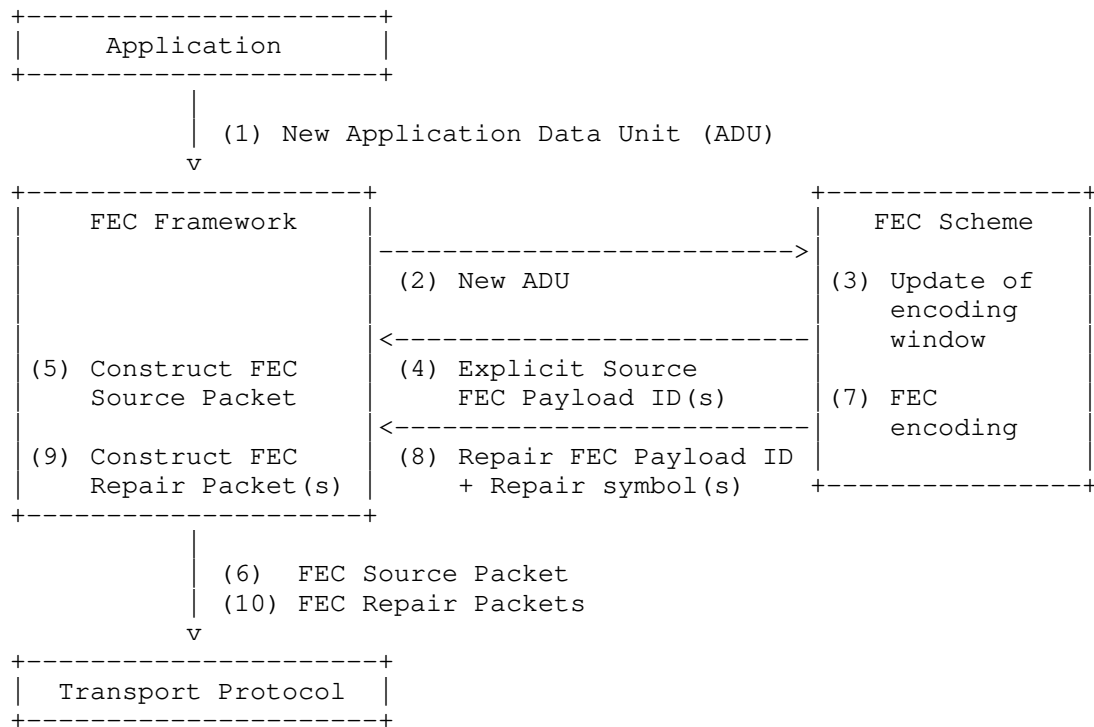


Figure 2: Sender Operation with Sliding Window FEC Codes

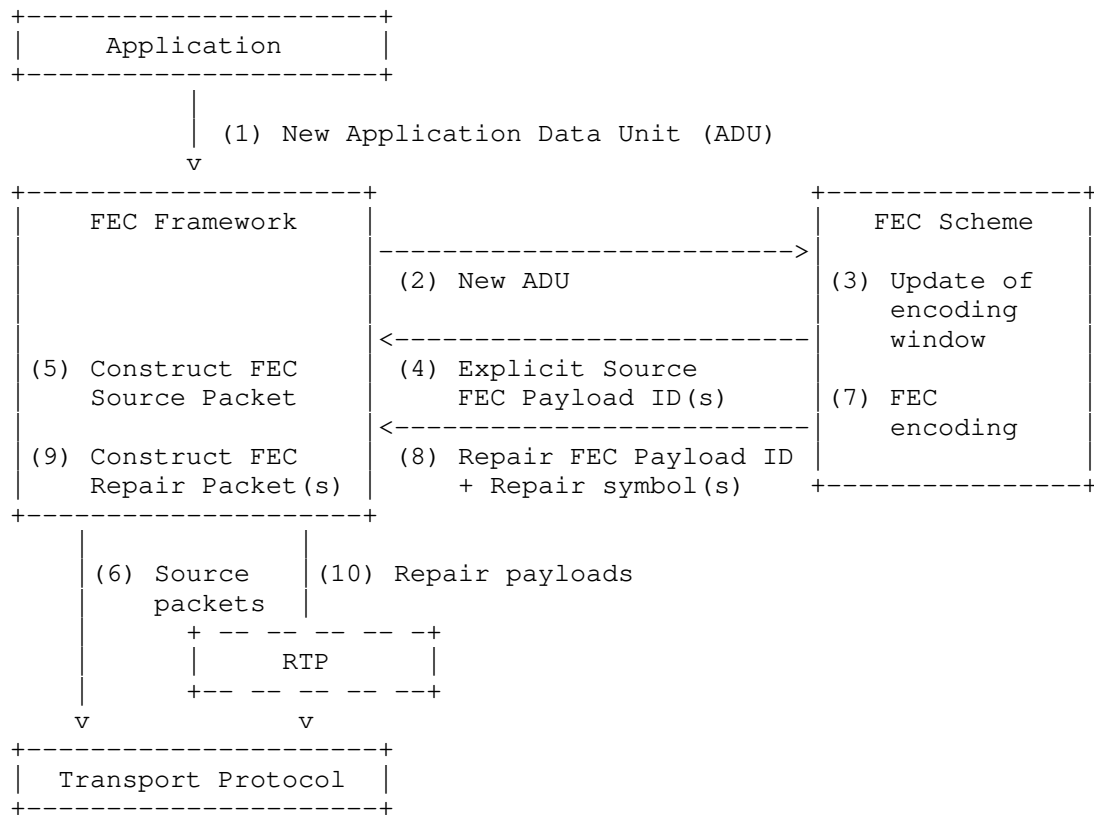


Figure 3: Sender Operation with Sliding Window FEC Codes and RTP Repair Flows

4.3. Receiver Operation with Sliding Window FEC Codes

With a Sliding Window FEC Scheme, the following operations, illustrated in Figure 4 for the generic case (non-RTP repair flows), and in Figure 5 for the case of RTP repair flows. The only differences with respect to block FEC codes lie in steps (4) and (5). Therefore this section does not repeat the other steps of [RFC6363], Section 4.3, "Receiver Operation". The new steps (4) and (5) are:

4. The FEC Scheme uses the received FEC Payload IDs (and derived FEC Source Payload IDs when the Explicit Source FEC Payload ID field is not used) to insert source and repair packets into the decoding window in the right way. If at least one source packet is missing and at least one repair packet has been received, then FEC decoding is attempted to recover missing source payloads. The FEC Scheme determines whether source packets have been lost

and whether enough repair packets have been received to decode any or all of the missing source payloads.

5. The FEC Scheme returns the received and decoded ADUs to the FEC Framework, along with indications of any ADUs that were missing and could not be decoded.

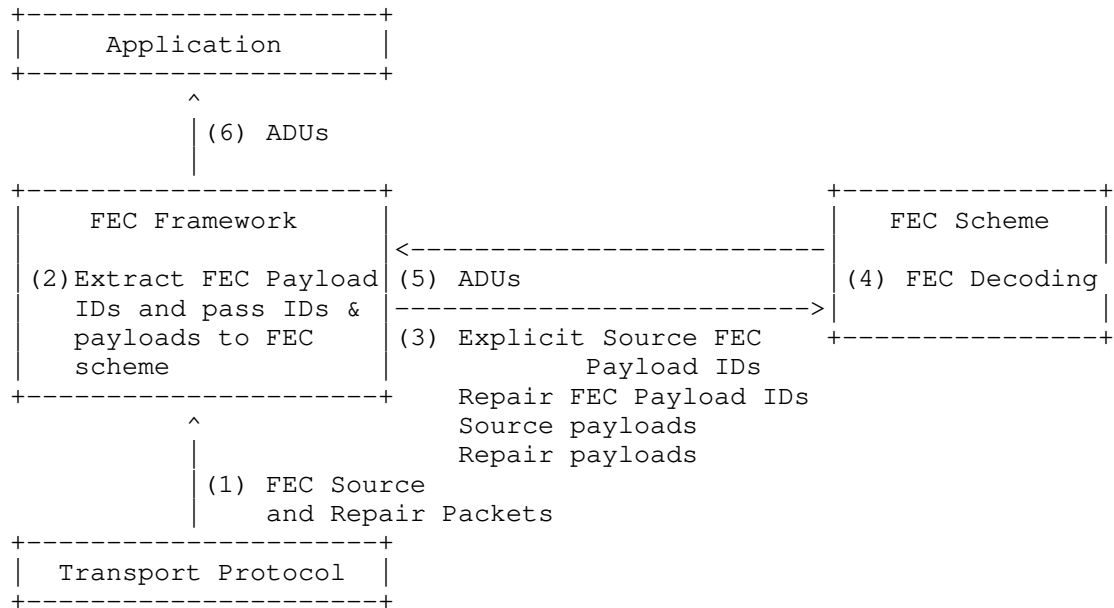


Figure 4: Receiver Operation with Sliding Window FEC Codes

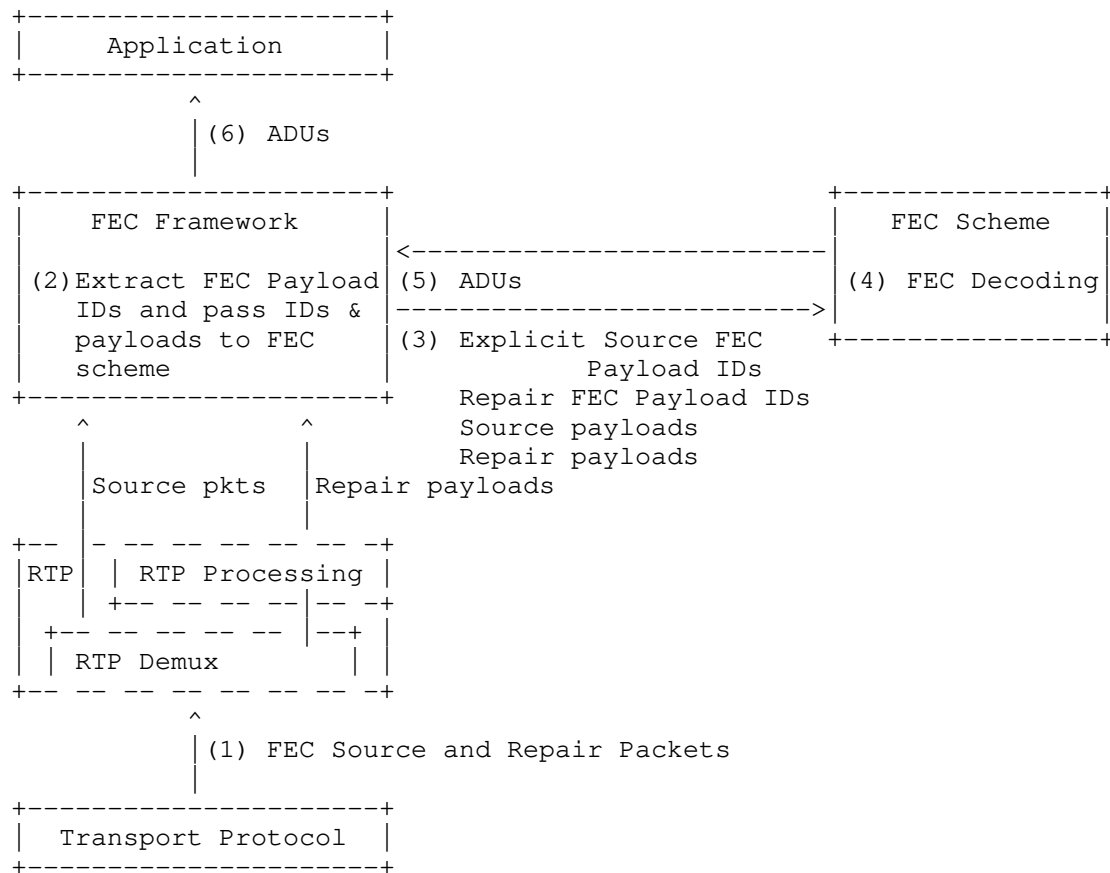


Figure 5: Receiver Operation with Sliding Window FEC Codes and RTP Repair Flows

5. Protocol Specification

5.1. General

This section discusses the protocol elements for the FEC Framework specific to Sliding Window FEC schemes. The global formats of source data packets (i.e., [RFC6363], Figure 6) and repair data packets (i.e., [RFC6363], Figures 7 and 8) remain the same with Sliding Window FEC codes. They are not repeated here.

5.2. FEC Framework Configuration Information

The FEC Framework Configuration Information considerations of [RFC6363], Section 5.5, equally applies to this FECFRAME extension and is not repeated here.

5.3. FEC Scheme Requirements

The FEC Scheme requirements of [RFC6363], Section 5.6, mostly apply to this FECFRAME extension and are not repeated here. An exception though is the "full specification of the FEC code", item (4), that is specific to block FEC codes. The following item (4-bis) applies in case of Sliding Window FEC schemes:

4-bis. A full specification of the Sliding Window FEC code

This specification MUST precisely define the valid FEC-Scheme-Specific Information values, the valid FEC Payload ID values, and the valid packet payload sizes (where packet payload refers to the space within a packet dedicated to carrying encoding symbols).

Furthermore, given valid values of the FEC-Scheme-Specific Information, a valid Repair FEC Payload ID value, a valid packet payload size, and a valid encoding window (i.e., a set of source symbols), the specification MUST uniquely define the values of the encoding symbol (or symbols) to be included in the repair packet payload with the given Repair FEC Payload ID value.

Additionally, the FEC Scheme associated to a Sliding Window FEC Code:

- o MUST define the relationships between ADUs and the associated source symbols (mapping);
- o MUST define the management of the encoding window that slides over the set of ADUs. Appendix A provides non normative hints about what FEC Scheme designers need to consider;
- o MUST define the management of the decoding window. This usually consists in managing a system of linear equations (in case of a linear FEC code);

6. Feedback

The discussion of [RFC6363], Section 6, equally applies to this FECFRAME extension and is not repeated here.

7. Transport Protocols

The discussion of [RFC6363], Section 7, equally applies to this FECFRAME extension and is not repeated here.

8. Congestion Control

The discussion of [RFC6363], Section 8, equally applies to this FECFRAME extension and is not repeated here.

9. Implementation Status

Editor's notes: RFC Editor, please remove this section motivated by RFC 7942 before publishing the RFC. Thanks!

An implementation of FECFRAME extended to Sliding Window codes exists:

- o Organisation: Inria
- o Description: This is an implementation of FECFRAME extended to Sliding Window codes and supporting the RLC FEC Scheme [RLC-ID]. It is based on: (1) a proprietary implementation of FECFRAME, made by Inria and Expway for which interoperability tests have been conducted; and (2) a proprietary implementation of RLC Sliding Window FEC Codes.
- o Maturity: the basic FECFRAME maturity is "production", the FECFRAME extension maturity is "under progress".
- o Coverage: the software implements a subset of [RFC6363], as specialized by the 3GPP eMBMS standard [MBMSTS]. This software also covers the additional features of FECFRAME extended to Sliding Window codes, in particular the RLC FEC Scheme.
- o Licensing: proprietary.
- o Implementation experience: maximum.
- o Information update date: March 2018.
- o Contact: vincent.roca@inria.fr

10. Security Considerations

This FECFRAME extension does not add any new security consideration. All the considerations of [RFC6363], Section 9, apply to this document as well. However, for the sake of completeness, the

following goal can be added to the list provided in Section 9.1 "Problem Statement" of [RFC6363]:

- o Attacks can try to corrupt source flows in order to modify the receiver application's behavior (as opposed to just denying service).

11. Operations and Management Considerations

This FECFRAME extension does not add any new Operations and Management Consideration. All the considerations of [RFC6363], Section 10, apply to this document as well.

12. IANA Considerations

No IANA actions are required for this document.

A FEC Scheme for use with this FEC Framework is identified via its FEC Encoding ID. It is subject to IANA registration in the "FEC Framework (FECFRAME) FEC Encoding IDs" registry. All the rules of [RFC6363], Section 11, apply and are not repeated here.

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Appendix A. About Sliding Encoding Window Management (informational)

The FEC Framework does not specify the management of the sliding encoding window which is the responsibility of the FEC Scheme. This annex only provides a few informational hints.

Source symbols are added to the sliding encoding window each time a new ADU is available at the sender, after the ADU-to-source-symbol mapping specific to the FEC Scheme.

Source symbols are removed from the sliding encoding window, for instance:

- o after a certain delay, when an "old" ADU of a real-time flow times out. The source symbol retention delay in the sliding encoding window should therefore be initialized according to the real-time features of incoming flow(s) when applicable;
- o once the sliding encoding window has reached its maximum size (there is usually an upper limit to the sliding encoding window size). In that case the oldest symbol is removed each time a new source symbol is added.

Several considerations can impact the management of this sliding encoding window:

- o at the source flows level: real-time constraints can limit the total time source symbols can remain in the encoding window;
- o at the FEC code level: theoretical or practical limitations (e.g., because of computational complexity) can limit the number of source symbols in the encoding window;
- o at the FEC Scheme level: signaling and window management are intrinsically related. For instance, an encoding window composed of a non-sequential set of source symbols requires an appropriate signaling to inform a receiver of the composition of the encoding window, and the associated transmission overhead can limit the maximum encoding window size. On the opposite, an encoding window always composed of a sequential set of source symbols simplifies signaling: providing the identity of the first source symbol plus their number is sufficient, which creates a fixed and relatively small transmission overhead.

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Low Latency, Low Loss, Scalable Throughput (L4S) Internet Service:
Architecture
draft-ietf-tsvwg-l4s-arch-17

Abstract

This document describes the L4S architecture, which enables Internet applications to achieve Low queuing Latency, Low Loss, and Scalable throughput (L4S). The insight on which L4S is based is that the root cause of queuing delay is in the congestion controllers of senders, not in the queue itself. With the L4S architecture all Internet applications could (but do not have to) transition away from congestion control algorithms that cause substantial queuing delay, to a new class of congestion controls that induce very little queuing, aided by explicit congestion signalling from the network. This new class of congestion controls can provide low latency for capacity-seeking flows, so applications can achieve both high bandwidth and low latency.

The architecture primarily concerns incremental deployment. It defines mechanisms that allow the new class of L4S congestion controls to coexist with 'Classic' congestion controls in a shared network. These mechanisms aim to ensure that the latency and throughput performance using an L4S-compliant congestion controller is usually much better (and rarely worse) than performance would have been using a 'Classic' congestion controller, and that competing flows continuing to use 'Classic' controllers are typically not impacted by the presence of L4S. These characteristics are important to encourage adoption of L4S congestion control algorithms and L4S compliant network elements.

The L4S architecture consists of three components: network support to isolate L4S traffic from classic traffic; protocol features that allow network elements to identify L4S traffic; and host support for L4S congestion controls.

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1. Introduction

At any one time, it is increasingly common for all of the traffic in a bottleneck link (e.g. a household's Internet access) to come from applications that prefer low delay: interactive Web, Web services, voice, conversational video, interactive video, interactive remote presence, instant messaging, online gaming, remote desktop, cloud-based applications and video-assisted remote control of machinery and industrial processes. In the last decade or so, much has been done to reduce propagation delay by placing caches or servers closer to users. However, queuing remains a major, albeit intermittent, component of latency. For instance spikes of hundreds of milliseconds are not uncommon, even with state-of-the-art active queue management (AQM) [COBALT], [DOCSIS3AQM]. Queuing in access network bottlenecks is typically configured to cause overall network delay to roughly double during a long-running flow, relative to expected base (unloaded) path delay [BufferSize]. Low loss is also important because, for interactive applications, losses translate into even longer retransmission delays.

It has been demonstrated that, once access network bit rates reach levels now common in the developed world, increasing capacity offers diminishing returns if latency (delay) is not addressed [Dukkipati06], [Rajiullah15]. Therefore, the goal is an Internet service with very Low queueing Latency, very Low Loss and Scalable throughput (L4S). Very low queueing latency means less than 1 millisecond (ms) on average and less than about 2 ms at the 99th percentile. This document describes the L4S architecture for achieving these goals.

Differentiated services (Diffserv) offers Expedited Forwarding (EF [RFC3246]) for some packets at the expense of others, but this makes no difference when all (or most) of the traffic at a bottleneck at any one time requires low latency. In contrast, L4S still works well when all traffic is L4S - a service that gives without taking needs none of the configuration or management baggage (traffic policing, traffic contracts) associated with favouring some traffic flows over others.

Queuing delay degrades performance intermittently [Hohlfeld14]. It occurs when a large enough capacity-seeking (e.g. TCP) flow is running alongside the user's traffic in the bottleneck link, which is typically in the access network. Or when the low latency application is itself a large capacity-seeking or adaptive rate (e.g. interactive video) flow. At these times, the performance improvement from L4S must be sufficient that network operators will be motivated to deploy it.

Active Queue Management (AQM) is part of the solution to queuing under load. AQM improves performance for all traffic, but there is a limit to how much queuing delay can be reduced by solely changing the network; without addressing the root of the problem.

The root of the problem is the presence of standard TCP congestion control (Reno [RFC5681]) or compatible variants (e.g. TCP Cubic [RFC8312]). We shall use the term 'Classic' for these Reno-friendly congestion controls. Classic congestion controls induce relatively large saw-tooth-shaped excursions up the queue and down again, which have been growing as flow rate scales [RFC3649]. So if a network operator naively attempts to reduce queuing delay by configuring an AQM to operate at a shallower queue, a Classic congestion control will significantly underutilize the link at the bottom of every saw-tooth.

It has been demonstrated that if the sending host replaces a Classic congestion control with a 'Scalable' alternative, when a suitable AQM is deployed in the network the performance under load of all the above interactive applications can be significantly improved. For instance, queuing delay under heavy load with the example DCTCP/DualQ solution cited below on a DSL or Ethernet link is roughly 1 to 2 milliseconds at the 99th percentile without losing link utilization [DualPI2Linux], [DCTtH19] (for other link types, see Section 6.3). This compares with 5-20 ms on average with a Classic congestion control and current state-of-the-art AQMs such as FQ-CoDel [RFC8290], PIE [RFC8033] or DOCSIS PIE [RFC8034] and about 20-30 ms at the 99th percentile [DualPI2Linux].

L4S is designed for incremental deployment. It is possible to deploy the L4S service at a bottleneck link alongside the existing best efforts service [DualPI2Linux] so that unmodified applications can start using it as soon as the sender's stack is updated. Access networks are typically designed with one link as the bottleneck for each site (which might be a home, small enterprise or mobile device), so deployment at either or both ends of this link should give nearly all the benefit in the respective direction. With some transport protocols, namely TCP and SCTP, the sender has to check for suitably updated receiver feedback, whereas with more recent transport protocols such as QUIC and DCCP, all receivers have always been suitable.

This document presents the L4S architecture, by describing and justifying the component parts and how they interact to provide the scalable, low latency, low loss Internet service. It also details the approach to incremental deployment, as briefly summarized above.

1.1. Document Roadmap

This document describes the L4S architecture in three passes. First this brief overview gives the very high level idea and states the main components with minimal rationale. This is only intended to give some context for the terminology definitions that follow in Section 3, and to explain the structure of the rest of the document. Then Section 4 goes into more detail on each component with some rationale, but still mostly stating what the architecture is, rather than why. Finally Section 5 justifies why each element of the solution was chosen (Section 5.1) and why these choices were different from other solutions (Section 5.2).

Having described the architecture, Section 6 clarifies its applicability; that is, the applications and use-cases that motivated the design, the challenges applying the architecture to various link technologies, and various incremental deployment models: including the two main deployment topologies, different sequences for incremental deployment and various interactions with pre-existing approaches. The document ends with the usual tail pieces, including extensive discussion of traffic policing and other security considerations Section 8.

2. L4S Architecture Overview

Below we outline the three main components to the L4S architecture; 1) the scalable congestion control on the sending host; 2) the AQM at the network bottleneck; and 3) the protocol between them.

But first, the main point to grasp is that low latency is not provided by the network - low latency results from the careful behaviour of the scalable congestion controllers used by L4S senders. The network does have a role - primarily to isolate the low latency of the carefully behaving L4S traffic from the higher queuing delay needed by traffic with pre-existing Classic behaviour. The network also alters the way it signals queue growth to the transport - It uses the Explicit Congestion Notification (ECN) protocol, but it signals the very start of queue growth - immediately without the smoothing delay typical of Classic AQMs. Because ECN support is essential for L4S, senders use the ECN field as the protocol to identify to the network which packets are L4S and which are Classic.

- 1) Host: Scalable congestion controls already exist. They solve the scaling problem with Classic congestion controls, such as Reno or Cubic. Because flow rate has scaled since TCP congestion control was first designed in 1988, assuming the flow lasts long enough, it now takes hundreds of round trips (and growing) to recover after a congestion signal (whether a loss or an ECN mark) as shown in the examples in Section 5.1 and [RFC3649]. Therefore control of queuing and utilization becomes very slack, and the slightest disturbances (e.g. from new flows starting) prevent a high rate from being attained.

With a scalable congestion control, the average time from one congestion signal to the next (the recovery time) remains invariant as the flow rate scales, all other factors being equal. This maintains the same degree of control over queuing and utilization whatever the flow rate, as well as ensuring that high throughput is more robust to disturbances. The scalable control used most widely (in controlled environments) is Data Center TCP (DCTCP [RFC8257]), which has been implemented and deployed in Windows Server Editions (since 2012), in Linux and in FreeBSD. Although DCTCP as-is functions well over wide-area round trip times, most implementations lack certain safety features that would be necessary for use outside controlled environments like data centres (see Section 6.4.3). So scalable congestion control needs to be implemented in TCP and other transport protocols (QUIC, SCTP, RTP/RTCP, RMCAT, etc.). Indeed, between the present document being drafted and published, the following scalable congestion controls were implemented: TCP Prague [PragueLinux], QUIC Prague, an L4S variant of the RMCAT SCReAM controller [SCReAM] and the L4S ECN part of BBRv2 [BBRv2] intended for TCP and QUIC transports.

- 2) Network: L4S traffic needs to be isolated from the queuing

latency of Classic traffic. One queue per application flow (FQ) is one way to achieve this, e.g. FQ-CoDel [RFC8290]. However, just two queues is sufficient and does not require inspection of transport layer headers in the network, which is not always possible (see Section 5.2). With just two queues, it might seem impossible to know how much capacity to schedule for each queue without inspecting how many flows at any one time are using each. And it would be undesirable to arbitrarily divide access network capacity into two partitions. The Dual Queue Coupled AQM was developed as a minimal complexity solution to this problem. It acts like a 'semi-permeable' membrane that partitions latency but not bandwidth. As such, the two queues are for transition from Classic to L4S behaviour, not bandwidth prioritization.

Section 4 gives a high level explanation of how the per-flow-queue (FQ) and DualQ variants of L4S work, and [I-D.ietf-tsvwg-aqm-dualq-coupled] gives a full explanation of the DualQ Coupled AQM framework. A specific marking algorithm is not mandated for L4S AQMs. Appendices of [I-D.ietf-tsvwg-aqm-dualq-coupled] give non-normative examples that have been implemented and evaluated, and give recommended default parameter settings. It is expected that L4S experiments will improve knowledge of parameter settings and whether the set of marking algorithms needs to be limited.

- 3) Protocol: A host needs to distinguish L4S and Classic packets with an identifier so that the network can classify them into their separate treatments. The L4S identifier spec. [I-D.ietf-tsvwg-ecn-l4s-id] concludes that all alternatives involve compromises, but the ECT(1) and CE codepoints of the ECN field represent a workable solution. As already explained, the network also uses ECN to immediately signal the very start of queue growth to the transport.

3. Terminology

Note: The following definitions are copied from the L4S ECN spec [I-D.ietf-tsvwg-ecn-l4s-id] for convenience. If there are accidental differences, those in [I-D.ietf-tsvwg-ecn-l4s-id] take precedence.

Classic Congestion Control: A congestion control behaviour that can co-exist with standard Reno [RFC5681] without causing significantly negative impact on its flow rate [RFC5033]. The scaling problem with Classic congestion control is explained, with examples, in Section 5.1 and in [RFC3649].

Scalable Congestion Control: A congestion control where the average

time from one congestion signal to the next (the recovery time) remains invariant as the flow rate scales, all other factors being equal. For instance, DCTCP averages 2 congestion signals per round-trip whatever the flow rate, as do other recently developed scalable congestion controls, e.g. Relentless TCP [Mathis09], TCP Prague [I-D.briscoe-iccrp-prague-congestion-control], [PragueLinux], BBRv2 [BBRv2], [I-D.cardwell-iccrp-bbr-congestion-control] and the L4S variant of SCReAM for real-time media [SCReAM], [RFC8298]). See Section 4.3 of [I-D.ietf-tsvwg-ecn-l4s-id] for more explanation.

Classic service: The Classic service is intended for all the congestion control behaviours that co-exist with Reno [RFC5681] (e.g. Reno itself, Cubic [RFC8312], Compound [I-D.sridharan-tcpm-ctcp], TFRC [RFC5348]). The term 'Classic queue' means a queue providing the Classic service.

Low-Latency, Low-Loss Scalable throughput (L4S) service: The 'L4S' service is intended for traffic from scalable congestion control algorithms, such as the Prague congestion control [I-D.briscoe-iccrp-prague-congestion-control], which was derived from DCTCP [RFC8257]. The L4S service is for more general traffic than just TCP Prague -- it allows the set of congestion controls with similar scaling properties to Prague to evolve, such as the examples listed above (Relentless, SCReAM). The term 'L4S queue' means a queue providing the L4S service.

The terms Classic or L4S can also qualify other nouns, such as 'queue', 'codepoint', 'identifier', 'classification', 'packet', 'flow'. For example: an L4S packet means a packet with an L4S identifier sent from an L4S congestion control.

Both Classic and L4S services can cope with a proportion of unresponsive or less-responsive traffic as well, but in the L4S case its rate has to be smooth enough or low enough not build a queue (e.g. DNS, VoIP, game sync datagrams, etc).

Reno-friendly: The subset of Classic traffic that is friendly to the standard Reno congestion control defined for TCP in [RFC5681]. The TFRC spec. [RFC5348] indirectly implies that 'friendly' is defined as "generally within a factor of two of the sending rate of a TCP flow under the same conditions". Reno-friendly is used here in place of 'TCP-friendly', given the latter has become imprecise, because the TCP protocol is now used with so many different congestion control behaviours, and Reno is used in non-TCP transports such as QUIC [RFC9000].

Classic ECN: The original Explicit Congestion Notification (ECN)

protocol [RFC3168], which requires ECN signals to be treated as equivalent to drops, both when generated in the network and when responded to by the sender.

L4S uses the ECN field as an identifier [I-D.ietf-tsvwg-ecn-l4s-id] with the names for the four codepoints of the 2-bit IP-ECN field unchanged from those defined in the ECN spec [RFC3168]: Not ECT, ECT(0), ECT(1) and CE, where ECT stands for ECN-Capable Transport and CE stands for Congestion Experienced. A packet marked with the CE codepoint is termed 'ECN-marked' or sometimes just 'marked' where the context makes ECN obvious.

Site: A home, mobile device, small enterprise or campus, where the network bottleneck is typically the access link to the site. Not all network arrangements fit this model but it is a useful, widely applicable generalization.

4. L4S Architecture Components

The L4S architecture is composed of the elements in the following three subsections.

4.1. Protocol Mechanisms

The L4S architecture involves: a) unassignment of an identifier; b) reassignment of the same identifier; and c) optional further identifiers:

- a. An essential aspect of a scalable congestion control is the use of explicit congestion signals. 'Classic' ECN [RFC3168] requires an ECN signal to be treated as equivalent to drop, both when it is generated in the network and when it is responded to by hosts. L4S needs networks and hosts to support a more fine-grained meaning for each ECN signal that is less severe than a drop, so that the L4S signals:

- * can be much more frequent;
- * can be signalled immediately, without the significant delay required to smooth out fluctuations in the queue.

To enable L4S, the standards track Classic ECN spec. [RFC3168] has had to be updated to allow L4S packets to depart from the 'equivalent to drop' constraint. [RFC8311] is a standards track update to relax specific requirements in RFC 3168 (and certain other standards track RFCs), which clears the way for the experimental changes proposed for L4S. [RFC8311] also reclassifies the original experimental assignment of the ECT(1) codepoint as an ECN nonce [RFC3540] as historic.

- b. [I-D.ietf-tsvwg-ecn-l4s-id] specifies that ECT(1) is used as the identifier to classify L4S packets into a separate treatment from Classic packets. This satisfies the requirement for identifying an alternative ECN treatment in [RFC4774].

The CE codepoint is used to indicate Congestion Experienced by both L4S and Classic treatments. This raises the concern that a Classic AQM earlier on the path might have marked some ECT(0) packets as CE. Then these packets will be erroneously classified into the L4S queue. Appendix B of the L4S ECN spec [I-D.ietf-tsvwg-ecn-l4s-id] explains why five unlikely eventualities all have to coincide for this to have any detrimental effect, which even then would only involve a vanishingly small likelihood of a spurious retransmission.

- c. A network operator might wish to include certain unresponsive, non-L4S traffic in the L4S queue if it is deemed to be smoothly enough paced and low enough rate not to build a queue. For instance, VoIP, low rate datagrams to sync online games, relatively low rate application-limited traffic, DNS, LDAP, etc. This traffic would need to be tagged with specific identifiers, e.g. a low latency Diffserv Codepoint such as Expedited Forwarding (EF [RFC3246]), Non-Queue-Building (NQB [I-D.ietf-tsvwg-nqb]), or operator-specific identifiers.

4.2. Network Components

The L4S architecture aims to provide low latency without the need for per-flow operations in network components. Nonetheless, the architecture does not preclude per-flow solutions. The following bullets describe the known arrangements: a) the DualQ Coupled AQM with an L4S AQM in one queue coupled from a Classic AQM in the other; b) Per-Flow Queues with an instance of a Classic and an L4S AQM in each queue; c) Dual queues with per-flow AQMs, but no per-flow queues:

- a. The Dual Queue Coupled AQM (illustrated in Figure 1) achieves the 'semi-permeable' membrane property mentioned earlier as follows:

- * Latency isolation: Two separate queues are used to isolate L4S queuing delay from the larger queue that Classic traffic needs to maintain full utilization.
- * Bandwidth pooling: The two queues act as if they are a single pool of bandwidth in which flows of either type get roughly equal throughput without the scheduler needing to identify any flows. This is achieved by having an AQM in each queue, but the Classic AQM provides a congestion signal to both queues in a manner that ensures a consistent response from the two classes of congestion control. Specifically, the Classic AQM generates a drop/mark probability based on congestion in its own queue, which it uses both to drop/mark packets in its own queue and to affect the marking probability in the L4S queue. The strength of the coupling of the congestion signalling between the two queues is enough to make the L4S flows slow down to leave the right amount of capacity for the Classic flows (as they would if they were the same type of traffic sharing the same queue).

Then the scheduler can serve the L4S queue with priority (denoted by the '1' on the higher priority input), because the L4S traffic isn't offering up enough traffic to use all the priority that it is given. Therefore:

- * for latency isolation on short time-scales (sub-round-trip) the prioritization of the L4S queue protects its low latency by allowing bursts to dissipate quickly;
- * but for bandwidth pooling on longer time-scales (round-trip and longer) the Classic queue creates an equal and opposite pressure against the L4S traffic to ensure that neither has priority when it comes to bandwidth - the tension between prioritizing L4S and coupling the marking from the Classic AQM results in approximate per-flow fairness.

To protect against unresponsive traffic taking advantage of the prioritization of the L4S queue and starving the Classic queue, it is advisable for the priority to be conditional, not strict (see Appendix A of the DualQ spec [I-D.ietf-tsvwg-aqm-dualq-coupled]).

When there is no Classic traffic, the L4S queue's own AQM comes into play. It starts congestion marking with a very shallow queue, so L4S traffic maintains very low queuing delay.

If either queue becomes persistently overloaded, drop of ECN-capable packets is introduced, as recommended in Section 7 of the ECN spec [RFC3168] and Section 4.2.1 of the AQM recommendations [RFC7567]. Then both queues introduce the same level of drop (not shown in the figure).

The Dual Queue Coupled AQM has been specified as generically as possible [I-D.ietf-tsvwg-aqm-dualq-coupled] without specifying the particular AQMs to use in the two queues so that designers are free to implement diverse ideas. Informational appendices in that draft give pseudocode examples of two different specific AQM approaches: one called DualPI2 (pronounced Dual PI Squared) [DualPI2Linux] that uses the PI2 variant of PIE, and a zero-config variant of RED called Curvy RED. A DualQ Coupled AQM based on PIE has also been specified and implemented for Low Latency DOCSIS [DOCSIS3.1].

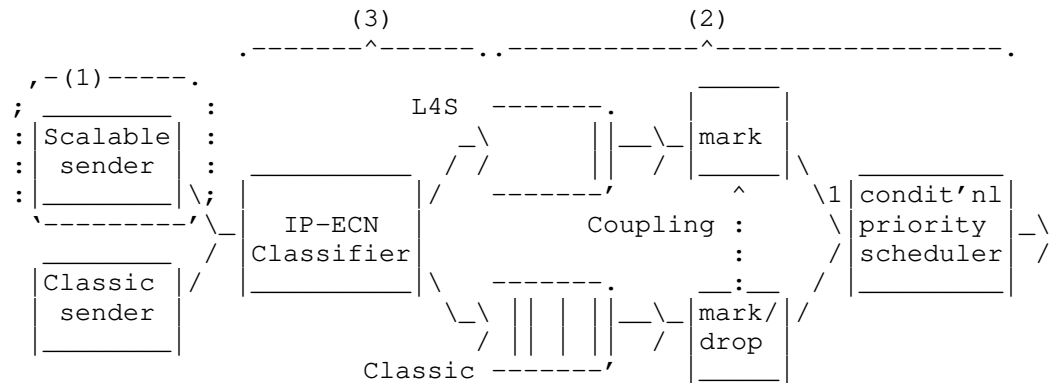


Figure 1: Components of an L4S DualQ Coupled AQM Solution: 1) Scalable Sending Host; 2) Isolation in separate network queues; and 3) Packet Identification Protocol

- b. **Per-Flow Queues and AQMs:** A scheduler with per-flow queues such as FQ-CoDel or FQ-PIE can be used for L4S. For instance within each queue of an FQ-CoDel system, as well as a CoDel AQM, there is typically also the option of ECN marking at an immediate (unsmoothed) shallow threshold to support use in data centres (see Sec.5.2.7 of the FQ-CoDel spec [RFC8290]). In Linux, this has been modified so that the shallow threshold can be solely applied to ECT(1) packets [FQ_CoDel_Thresh]. Then if there is a flow of non-ECN or ECT(0) packets in the per-flow-queue, the Classic AQM (e.g. CoDel) is applied; while if there is a flow of ECT(1) packets in the queue, the shallower (typically sub-millisecond) threshold is applied. In addition, ECT(0) and not-ECT packets could potentially be classified into a separate flow-queue from ECT(1) and CE packets to avoid them mixing if they share a common flow-identifier (e.g. in a VPN).
- c. **Dual-queues, but per-flow AQMs:** It should also be possible to use dual queues for isolation, but with per-flow marking to control flow-rates (instead of the coupled per-queue marking of the Dual Queue Coupled AQM). One of the two queues would be for isolating L4S packets, which would be classified by the ECN codepoint. Flow rates could be controlled by flow-specific marking. The policy goal of the marking could be to differentiate flow rates (e.g. [Nadas20], which requires additional signalling of a per-flow 'value'), or to equalize flow-rates (perhaps in a similar way to Approx Fair CoDel [AFCD], [I-D.morton-tsvwg-codel-approx-fair], but with two queues not one).

Note that whenever the term 'DualQ' is used loosely without saying whether marking is per-queue or per-flow, it means a dual queue AQM with per-queue marking.

4.3. Host Mechanisms

The L4S architecture includes two main mechanisms in the end host that we enumerate next:

- a. **Scalable Congestion Control at the sender:** Section 2 defines a scalable congestion control as one where the average time from one congestion signal to the next (the recovery time) remains invariant as the flow rate scales, all other factors being equal. Data Center TCP is the most widely used example. It has been documented as an informational record of the protocol currently in use in controlled environments [RFC8257]. A draft list of safety and performance improvements for a scalable congestion control to be usable on the public Internet has been drawn up (the so-called 'Prague L4S requirements' in Appendix A of

[I-D.ietf-tsvwg-ecn-l4s-id]). The subset that involve risk of harm to others have been captured as normative requirements in Section 4 of [I-D.ietf-tsvwg-ecn-l4s-id]. TCP Prague [I-D.briscoe-iccrp-prague-congestion-control] has been implemented in Linux as a reference implementation to address these requirements [PragueLinux].

Transport protocols other than TCP use various congestion controls that are designed to be friendly with Reno. Before they can use the L4S service, they will need to be updated to implement a scalable congestion response, which they will have to indicate by using the ECT(1) codepoint. Scalable variants are under consideration for more recent transport protocols, e.g. QUIC, and the L4S ECN part of BBRv2 [BBRv2], [I-D.cardwell-iccrp-bbr-congestion-control] is a scalable congestion control intended for the TCP and QUIC transports, amongst others. Also an L4S variant of the RMCAT SCReAM controller [RFC8298] has been implemented [SCReAM] for media transported over RTP.

Section 4.3 of the L4S ECN spec [I-D.ietf-tsvwg-ecn-l4s-id] defines scalable congestion control in more detail, and specifies that requirements that an L4S scalable congestion control has to comply with.

- b. The ECN feedback in some transport protocols is already sufficiently fine-grained for L4S (specifically DCCP [RFC4340] and QUIC [RFC9000]). But others either require update or are in the process of being updated:
 - * For the case of TCP, the feedback protocol for ECN embeds the assumption from Classic ECN [RFC3168] that an ECN mark is equivalent to a drop, making it unusable for a scalable TCP. Therefore, the implementation of TCP receivers will have to be upgraded [RFC7560]. Work to standardize and implement more accurate ECN feedback for TCP (AccECN) is in progress [I-D.ietf-tcpm-accurate-ecn], [PragueLinux].
 - * ECN feedback is only roughly sketched in an appendix of the SCTP specification [RFC4960]. A fuller specification has been proposed in a long-expired draft [I-D.stewart-tsvwg-sctpecn], which would need to be implemented and deployed before SCTCP could support L4S.
 - * For RTP, sufficient ECN feedback was defined in [RFC6679], but [RFC8888] defines the latest standards track improvements.

5. Rationale

5.1. Why These Primary Components?

Explicit congestion signalling (protocol): Explicit congestion signalling is a key part of the L4S approach. In contrast, use of drop as a congestion signal creates a tension because drop is both an impairment (less would be better) and a useful signal (more would be better):

- * Explicit congestion signals can be used many times per round trip, to keep tight control, without any impairment. Under heavy load, even more explicit signals can be applied so the queue can be kept short whatever the load. In contrast, Classic AQMs have to introduce very high packet drop at high load to keep the queue short. By using ECN, an L4S congestion control's sawtooth reduction can be smaller and therefore return to the operating point more often, without worrying that more sawteeth will cause more signals. The consequent smaller amplitude sawteeth fit between an empty queue and a very shallow marking threshold (~1 ms in the public Internet), so queue delay variation can be very low, without risk of under-utilization.
- * Explicit congestion signals can be emitted immediately to track fluctuations of the queue. L4S shifts smoothing from the network to the host. The network doesn't know the round trip times of any of the flows. So if the network is responsible for smoothing (as in the Classic approach), it has to assume a worst case RTT, otherwise long RTT flows would become unstable. This delays Classic congestion signals by 100-200 ms. In contrast, each host knows its own round trip time. So, in the L4S approach, the host can smooth each flow over its own RTT, introducing no more soothing delay than strictly necessary (usually only a few milliseconds). A host can also choose not to introduce any smoothing delay if appropriate, e.g. during flow start-up.

Neither of the above are feasible if explicit congestion signalling has to be considered 'equivalent to drop' (as was required with Classic ECN [RFC3168]), because drop is an impairment as well as a signal. So drop cannot be excessively frequent, and drop cannot be immediate, otherwise too many drops would turn out to have been due to only a transient fluctuation in the queue that would not have warranted dropping a packet in hindsight. Therefore, in an L4S AQM, the L4S queue uses a new L4S variant of ECN that is not equivalent to drop (see section 5.2 of

the L4S ECN spec [I-D.ietf-tsvwg-ecn-l4s-id]), while the Classic queue uses either Classic ECN [RFC3168] or drop, which are equivalent to each other.

Before Classic ECN was standardized, there were various proposals to give an ECN mark a different meaning from drop. However, there was no particular reason to agree on any one of the alternative meanings, so 'equivalent to drop' was the only compromise that could be reached. RFC 3168 contains a statement that:

"An environment where all end nodes were ECN-Capable could allow new criteria to be developed for setting the CE codepoint, and new congestion control mechanisms for end-node reaction to CE packets. However, this is a research issue, and as such is not addressed in this document."

Latency isolation (network): L4S congestion controls keep queue delay low whereas Classic congestion controls need a queue of the order of the RTT to avoid under-utilization. One queue cannot have two lengths, therefore L4S traffic needs to be isolated in a separate queue (e.g. DualQ) or queues (e.g. FQ).

Coupled congestion notification: Coupling the congestion notification between two queues as in the DualQ Coupled AQM is not necessarily essential, but it is a simple way to allow senders to determine their rate, packet by packet, rather than be overridden by a network scheduler. An alternative is for a network scheduler to control the rate of each application flow (see discussion in Section 5.2).

L4S packet identifier (protocol): Once there are at least two treatments in the network, hosts need an identifier at the IP layer to distinguish which treatment they intend to use.

Scalable congestion notification: A scalable congestion control in the host keeps the signalling frequency from the network high whatever the flow rate, so that queue delay variations can be small when conditions are stable, and rate can track variations in available capacity as rapidly as possible otherwise.

Low loss: Latency is not the only concern of L4S. The 'Low Loss' part of the name denotes that L4S generally achieves zero congestion loss due to its use of ECN. Otherwise, loss would itself cause delay, particularly for short flows, due to retransmission delay [RFC2884].

Scalable throughput: The "Scalable throughput" part of the name

denotes that the per-flow throughput of scalable congestion controls should scale indefinitely, avoiding the imminent scaling problems with Reno-friendly congestion control algorithms [RFC3649]. It was known when TCP congestion avoidance was first developed in 1988 that it would not scale to high bandwidth-delay products (see footnote 6 in [TCP-CA]). Today, regular broadband flow rates over WAN distances are already beyond the scaling range of Classic Reno congestion control. So 'less unscalable' Cubic [RFC8312] and Compound [I-D.sridharan-tcpm-ctcp] variants of TCP have been successfully deployed. However, these are now approaching their scaling limits.

For instance, we will consider a scenario with a maximum RTT of 30 ms at the peak of each sawtooth. As Reno packet rate scales 8x from 1,250 to 10,000 packet/s (from 15 to 120 Mb/s with 1500 B packets), the time to recover from a congestion event rises proportionately by 8x as well, from 422 ms to 3.38 s. It is clearly problematic for a congestion control to take multiple seconds to recover from each congestion event. Cubic [RFC8312] was developed to be less unscalable, but it is approaching its scaling limit; with the same max RTT of 30 ms, at 120 Mb/s Cubic is still fully in its Reno-friendly mode, so it takes about 4.3 s to recover. However, once the flow rate scales by 8x again to 960 Mb/s it enters true Cubic mode, with a recovery time of 12.2 s. From then on, each further scaling by 8x doubles Cubic's recovery time (because the cube root of 8 is 2), e.g. at 7.68 Gb/s the recovery time is 24.3 s. In contrast a scalable congestion control like DCTCP or TCP Prague induces 2 congestion signals per round trip on average, which remains invariant for any flow rate, keeping dynamic control very tight.

For a feel of where the global average lone-flow download sits on this scale at the time of writing (2021), according to [BDPdata] globally averaged fixed access capacity was 103 Mb/s in 2020 and averaged base RTT to a CDN was 25-34ms in 2019. Averaging of per-country data was weighted by Internet user population (data collected globally is necessarily of variable quality, but the paper does double-check that the outcome compares well against a second source). So a lone CUBIC flow would at best take about 200 round trips (5 s) to recover from each of its sawtooth reductions, if the flow even lasted that long. This is described as 'at best' because it assume everyone uses an AQM, whereas in reality most users still have a (probably bloated) tail-drop buffer. In the tail-drop case, likely average recovery time would be at least 4x 5 s, if not more, because RTT under load would be at least double that of an AQM, and recovery time depends on the square of RTT.

Although work on scaling congestion controls tends to start with TCP as the transport, the above is not intended to exclude other transports (e.g. SCTP, QUIC) or less elastic algorithms (e.g. RMCAT), which all tend to adopt the same or similar developments.

5.2. What L4S adds to Existing Approaches

All the following approaches address some part of the same problem space as L4S. In each case, it is shown that L4S complements them or improves on them, rather than being a mutually exclusive alternative:

Diffserv: Diffserv addresses the problem of bandwidth apportionment for important traffic as well as queuing latency for delay-sensitive traffic. Of these, L4S solely addresses the problem of queuing latency. Diffserv will still be necessary where important traffic requires priority (e.g. for commercial reasons, or for protection of critical infrastructure traffic) - see [I-D.briscoe-tsvwg-l4s-diffserv]. Nonetheless, the L4S approach can provide low latency for all traffic within each Diffserv class (including the case where there is only the one default Diffserv class).

Also, Diffserv can only provide a latency benefit if a small subset of the traffic on a bottleneck link requests low latency. As already explained, it has no effect when all the applications in use at one time at a single site (home, small business or mobile device) require low latency. In contrast, because L4S works for all traffic, it needs none of the management baggage (traffic policing, traffic contracts) associated with favouring some packets over others. This lack of management baggage ought to give L4S a better chance of end-to-end deployment.

In particular, because networks tend not to trust end systems to identify which packets should be favoured over others, where networks assign packets to Diffserv classes they tend to use packet inspection of application flow identifiers or deeper inspection of application signatures. Thus, nowadays, Diffserv doesn't always sit well with encryption of the layers above IP [RFC8404]. So users have to choose between privacy and QoS.

As with Diffserv, the L4S identifier is in the IP header. But, in contrast to Diffserv, the L4S identifier does not convey a want or a need for a certain level of quality. Rather, it promises a certain behaviour (scalable congestion response), which networks can objectively verify if they need to. This is because low delay depends on collective host behaviour, whereas bandwidth priority depends on network behaviour.

State-of-the-art AQMs: AQMs such as PIE and FQ-CoDel give a significant reduction in queuing delay relative to no AQM at all. L4S is intended to complement these AQMs, and should not distract from the need to deploy them as widely as possible. Nonetheless, AQMs alone cannot reduce queuing delay too far without significantly reducing link utilization, because the root cause of the problem is on the host - where Classic congestion controls use large saw-toothed rate variations. The L4S approach resolves this tension between delay and utilization by enabling hosts to minimize the amplitude of their sawteeth. A single-queue Classic AQM is not sufficient to allow hosts to use small sawteeth for two reasons: i) smaller sawteeth would not get lower delay in an AQM designed for larger amplitude Classic sawteeth, because a queue can only have one length at a time; and ii) much smaller sawteeth implies much more frequent sawteeth, so L4S flows would drive a Classic AQM into a high level of ECN-marking, which would appear as heavy congestion to Classic flows, which in turn would greatly reduce their rate as a result (see Section 6.4.4).

Per-flow queuing or marking: Similarly, per-flow approaches such as FQ-CoDel or Approx Fair CoDel [AFCD] are not incompatible with the L4S approach. However, per-flow queuing alone is not enough - it only isolates the queuing of one flow from others; not from itself. Per-flow implementations need to have support for scalable congestion control added, which has already been done for FQ-CoDel in Linux (see Sec.5.2.7 of [RFC8290] and [FQ_CoDel_Thresh]). Without this simple modification, per-flow AQMs like FQ-CoDel would still not be able to support applications that need both very low delay and high bandwidth, e.g. video-based control of remote procedures, or interactive cloud-based video (see Note 1 below).

Although per-flow techniques are not incompatible with L4S, it is important to have the DualQ alternative. This is because handling end-to-end (layer 4) flows in the network (layer 3 or 2) precludes some important end-to-end functions. For instance:

- a. Per-flow forms of L4S like FQ-CoDel are incompatible with full end-to-end encryption of transport layer identifiers for privacy and confidentiality (e.g. IPSec or encrypted VPN tunnels, as opposed to TLS over UDP), because they require packet inspection to access the end-to-end transport flow identifiers.

In contrast, the DualQ form of L4S requires no deeper inspection than the IP layer. So, as long as operators take the DualQ approach, their users can have both very low queuing delay and full end-to-end encryption [RFC8404].

- b. With per-flow forms of L4S, the network takes over control of the relative rates of each application flow. Some see it as an advantage that the network will prevent some flows running faster than others. Others consider it an inherent part of the Internet's appeal that applications can control their rate while taking account of the needs of others via congestion signals. They maintain that this has allowed applications with interesting rate behaviours to evolve, for instance, variable bit-rate video that varies around an equal share rather than being forced to remain equal at every instant, or e2e scavenger behaviours [RFC6817] that use less than an equal share of capacity [LEDBAT_AQM].

The L4S architecture does not require the IETF to commit to one approach over the other, because it supports both, so that the 'market' can decide. Nonetheless, in the spirit of 'Do one thing and do it well' [McIlroy78], the DualQ option provides low delay without prejudging the issue of flow-rate control. Then, flow rate policing can be added separately if desired. This allows application control up to a point, but the network can still choose to set the point at which it intervenes to prevent one flow completely starving another.

Note:

1. It might seem that self-inflicted queuing delay within a per-flow queue should not be counted, because if the delay wasn't in the network it would just shift to the sender. However, modern adaptive applications, e.g. HTTP/2 [RFC7540] or some interactive media applications (see Section 6.1), can keep low latency objects at the front of their local send queue by shuffling priorities of other objects dependent on the progress of other transfers (for example see [lowat]). They cannot shuffle objects once they have released them into the network.

Alternative Back-off ECN (ABE): Here again, L4S is not an alternative to ABE but a complement that introduces much lower queuing delay. ABE [RFC8511] alters the host behaviour in response to ECN marking to utilize a link better and give ECN flows faster throughput. It uses ECT(0) and assumes the network still treats ECN and drop the same. Therefore ABE exploits any lower queuing delay that AQMs can provide. But as explained above, AQMs still cannot reduce queuing delay too far without losing link utilization (to allow for other, non-ABE, flows).

BBR: Bottleneck Bandwidth and Round-trip propagation time

(BBR [I-D.cardwell-iccr-g-bbr-congestion-control]) controls queuing delay end-to-end without needing any special logic in the network, such as an AQM. So it works pretty-much on any path. BBR keeps queuing delay reasonably low, but perhaps not quite as low as with state-of-the-art AQMs such as PIE or FQ-CoDel, and certainly nowhere near as low as with L4S. Queuing delay is also not consistently low, due to BBR's regular bandwidth probing spikes and its aggressive flow start-up phase.

L4S complements BBR. Indeed BBRv2 can use L4S ECN where available and a scalable L4S congestion control behaviour in response to any ECN signalling from the path [BBRv2]. The L4S ECN signal complements the delay based congestion control aspects of BBR with an explicit indication that hosts can use, both to converge on a fair rate and to keep below a shallow queue target set by the network. Without L4S ECN, both these aspects need to be assumed or estimated.

6. Applicability

6.1. Applications

A transport layer that solves the current latency issues will provide new service, product and application opportunities.

With the L4S approach, the following existing applications also experience significantly better quality of experience under load:

- * Gaming, including cloud based gaming;
- * VoIP;
- * Video conferencing;
- * Web browsing;
- * (Adaptive) video streaming;
- * Instant messaging.

The significantly lower queuing latency also enables some interactive application functions to be offloaded to the cloud that would hardly even be usable today:

- * Cloud based interactive video;
- * Cloud based virtual and augmented reality.

The above two applications have been successfully demonstrated with L4S, both running together over a 40 Mb/s broadband access link loaded up with the numerous other latency sensitive applications in the previous list as well as numerous downloads - all sharing the same bottleneck queue simultaneously [L4Sdemo16]. For the former, a panoramic video of a football stadium could be swiped and pinched so that, on the fly, a proxy in the cloud could generate a sub-window of the match video under the finger-gesture control of each user. For the latter, a virtual reality headset displayed a viewport taken from a 360 degree camera in a racing car. The user's head movements controlled the viewport extracted by a cloud-based proxy. In both cases, with 7 ms end-to-end base delay, the additional queuing delay of roughly 1 ms was so low that it seemed the video was generated locally.

Using a swiping finger gesture or head movement to pan a video are extremely latency-demanding actions -- far more demanding than VoIP. Because human vision can detect extremely low delays of the order of single milliseconds when delay is translated into a visual lag between a video and a reference point (the finger or the orientation of the head sensed by the balance system in the inner ear -- the vestibular system).

Without the low queuing delay of L4S, cloud-based applications like these would not be credible without significantly more access bandwidth (to deliver all possible video that might be viewed) and more local processing, which would increase the weight and power consumption of head-mounted displays. When all interactive processing can be done in the cloud, only the data to be rendered for the end user needs to be sent.

Other low latency high bandwidth applications such as:

- * Interactive remote presence;
- * Video-assisted remote control of machinery or industrial processes.

are not credible at all without very low queuing delay. No amount of extra access bandwidth or local processing can make up for lost time.

6.2. Use Cases

The following use-cases for L4S are being considered by various interested parties:

- * Where the bottleneck is one of various types of access network:
e.g. DSL, Passive Optical Networks (PON), DOCSIS cable, mobile, satellite (see Section 6.3 for some technology-specific details)
- * Private networks of heterogeneous data centres, where there is no single administrator that can arrange for all the simultaneous changes to senders, receivers and network needed to deploy DCTCP:
 - a set of private data centres interconnected over a wide area with separate administrations, but within the same company
 - a set of data centres operated by separate companies interconnected by a community of interest network (e.g. for the finance sector)
 - multi-tenant (cloud) data centres where tenants choose their operating system stack (Infrastructure as a Service - IaaS)
- * Different types of transport (or application) congestion control:
 - elastic (TCP/SCTP);
 - real-time (RTP, RMCAT);
 - query (DNS/LDAP).
- * Where low delay quality of service is required, but without inspecting or intervening above the IP layer [RFC8404]:
 - mobile and other networks have tended to inspect higher layers in order to guess application QoS requirements. However, with growing demand for support of privacy and encryption, L4S offers an alternative. There is no need to select which traffic to favour for queuing, when L4S can give favourable queuing to all traffic.
- * If queuing delay is minimized, applications with a fixed delay budget can communicate over longer distances, or via a longer chain of service functions [RFC7665] or onion routers.
- * If delay jitter is minimized, it is possible to reduce the dejitter buffers on the receive end of video streaming, which should improve the interactive experience

6.3. Applicability with Specific Link Technologies

Certain link technologies aggregate data from multiple packets into bursts, and buffer incoming packets while building each burst. WiFi, PON and cable all involve such packet aggregation, whereas fixed Ethernet and DSL do not. No sender, whether L4S or not, can do anything to reduce the buffering needed for packet aggregation. So an AQM should not count this buffering as part of the queue that it controls, given no amount of congestion signals will reduce it.

Certain link technologies also add buffering for other reasons, specifically:

- * Radio links (cellular, WiFi, satellite) that are distant from the source are particularly challenging. The radio link capacity can vary rapidly by orders of magnitude, so it is considered desirable to hold a standing queue that can utilize sudden increases of capacity;
- * Cellular networks are further complicated by a perceived need to buffer in order to make hand-overs imperceptible;

L4S cannot remove the need for all these different forms of buffering. However, by removing 'the longest pole in the tent' (buffering for the large sawteeth of Classic congestion controls), L4S exposes all these 'shorter poles' to greater scrutiny.

Until now, the buffering needed for these additional reasons tended to be over-specified - with the excuse that none were 'the longest pole in the tent'. But having removed the 'longest pole', it becomes worthwhile to minimize them, for instance reducing packet aggregation burst sizes and MAC scheduling intervals.

6.4. Deployment Considerations

L4S AQMs, whether DualQ [I-D.ietf-tsvwg-aqm-dualq-coupled] or FQ, e.g. [RFC8290] are, in themselves, an incremental deployment mechanism for L4S - so that L4S traffic can coexist with existing Classic (Reno-friendly) traffic. Section 6.4.1 explains why only deploying an L4S AQM in one node at each end of the access link will realize nearly all the benefit of L4S.

L4S involves both end systems and the network, so Section 6.4.2 suggests some typical sequences to deploy each part, and why there will be an immediate and significant benefit after deploying just one part.

Section 6.4.3 and Section 6.4.4 describe the converse incremental deployment case where there is no L4S AQM at the network bottleneck, so any L4S flow traversing this bottleneck has to take care in case it is competing with Classic traffic.

6.4.1. Deployment Topology

L4S AQMs will not have to be deployed throughout the Internet before L4S can benefit anyone. Operators of public Internet access networks typically design their networks so that the bottleneck will nearly always occur at one known (logical) link. This confines the cost of queue management technology to one place.

The case of mesh networks is different and will be discussed later in this section. But the known bottleneck case is generally true for Internet access to all sorts of different 'sites', where the word 'site' includes home networks, small- to medium-sized campus or enterprise networks and even cellular devices (Figure 2). Also, this known-bottleneck case tends to be applicable whatever the access link technology; whether xDSL, cable, PON, cellular, line of sight wireless or satellite.

Therefore, the full benefit of the L4S service should be available in the downstream direction when an L4S AQM is deployed at the ingress to this bottleneck link. And similarly, the full upstream service will be available once an L4S AQM is deployed at the ingress into the upstream link. (Of course, multi-homed sites would only see the full benefit once all their access links were covered.)

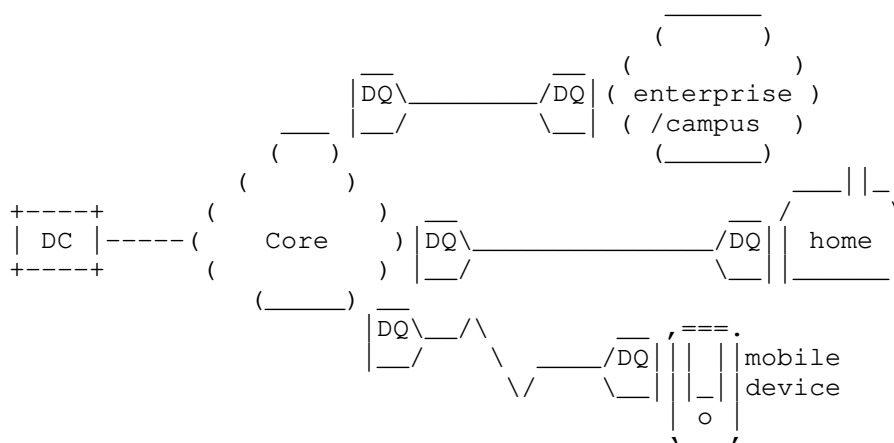


Figure 2: Likely location of DualQ (DQ) Deployments in common access topologies

Deployment in mesh topologies depends on how overbooked the core is. If the core is non-blocking, or at least generously provisioned so that the edges are nearly always the bottlenecks, it would only be necessary to deploy an L4S AQM at the edge bottlenecks. For example, some data-centre networks are designed with the bottleneck in the hypervisor or host NICs, while others bottleneck at the top-of-rack switch (both the output ports facing hosts and those facing the core).

An L4S AQM would often next be needed where the WiFi links in a home sometimes become the bottleneck. And an L4S AQM would eventually also need to be deployed at any other persistent bottlenecks such as network interconnections, e.g. some public Internet exchange points and the ingress and egress to WAN links interconnecting data-centres.

6.4.2. Deployment Sequences

For any one L4S flow to provide benefit, it requires three (or sometimes two) parts to have been deployed: i) the congestion control at the sender; ii) the AQM at the bottleneck; and iii) older transports (namely TCP) need upgraded receiver feedback too. This was the same deployment problem that ECN faced [RFC8170] so we have learned from that experience.

Firstly, L4S deployment exploits the fact that DCTCP already exists on many Internet hosts (Windows, FreeBSD and Linux); both servers and clients. Therefore, an L4S AQM can be deployed at a network bottleneck to immediately give a working deployment of all the L4S parts for testing, as long as the ECT(0) codepoint is switched to ECT(1). DCTCP needs some safety concerns to be fixed for general use over the public Internet (see Section 4.3 of the L4S ECN spec [I-D.ietf-tsvwg-ecn-l4s-id]), but DCTCP is not on by default, so these issues can be managed within controlled deployments or controlled trials.

Secondly, the performance improvement with L4S is so significant that it enables new interactive services and products that were not previously possible. It is much easier for companies to initiate new work on deployment if there is budget for a new product trial. If, in contrast, there were only an incremental performance improvement (as with Classic ECN), spending on deployment tends to be much harder to justify.

Thirdly, the L4S identifier is defined so that initially network operators can enable L4S exclusively for certain customers or certain applications. But this is carefully defined so that it does not compromise future evolution towards L4S as an Internet-wide service. This is because the L4S identifier is defined not only as the end-to-

end ECN field, but it can also optionally be combined with any other packet header or some status of a customer or their access link (see section 5.4 of [I-D.ietf-tsvwg-ecn-l4s-id]). Operators could do this anyway, even if it were not blessed by the IETF. However, it is best for the IETF to specify that, if they use their own local identifier, it must be in combination with the IETF's identifier. Then, if an operator has opted for an exclusive local-use approach, later they only have to remove this extra rule to make the service work Internet-wide - it will already traverse middleboxes, peerings, etc.

	Servers or proxies	Access link	Clients
0	DCTCP (existing)		DCTCP (existing)
1	Add L4S AQM downstream WORKS DOWNSTREAM FOR CONTROLLED DEPLOYMENTS/TRIALS		
2	Upgrade DCTCP to TCP Prague	FULLY WORKS DOWNSTREAM	Replace DCTCP feedb'k with AccECN
3		Add L4S AQM upstream	Upgrade DCTCP to TCP Prague
	FULLY WORKS UPSTREAM AND DOWNSTREAM		

Figure 3: Example L4S Deployment Sequence

Figure 3 illustrates some example sequences in which the parts of L4S might be deployed. It consists of the following stages:

1. Here, the immediate benefit of a single AQM deployment can be seen, but limited to a controlled trial or controlled deployment. In this example downstream deployment is first, but in other scenarios the upstream might be deployed first. If no AQM at all was previously deployed for the downstream access, an L4S AQM greatly improves the Classic service (as well as adding the L4S service). If an AQM was already deployed, the Classic service will be unchanged (and L4S will add an improvement on top).
2. In this stage, the name 'TCP Prague' [I-D.briscoe-iccrp-prague-congestion-control] is used to represent a variant of DCTCP that is designed to be used in a production Internet environment (assuming it complies with the requirements in Section 4 of the L4S ECN spec [I-D.ietf-tsvwg-ecn-l4s-id]). If the application is

primarily unidirectional, 'TCP Prague' at one end will provide all the benefit needed. For TCP transports, Accurate ECN feedback (AccECN) [I-D.ietf-tcpm-accurate-ecn] is needed at the other end, but it is a generic ECN feedback facility that is already planned to be deployed for other purposes, e.g. DCTCP, BBR. The two ends can be deployed in either order, because, in TCP, an L4S congestion control only enables itself if it has negotiated the use of AccECN feedback with the other end during the connection handshake. Thus, deployment of TCP Prague on a server enables L4S trials to move to a production service in one direction, wherever AccECN is deployed at the other end. This stage might be further motivated by the performance improvements of TCP Prague relative to DCTCP (see Appendix A.2 of the L4S ECN spec [I-D.ietf-tsvwg-ecn-l4s-id]).

Unlike TCP, from the outset, QUIC ECN feedback [RFC9000] has supported L4S. Therefore, if the transport is QUIC, one-ended deployment of a Prague congestion control at this stage is simple and sufficient.

3. This is a two-move stage to enable L4S upstream. An L4S AQM or TCP Prague can be deployed in either order as already explained. To motivate the first of two independent moves, the deferred benefit of enabling new services after the second move has to be worth it to cover the first mover's investment risk. As explained already, the potential for new interactive services provides this motivation. An L4S AQM also improves the upstream Classic service - significantly if no other AQM has already been deployed.

Note that other deployment sequences might occur. For instance: the upstream might be deployed first; a non-TCP protocol might be used end-to-end, e.g. QUIC, RTP; a body such as the 3GPP might require L4S to be implemented in 5G user equipment, or other random acts of kindness.

6.4.3. L4S Flow but Non-ECN Bottleneck

If L4S is enabled between two hosts, the L4S sender is required to coexist safely with Reno in response to any drop (see Section 4.3 of the L4S ECN spec [I-D.ietf-tsvwg-ecn-l4s-id]).

Unfortunately, as well as protecting Classic traffic, this rule degrades the L4S service whenever there is any loss, even if the cause is not persistent congestion at a bottleneck, e.g.:

- * congestion loss at other transient bottlenecks, e.g. due to bursts in shallower queues;

- * transmission errors, e.g. due to electrical interference;
- * rate policing.

Three complementary approaches are in progress to address this issue, but they are all currently research:

- * In Prague congestion control, ignore certain losses deemed unlikely to be due to congestion (using some ideas from BBR [I-D.cardwell-iccr-g-bbr-congestion-control] regarding isolated losses). This could mask any of the above types of loss while still coexisting with drop-based congestion controls.
- * A combination of RACK, L4S and link retransmission without resequencing could repair transmission errors without the head of line blocking delay usually associated with link-layer retransmission [UnorderedLTE], [I-D.ietf-tsvwg-ecn-l4s-id];
- * Hybrid ECN/drop rate policers (see Section 8.3).

L4S deployment scenarios that minimize these issues (e.g. over wireline networks) can proceed in parallel to this research, in the expectation that research success could continually widen L4S applicability.

6.4.4. L4S Flow but Classic ECN Bottleneck

Classic ECN support is starting to materialize on the Internet as an increased level of CE marking. It is hard to detect whether this is all due to the addition of support for ECN in implementations of FQ-CoDel and/or FQ-COBALT, which is not generally problematic, because flow-queue (FQ) scheduling inherently prevents a flow from exceeding the 'fair' rate irrespective of its aggressiveness. However, some of this Classic ECN marking might be due to single-queue ECN deployment. This case is discussed in Section 4.3 of the L4S ECN spec [I-D.ietf-tsvwg-ecn-l4s-id].

6.4.5. L4S AQM Deployment within Tunnels

An L4S AQM uses the ECN field to signal congestion. So, in common with Classic ECN, if the AQM is within a tunnel or at a lower layer, correct functioning of ECN signalling requires correct propagation of the ECN field up the layers [RFC6040], [I-D.ietf-tsvwg-rfc6040update-shim], [I-D.ietf-tsvwg-ecn-encap-guidelines].

7. IANA Considerations (to be removed by RFC Editor)

This specification contains no IANA considerations.

8. Security Considerations

8.1. Traffic Rate (Non-)Policing

In the current Internet, scheduling usually enforces separation between 'sites' (e.g. households, businesses or mobile users [RFC0970]) and various techniques like redirection to traffic scrubbing facilities deal with flooding attacks. However, there has never been a universal need to police the rate of individual application flows - the Internet has generally always relied on self-restraint of congestion controls at senders for sharing intra-'site' capacity.

As explained in Section 5.2, the DualQ variant of L4S provides low delay without prejudging the issue of flow-rate control. Then, if flow-rate control is needed, per-flow-queuing (FQ) can be used instead, or flow rate policing can be added as a modular addition to a DualQ.

Because the L4S service reduces delay without increasing the delay of Classic traffic, it should not be necessary to rate-police access to the L4S service. In contrast, Section 5.2 explains how Diffserv only makes a difference if some packets get less favourable treatment than others, which typically requires traffic rate policing, which can, in turn, lead to further complexity such as traffic contracts at trust boundaries. Because L4S avoids this management complexity, it is more likely to work end-to-end.

During early deployment (and perhaps always), some networks will not offer the L4S service. In general, these networks should not need to police L4S traffic. They are required (by both the ECN spec [RFC3168] and the L4S ECN spec [I-D.ietf-tsvwg-ecn-l4s-id]) not to change the L4S identifier, which would interfere with end-to-end congestion control. Instead they can merely treat L4S traffic as Not-ECT, as they might already treat all ECN traffic today. At a bottleneck, such networks will introduce some queuing and dropping. When a scalable congestion control detects a drop it will have to respond safely with respect to Classic congestion controls (as required in Section 4.3 of [I-D.ietf-tsvwg-ecn-l4s-id]). This will degrade the L4S service to be no better (but never worse) than Classic best efforts, whenever a non-ECN bottleneck is encountered on a path (see Section 6.4.3).

In cases that are expected to be rare, networks that solely support Classic ECN [RFC3168] in a single queue bottleneck might opt to police L4S traffic so as to protect competing Classic ECN traffic (for instance, see Section 6.1.3 of the L4S operational guidance [I-D.ietf-tsvwg-l4sops]). However, Section 4.3 of the L4S ECN spec [I-D.ietf-tsvwg-ecn-l4s-id] recommends that the sender adapts its congestion response to properly coexist with Classic ECN flows, i.e. reverting to the self-restraint approach.

Certain network operators might choose to restrict access to the L4S class, perhaps only to selected premium customers as a value-added service. Their packet classifier (item 2 in Figure 1) could identify such customers against some other field (e.g. source address range) as well as classifying on the ECN field. If only the ECN L4S identifier matched, but not the source address (say), the classifier could direct these packets (from non-premium customers) into the Classic queue. Explaining clearly how operators can use an additional local classifiers (see section 5.4 of the L4S ECN spec [I-D.ietf-tsvwg-ecn-l4s-id]) is intended to remove any motivation to clear the L4S identifier. Then at least the L4S ECN identifier will be more likely to survive end-to-end even though the service may not be supported at every hop. Such local arrangements would only require simple registered/not-registered packet classification, rather than the managed, application-specific traffic policing against customer-specific traffic contracts that Diffserv uses.

8.2. 'Latency Friendliness'

Like the Classic service, the L4S service relies on self-restraint - limiting rate in response to congestion. In addition, the L4S service requires self-restraint in terms of limiting latency (burstiness). It is hoped that self-interest and guidance on dynamic behaviour (especially flow start-up, which might need to be standardized) will be sufficient to prevent transports from sending excessive bursts of L4S traffic, given the application's own latency will suffer most from such behaviour.

Whether burst policing becomes necessary remains to be seen. Without it, there will be potential for attacks on the low latency of the L4S service.

If needed, various arrangements could be used to address this concern:

Local bottleneck queue protection: A per-flow (5-tuple) queue

protection function [I-D.briscoe-docsis-q-protection] has been developed for the low latency queue in DOCSIS, which has adopted the DualQ L4S architecture. It protects the low latency service from any queue-building flows that accidentally or maliciously classify themselves into the low latency queue. It is designed to score flows based solely on their contribution to queuing (not flow rate in itself). Then, if the shared low latency queue is at risk of exceeding a threshold, the function redirects enough packets of the highest scoring flow(s) into the Classic queue to preserve low latency.

Distributed traffic scrubbing: Rather than policing locally at each bottleneck, it may only be necessary to address problems reactively, e.g. punitively target any deployments of new bursty malware, in a similar way to how traffic from flooding attack sources is rerouted via scrubbing facilities.

Local bottleneck per-flow scheduling: Per-flow scheduling should inherently isolate non-bursty flows from bursty (see Section 5.2 for discussion of the merits of per-flow scheduling relative to per-flow policing).

Distributed access subnet queue protection: Per-flow queue protection could be arranged for a queue structure distributed across a subnet inter-communicating using lower layer control messages (see Section 2.1.4 of [QDyn]). For instance, in a radio access network, user equipment already sends regular buffer status reports to a radio network controller, which could use this information to remotely police individual flows.

Distributed Congestion Exposure to Ingress Policers: The Congestion Exposure (ConEx) architecture [RFC7713] which uses egress audit to motivate senders to truthfully signal path congestion in-band where it can be used by ingress policers. An edge-to-edge variant of this architecture is also possible.

Distributed Domain-edge traffic conditioning: An architecture similar to Diffserv [RFC2475] may be preferred, where traffic is proactively conditioned on entry to a domain, rather than reactively policed only if it leads to queuing once combined with other traffic at a bottleneck.

Distributed core network queue protection: The policing function

could be divided between per-flow mechanisms at the network ingress that characterize the burstiness of each flow into a signal carried with the traffic, and per-class mechanisms at bottlenecks that act on these signals if queuing actually occurs once the traffic converges. This would be somewhat similar to [Nadas20], which is in turn similar to the idea behind core stateless fair queuing.

None of these possible queue protection capabilities are considered a necessary part of the L4S architecture, which works without them (in a similar way to how the Internet works without per-flow rate policing). Indeed, even where latency policers are deployed, under normal circumstances they would not intervene, and if operators found they were not necessary they could disable them. Part of the L4S experiment will be to see whether such a function is necessary, and which arrangements are most appropriate to the size of the problem.

8.3. Interaction between Rate Policing and L4S

As mentioned in Section 5.2, L4S should remove the need for low latency Diffserv classes. However, those Diffserv classes that give certain applications or users priority over capacity, would still be applicable in certain scenarios (e.g. corporate networks). Then, within such Diffserv classes, L4S would often be applicable to give traffic low latency and low loss as well. Within such a Diffserv class, the bandwidth available to a user or application is often limited by a rate policer. Similarly, in the default Diffserv class, rate policers are used to partition shared capacity.

A classic rate policer drops any packets exceeding a set rate, usually also giving a burst allowance (variants exist where the policer re-marks non-compliant traffic to a discard-eligible Diffserv codepoint, so they can be dropped elsewhere during contention). Whenever L4S traffic encounters one of these rate policers, it will experience drops and the source will have to fall back to a Classic congestion control, thus losing the benefits of L4S (Section 6.4.3). So, in networks that already use rate policers and plan to deploy L4S, it will be preferable to redesign these rate policers to be more friendly to the L4S service.

L4S-friendly rate policing is currently a research area (note that this is not the same as latency policing). It might be achieved by setting a threshold where ECN marking is introduced, such that it is just under the policed rate or just under the burst allowance where drop is introduced. For instance the two-rate three-colour marker [RFC2698] or a PCN threshold and excess-rate marker [RFC5670] could mark ECN at the lower rate and drop at the higher. Or an existing rate policer could have congestion-rate policing added,

e.g. using the 'local' (non-ConEx) variant of the ConEx aggregate congestion policer [I-D.briscoe-conex-policing]. It might also be possible to design scalable congestion controls to respond less catastrophically to loss that has not been preceded by a period of increasing delay.

The design of L4S-friendly rate policers will require a separate dedicated document. For further discussion of the interaction between L4S and Diffserv, see [I-D.briscoe-tsvwg-l4s-diffserv].

8.4. ECN Integrity

Receiving hosts can fool a sender into downloading faster by suppressing feedback of ECN marks (or of losses if retransmissions are not necessary or available otherwise). Various ways to protect transport feedback integrity have been developed. For instance:

- * The sender can test the integrity of the receiver's feedback by occasionally setting the IP-ECN field to the congestion experienced (CE) codepoint, which is normally only set by a congested link. Then the sender can test whether the receiver's feedback faithfully reports what it expects (see 2nd para of Section 20.2 of the Classic ECN spec [RFC3168]).
- * A network can enforce a congestion response to its ECN markings (or packet losses) by auditing congestion exposure (ConEx) [RFC7713].
- * Transport layer authentication such as the TCP authentication option (TCP-AO [RFC5925]) or QUIC's use of TLS [RFC9001] can detect any tampering with congestion feedback.
- * The ECN Nonce [RFC3540] was proposed to detect tampering with congestion feedback, but it has been reclassified as historic [RFC8311].

Appendix C.1 of the L4S ECN spec [I-D.ietf-tsvwg-ecn-l4s-id] gives more details of these techniques including their applicability and pros and cons.

8.5. Privacy Considerations

As discussed in Section 5.2, the L4S architecture does not preclude approaches that inspect end-to-end transport layer identifiers. For instance, L4S support has been added to FQ-CoDel, which classifies by application flow ID in the network. However, the main innovation of L4S is the DualQ AQM framework that does not need to inspect any deeper than the outermost IP header, because the L4S identifier is in

the IP-ECN field.

Thus, the L4S architecture enables very low queuing delay without requiring inspection of information above the IP layer. This means that users who want to encrypt application flow identifiers, e.g. in IPSec or other encrypted VPN tunnels, don't have to sacrifice low delay [RFC8404].

Because L4S can provide low delay for a broad set of applications that choose to use it, there is no need for individual applications or classes within that broad set to be distinguishable in any way while traversing networks. This removes much of the ability to correlate between the delay requirements of traffic and other identifying features [RFC6973]. There may be some types of traffic that prefer not to use L4S, but the coarse binary categorization of traffic reveals very little that could be exploited to compromise privacy.

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A Lower Effort Per-Hop Behavior (LE PHB) for Differentiated Services
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Abstract

This document specifies properties and characteristics of a Lower Effort (LE) per-hop behavior (PHB). The primary objective of this LE PHB is to protect best-effort (BE) traffic (packets forwarded with the default PHB) from LE traffic in congestion situations, i.e., when resources become scarce, best-effort traffic has precedence over LE traffic and may preempt it. Alternatively, packets forwarded by the LE PHB can be associated with a scavenger service class, i.e., they scavenge otherwise unused resources only. There are numerous uses for this PHB, e.g., for background traffic of low precedence, such as bulk data transfers with low priority in time, non time-critical backups, larger software updates, web search engines while gathering information from web servers and so on. This document recommends a standard DSCP value for the LE PHB. This specification obsoletes RFC 3662 and updates the DSCP recommended in RFC 4594 and RFC 8325 to use the DSCP assigned in this specification.

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1. Introduction

This document defines a Differentiated Services per-hop behavior [RFC2474] called "Lower Effort" (LE), which is intended for traffic of sufficiently low urgency that all other traffic takes precedence over the LE traffic in consumption of network link bandwidth. Low urgency traffic has a low priority for timely forwarding, which does not necessarily imply that it is generally of minor importance. From this viewpoint, it can be considered as a network equivalent to a background priority for processes in an operating system. There may or may not be memory (buffer) resources allocated for this type of traffic.

Some networks carry packets that ought to consume network resources only when no other traffic is demanding them. In this point of view, packets forwarded by the LE PHB scavenge otherwise unused resources only, which led to the name "scavenger service" in early Internet2 deployments (see Appendix A). Other commonly used names for LE PHB type services are "Lower-than-best-effort" or "Less-than-best-effort". In summary, with the mentioned feature above, the LE PHB has two important properties: it should scavenge residual capacity and it must be preemptable by the default PHB (or other elevated PHBs) in case they need more resources. Consequently, the effect of this type of traffic on all other network traffic is strictly limited ("no harm" property). This is distinct from "best-effort" (BE) traffic since the network makes no commitment to deliver LE packets. In contrast, BE traffic receives an implied "good faith" commitment of at least some available network resources. This document proposes a Lower Effort Differentiated Services per-hop behavior (LE PHB) for handling this "optional" traffic in a differentiated services node.

2. Requirements Language

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119][RFC8174] when, and only when, they appear in all capitals, as shown here.

3. Applicability

A Lower Effort PHB is applicable for many applications that otherwise use best-effort delivery. More specifically, it is suitable for traffic and services that can tolerate strongly varying throughput

for their data flows, especially periods of very low throughput or even starvation (i.e., long interruptions due to significant or even complete packet loss). Therefore, an application sending an LE marked flow needs to be able to tolerate short or (even very) long interruptions due to the presence of severe congestion conditions during the transmission of the flow. Thus, there ought to be an expectation that packets of the LE PHB could be excessively delayed or dropped when any other traffic is present. It is application-dependent when a lack of progress is considered being a failure (e.g., if a transport connection fails due to timing out, the application may try several times to re-establish the transport connection in order to resume the application session before finally giving up). The LE PHB is suitable for sending traffic of low urgency across a Differentiated Services (DS) domain or DS region.

Just like best-effort traffic, LE traffic SHOULD be congestion controlled (i.e., use a congestion controlled transport or implement an appropriate congestion control method [RFC2914] [RFC8085]). Since LE traffic could be starved completely for a longer period of time, transport protocols or applications (and their related congestion control mechanisms) SHOULD be able to detect and react to such a starvation situation. An appropriate reaction would be to resume the transfer instead of aborting it, i.e., an LE optimized transport ought to use appropriate retry strategies (e.g., exponential back-off with an upper bound) as well as corresponding retry and timeout limits in order to avoid the loss of the connection due to the mentioned starvation periods. While it is desirable to achieve a quick resumption of the transfer as soon as resources become available again, it may be difficult to achieve this in practice. In lack of a transport protocol and congestion control that are adapted to LE, applications can also use existing common transport protocols and implement session resumption by trying to re-establish failed connections. Congestion control is not only useful to let the flows within the LE behavior aggregate adapt to the available bandwidth that may be highly fluctuating, but is also essential if LE traffic is mapped to the default PHB in DS domains that do not support LE. In this case, use of background transport protocols, e.g., similar to LEDBAT [RFC6817], is expedient.

Use of the LE PHB might assist a network operator in moving certain kinds of traffic or users to off-peak times. Furthermore, packets can be designated for the LE PHB when the goal is to protect all other packet traffic from competition with the LE aggregate while not completely banning LE traffic from the network. An LE PHB SHOULD NOT be used for a customer's "normal Internet" traffic and packets SHOULD NOT be "downgraded" to the LE PHB instead of being dropped, particularly when the packets are unauthorized traffic. The LE PHB

is expected to have applicability in networks that have at least some unused capacity at certain periods.

The LE PHB allows networks to protect themselves from selected types of traffic as a complement to giving preferential treatment to other selected traffic aggregates. LE ought not to be used for the general case of downgraded traffic, but could be used by design, e.g., to protect an internal network from untrusted external traffic sources. In this case there is no way for attackers to preempt internal (non LE) traffic by flooding. Another use case in this regard is forwarding of multicast traffic from untrusted sources. Multicast forwarding is currently enabled within domains only for specific sources within a domain, but not for sources from anywhere in the Internet. A major problem is that multicast routing creates traffic sources at (mostly) unpredictable branching points within a domain, potentially leading to congestion and packet loss. In the case of multicast traffic packets from untrusted sources are forwarded as LE traffic, they will not harm traffic from non-LE behavior aggregates. A further related use case is mentioned in [RFC3754]: preliminary forwarding of non-admitted multicast traffic.

There is no intrinsic reason to limit the applicability of the LE PHB to any particular application or type of traffic. It is intended as an additional traffic engineering tool for network administrators. For instance, it can be used to fill protection capacity of transmission links that is otherwise unused. Some network providers keep link utilization below 50% to ensure that all traffic is forwarded without loss after rerouting caused by a link failure (cf. Section 6 of [RFC3439]). LE marked traffic can utilize the normally unused capacity and will be preempted automatically in case of link failure when 100% of the link capacity is required for all other traffic. Ideally, applications mark their packets as LE traffic, since they know the urgency of flows. Since LE traffic may be starved for longer periods of time it is probably less suitable for real-time and interactive applications.

Example uses for the LE PHB:

- o For traffic caused by world-wide web search engines while they gather information from web servers.
- o For software updates or dissemination of new releases of operating systems.
- o For reporting errors or telemetry data from operating systems or applications.

- o For backup traffic or non-time critical synchronization or mirroring traffic.
- o For content distribution transfers between caches.
- o For preloading or prefetching objects from web sites.
- o For network news and other "bulk mail" of the Internet.
- o For "downgraded" traffic from some other PHB when this does not violate the operational objectives of the other PHB.
- o For multicast traffic from untrusted (e.g., non-local) sources.

4. PHB Description

The LE PHB is defined in relation to the default PHB (best-effort). A packet forwarded with the LE PHB SHOULD have lower precedence than packets forwarded with the default PHB, i.e., in the case of congestion, LE marked traffic SHOULD be dropped prior to dropping any default PHB traffic. Ideally, LE packets would be forwarded only when no packet with any other PHB is awaiting transmission. This means that in case of link resource contention LE traffic can be starved completely, which may not be always desired by the network operator's policy. The used scheduler to implement the LE PHB may reflect this policy accordingly.

A straightforward implementation could be a simple priority scheduler serving the default PHB queue with higher priority than the lower-effort PHB queue. Alternative implementations may use scheduling algorithms that assign a very small weight to the LE class. This, however, could sometimes cause better service for LE packets compared to BE packets in cases when the BE share is fully utilized and the LE share not.

If a dedicated LE queue is not available, an active queue management mechanism within a common BE/LE queue could also be used. This could drop all arriving LE packets as soon as certain queue length or sojourn time thresholds are exceeded.

Since congestion control is also useful within the LE traffic class, Explicit Congestion Notification (ECN) [RFC3168] SHOULD be used for LE packets, too. More specifically, an LE implementation SHOULD also apply CE marking for ECT marked packets and transport protocols used for LE SHOULD support and employ ECN. For more information on the benefits of using ECN see [RFC8087].

5. Traffic Conditioning Actions

If possible, packets SHOULD be pre-marked in DS-aware end systems by applications due to their specific knowledge about the particular precedence of packets. There is no incentive for DS domains to distrust this initial marking, because letting LE traffic enter a DS domain causes no harm. Thus, any policing such as limiting the rate of LE traffic is not necessary at the DS boundary.

As for most other PHBs an initial classification and marking can be also performed at the first DS boundary node according to the DS domain's own policies (e.g., as protection measure against untrusted sources). However, non-LE traffic (e.g., BE traffic) SHOULD NOT be remarked to LE. Remarketing traffic from another PHB results in that traffic being "downgraded". This changes the way the network treats this traffic and it is important not to violate the operational objectives of the original PHB. See also remarks with respect to downgrading in Section 3 and Section 8.

6. Recommended DS Codepoint

The RECOMMENDED codepoint for the LE PHB is '000001'.

Earlier specifications [RFC4594] recommended to use CS1 as codepoint (as mentioned in [RFC3662]). This is problematic since it may cause a priority inversion in Diffserv domains that treat CS1 as originally proposed in [RFC2474], resulting in forwarding LE packets with higher precedence than BE packets. Existing implementations SHOULD transition to use the unambiguous LE codepoint '000001' whenever possible.

This particular codepoint was chosen due to measurements on the currently observable DSCP remarking behavior in the Internet [ietf99-secchi]. Since some network domains set the former IP precedence bits to zero, it is possible that some other standardized DSCPs get mapped to the LE PHB DSCP if it were taken from the DSCP standards action pool 1 (xxxxx0).

7. Deployment Considerations

In order to enable LE support, DS nodes typically only need

- o A BA classifier (Behavior Aggregate classifier, see [RFC2475]) that classifies packets according to the LE DSCP
- o A dedicated LE queue
- o A suitable scheduling discipline, e.g., simple priority queueing

Alternatively, implementations could use active queue management mechanisms instead of a dedicated LE queue, e.g., dropping all arriving LE packets when certain queue length or sojourn time thresholds are exceeded.

Internet-wide deployment of the LE PHB is eased by the following properties:

- o No harm to other traffic: since the LE PHB has the lowest forwarding priority it does not consume resources from other PHBs. Deployment across different provider domains with LE support causes no trust issues or attack vectors to existing (non LE) traffic. Thus, providers can trust LE markings from end-systems, i.e., there is no need to police or remark incoming LE traffic.
- o No PHB parameters or configuration of traffic profiles: the LE PHB itself possesses no parameters that need to be set or configured. Similarly, since LE traffic requires no admission or policing, it is not necessary to configure traffic profiles.
- o No traffic conditioning mechanisms: the LE PHB requires no traffic meters, droppers, or shapers. See also Section 5 for further discussion.

Operators of DS domains that cannot or do not want to implement the LE PHB (e.g., because there is no separate LE queue available in the corresponding nodes) SHOULD NOT drop packets marked with the LE DSCP. They SHOULD map packets with this DSCP to the default PHB and SHOULD preserve the LE DSCP marking. DS domains operators that do not implement the LE PHB should be aware that they violate the "no harm" property of LE. See also Section 8 for further discussion of forwarding LE traffic with the default PHB instead.

8. Remarking to other DSCPs/PHBs

"DSCP bleaching", i.e., setting the DSCP to '000000' (default PHB) is NOT RECOMMENDED for this PHB. This may cause effects that are in contrast to the original intent in protecting BE traffic from LE traffic (no harm property). In the case that a DS domain does not support the LE PHB, its nodes SHOULD treat LE marked packets with the default PHB instead (by mapping the LE DSCP to the default PHB), but they SHOULD do so without remarking to DSCP '000000'. The reason for this is that later traversed DS domains may then have still the possibility to treat such packets according to the LE PHB.

Operators of DS domains that forward LE traffic within the BE aggregate need to be aware of the implications, i.e., induced congestion situations and quality-of-service degradation of the

original BE traffic. In this case, the LE property of not harming other traffic is no longer fulfilled. To limit the impact in such cases, traffic policing of the LE aggregate MAY be used.

In the case that LE marked packets are effectively carried within the default PHB (i.e., forwarded as best-effort traffic) they get a better forwarding treatment than expected. For some applications and services, it is favorable if the transmission is finished earlier than expected. However, in some cases it may be against the original intention of the LE PHB user to strictly send the traffic only if otherwise unused resources are available. In the case that LE traffic is mapped to the default PHB, LE traffic may compete with BE traffic for the same resources and thus adversely affect the original BE aggregate. Applications that want to ensure the lower precedence compared to BE traffic even in such cases SHOULD use additionally a corresponding Lower-than-Best-Effort transport protocol [RFC6297], e.g., LEDBAT [RFC6817].

A DS domain that still uses DSCP CS1 for marking LE traffic (including Low Priority-Data as defined in [RFC4594] or the old definition in [RFC3662]) SHOULD remark traffic to the LE DSCP '000001' at the egress to the next DS domain. This increases the probability that the DSCP is preserved end-to-end, whereas a CS1 marked packet may be remarked by the default DSCP if the next domain is applying Diffserv-Interconnection [RFC8100].

9. Multicast Considerations

Basically, the multicast considerations in [RFC3754] apply. However, using the Lower Effort PHB for multicast requires paying special attention to the way how packets get replicated inside routers. Due to multicast packet replication, resource contention may actually occur even before a packet is forwarded to its output port and in the worst case, these forwarding resources are missing for higher prioritized multicast or even unicast packets.

Several forward error correction coding schemes such as fountain codes (e.g., [RFC5053]) allow reliable data delivery even in environments with a potential high amount of packet loss in transmission. When used for example over satellite links or other broadcast media, this means that receivers that lose 80% of packets in transmission simply need 5 times as long to receive the complete data than those receivers experiencing no loss (without any receiver feedback required).

Superficially viewed, it may sound very attractive to use IP multicast with the LE PHB to build this type of opportunistic reliable distribution in IP networks, but it can only be usefully

deployed with routers that do not experience forwarding/replication resource starvation when a large amount of packets (virtually) need to be replicated to links where the LE queue is full.

Thus, packet replication of LE marked packets should consider the situation at the respective output links: it is a waste of internal forwarding resources if a packet is replicated to output links that have no resources left for LE forwarding. In those cases a packet would have been replicated just to be dropped immediately after finding a filled LE queue at the respective output port. Such behavior could be avoided for example by using a conditional internal packet replication: a packet would then only be replicated in case the output link is not fully used. This conditional replication, however, is probably not widely implemented.

While the resource contention problem caused by multicast packet replication is also true for other Diffserv PHBs, LE forwarding is special, because often it is assumed that LE packets only get forwarded in case of available resources at the output ports. The previously mentioned redundancy data traffic could nicely use the varying available residual bandwidth being utilized by LE PHB, but only if the specific requirements stated above for conditional replication in the internal implementation of the network devices are considered.

10. The Update to RFC 4594

[RFC4594] recommended to use CS1 as codepoint in section 4.10, whereas CS1 was defined in [RFC2474] to have a higher precedence than CS0, i.e., the default PHB. Consequently, Diffserv domains implementing CS1 according to [RFC2474] will cause a priority inversion for LE packets that contradicts with the original purpose of LE. Therefore, every occurrence of the CS1 DSCP is replaced by the LE DSCP.

Changes:

- o This update to RFC 4594 removes the following entry from figure 3:

Low-Priority Data	CS1	001000	Any flow that has no BW assurance
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and replaces this by the following entry:

Low-Priority Data	LE	000001	Any flow that has no BW assurance
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- o This update to RFC 4594 extends the Notes text below figure 3 that currently states "Notes for Figure 3: Default Forwarding (DF) and Class Selector 0 (CS0) provide equivalent behavior and use the same DS codepoint, '000000'." to state "Notes for Figure 3: Default Forwarding (DF) and Class Selector 0 (CS0) provide equivalent behavior and use the same DS codepoint, '000000'. The prior recommendation to use the CS1 DSCP for Low-Priority Data has been replaced by the current recommendation to use the LE DSCP, '000001'."
- o This update to RFC 4594 removes the following entry from figure 4:

Low-Priority Data	CS1	Not applicable	RFC3662	Rate	Yes
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and replaces this by the following entry:

Low-Priority Data	LE	Not applicable	RFCXXXX	Rate	Yes
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- o Section 2.3 of [RFC4594] specifies: "In network segments that use IP precedence marking, only one of the two service classes can be supported, High-Throughput Data or Low-Priority Data. We RECOMMEND that the DSCP value(s) of the unsupported service class be changed to 000xx1 on ingress and changed back to original value(s) on egress of the network segment that uses precedence marking. For example, if Low-Priority Data is mapped to Standard service class, then 000001 DSCP marking MAY be used to distinguish it from Standard marked packets on egress." This document removes this recommendation, because by using the herein defined LE DSCP such remarking is not necessary. So even if Low-Priority Data is unsupported (i.e., mapped to the default PHB) the LE DSCP should be kept across the domain as RECOMMENDED in Section 8. That removed text is replaced by: "In network segments that use IP Precedence marking, the Low-Priority Data service class receives the same Diffserv QoS as the Standard service class when the LE DSCP is used for Low-Priority Data traffic. This is acceptable behavior for the Low-Priority Data service class, although it is not the preferred behavior."

- o This document removes the following line of RFC 4594, Section 4.10: "The RECOMMENDED DSCP marking is CS1 (Class Selector 1)." and replaces this with the following text: "The RECOMMENDED DSCP marking is LE (Lower Effort), which replaces the prior recommendation for CS1 (Class Selector 1) marking."

11. The Update to RFC 8325

Section 4.2.10 of RFC 8325 [RFC8325] specifies "[RFC3662] and [RFC4594] both recommend Low-Priority Data be marked CS1 DSCP." which is updated to "[RFC3662] recommends that Low-Priority Data be marked CS1 DSCP. [RFC4594] as updated by [RFCXXXX] recommends Low-Priority Data be marked LE DSCP."

This document removes the following paragraph of RFC 8325, Section 4.2.10 because this document makes the anticipated change: "Note: This marking recommendation may change in the future, as [LE-PHB] defines a Lower Effort (LE) PHB for Low-Priority Data traffic and recommends an additional DSCP for this traffic."

Section 4.2.10 of RFC 8325 [RFC8325] specifies "therefore, it is RECOMMENDED to map Low-Priority Data traffic marked CS1 DSCP to UP 1" which is updated to "therefore, it is RECOMMENDED to map Low-Priority Data traffic marked with LE DSCP or legacy CS1 DSCP to UP 1"

This update to RFC 8325 replaces the following entry from figure 1:

Low-Priority Data	CS1	RFC 3662	1	AC_BK (Background)
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by the following entries:

Low-Priority Data	LE	RFCXXXX	1	AC_BK (Background)
Low-Priority Data (legacy)	CS1	RFC 3662	1	AC_BK (Background)

12. The Update to draft-ietf-tsvwg-rtcweb-qos

Section 5 of [I-D.ietf-tsvwg-rtcweb-qos] describes the Recommended DSCP Values for WebRTC Applications

This update to [I-D.ietf-tsvwg-rtcweb-qos] replaces all occurrences of CS1 with LE in Table 1:

Flow Type	Very Low	Low	Medium	High
Audio	LE (1)	DF (0)	EF (46)	EF (46)
Interactive Video with or without Audio	LE (1)	DF (0)	AF42, AF43 (36, 38)	AF41, AF42 (34, 36)
Non-Interactive Video with or without Audio	LE (1)	DF (0)	AF32, AF33 (28, 30)	AF31, AF32 (26, 28)
Data	LE (1)	DF (0)	AF11	AF21

and updates the following paragraph:

"The above table assumes that packets marked with CS1 are treated as "less than best effort", such as the LE behavior described in [RFC3662]. However, the treatment of CS1 is implementation dependent. If an implementation treats CS1 as other than "less than best effort", then the actual priority (or, more precisely, the per-hop-behavior) of the packets may be changed from what is intended. It is common for CS1 to be treated the same as DF, so applications and browsers using CS1 cannot assume that CS1 will be treated differently than DF [RFC7657]. However, it is also possible per [RFC2474] for CS1 traffic to be given better treatment than DF, thus caution should be exercised when electing to use CS1. This is one of the cases where marking packets using these recommendations can make things worse."

as follows:

"The above table assumes that packets marked with LE are treated as lower effort (i.e., "less than best effort"), such as the LE behavior described in [RFCXXXX]. However, the treatment of LE is implementation dependent. If an implementation treats LE as other than "less than best effort", then the actual priority (or, more precisely, the per-hop-behavior) of the packets may be changed from what is intended. It is common for LE to be treated the same as DF, so applications and browsers using LE cannot assume that LE will be treated differently than DF [RFC7657]. During development of this document, the CS1 DSCP was recommended for "very low" application

priority traffic; implementations that followed that recommendation SHOULD be updated to use the LE DSCP instead of the CS1 DSCP."

13. IANA Considerations

This document assigns the Differentiated Services Field Codepoint (DSCP) '000001' from the Differentiated Services Field Codepoints (DSCP) registry (<https://www.iana.org/assignments/dscp-registry/dscp-registry.xhtml>) (Pool 3, Codepoint Space xxxx01, Standards Action) to the LE PHB. This document suggests to use a DSCP from Pool 3 in order to avoid problems for other PHB marked flows to become accidentally remarked as LE PHB, e.g., due to partial DSCP bleaching. See [RFC8436] for re-classifying Pool 3 for Standards Action.

IANA is requested to update the registry as follows:

- o Name: LE
- o Value (Binary): 000001
- o Value (Decimal): 1
- o Reference: [RFC number of this memo]

14. Security Considerations

There are no specific security exposures for this PHB. Since it defines a new class of low forwarding priority, remarking other traffic as LE traffic may lead to quality-of-service degradation of such traffic. Thus, any attacker that is able to modify the DSCP of a packet to LE may carry out a downgrade attack. See the general security considerations in [RFC2474] and [RFC2475].

With respect to privacy, an attacker could use the information from the DSCP to infer that the transferred (probably even encrypted) content is considered of low priority or low urgency by a user, in case the DSCP was set on the user's request. On the one hand, this disclosed information is useful only if correlation with metadata (such as the user's IP address) and/or other flows reveal user identity. On the other hand, it might help an observer (e.g., a state level actor) who is interested in learning about the user's behavior from observed traffic: LE marked background traffic (such as software downloads, operating system updates, or telemetry data) may be less interesting for surveillance than general web traffic. Therefore, the LE marking may help the observer to focus on potentially more interesting traffic (however, the user may exploit this particular assumption and deliberately hide interesting traffic in the LE aggregate). Apart from such considerations, the impact of

disclosed information by the LE DSCP is likely negligible in most cases given the numerous traffic analysis possibilities and general privacy threats (e.g., see [RFC6973]).

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Appendix A. History of the LE PHB

A first version of this PHB was suggested by Roland Bless and Klaus Wehrle in September 1999 [draft-bless-diffserv-lbe-phb-00], named "A Lower Than Best-Effort Per-Hop Behavior". After some discussion in

the Diffserv Working Group Brian Carpenter and Kathie Nichols proposed a "bulk handling" per-domain behavior and believed a PHB was not necessary. Eventually, "Lower Effort" was specified as per-domain behavior and finally became [RFC3662]. More detailed information about its history can be found in Section 10 of [RFC3662].

There are several other names in use for this type of PHB or associated service classes. Well-known is the QBone Scavenger Service (QBSS) that was proposed in March 2001 within the Internet2 QoS Working Group. Alternative names are "Lower-than-best-effort" [carlberg-lbe-2001] or "Less-than-best-effort" [chown-lbe-2003].

Appendix B. Acknowledgments

Since text is partially borrowed from earlier Internet-Drafts and RFCs the co-authors of previous specifications are acknowledged here: Kathie Nichols and Klaus Wehrle. David Black, Olivier Bonaventure, Spencer Dawkins, Toerless Eckert, Gorrry Fairhurst, Ruediger Geib, and Kyle Rose provided helpful comments and (partially also text) suggestions.

Appendix C. Change History

This section briefly lists changes between Internet-Draft versions for convenience.

Changes in Version 10: (incorporated comments from IESG discussion as follows)

- o Appended "for Differentiated Services" to the title as suggested by Alexey.
- o Addressed Deborah Brungard's discuss: changed phrase to "However, non-LE traffic (e.g., BE traffic) SHOULD NOT be remarked to LE." with additional explanation as suggested by Gorrry.
- o Fixed the sentence "An LE PHB SHOULD NOT be used for a customer's "normal Internet" traffic nor should packets be "downgraded" to the LE PHB instead of being dropped, particularly when the packets are unauthorized traffic." according to Alice's and Mirja's comments.
- o Made reference to RFC8174 normative.
- o Added hint for the RFC editor to apply changes from section Section 12 and to delete it afterwards.

- o Incorporated Mirja's and Benjamin's suggestions.
- o Editorial suggested by Gorrry: In case => In the case that

Changes in Version 09:

- o Incorporated comments from IETF Last Call:
 - * from Olivier Bonaventure: added a bit of text for session resumption and congestion control aspects as well as ECN usage.
 - * from Kyle Rose: Revised privacy considerations text in Security Considerations Section

Changes in Version 08:

- o revised two sentences as suggested by Spencer Dawkins

Changes in Version 07:

- o revised some text for clarification according to comments from Spencer Dawkins

Changes in Version 06:

- o added Multicast Considerations section with input from Toerless Eckert
- o incorporated suggestions by David Black with respect to better reflect legacy CS1 handling

Changes in Version 05:

- o added scavenger service class into abstract
- o added some more history
- o added reference for "Myth of Over-Provisioning" in RFC3439 and references to presentations w.r.t. codepoint choices
- o added text to update draft-ietf-tsvwg-rtcweb-qos
- o revised text on congestion control in case of remarking to BE
- o added reference to DSCP measurement talk @IETF99
- o small typo fixes

Changes in Version 04:

- o Several editorial changes according to review from Gorrry Fairhurst
- o Changed the section structure a bit (moved subsections 1.1 and 1.2 into own sections 3 and 7 respectively)
- o updated section 2 on requirements language
- o added updates to RFC 8325
- o tried to be more explicit what changes are required to RFCs 4594 and 8325

Changes in Version 03:

- o Changed recommended codepoint to 000001
- o Added text to explain the reasons for the DSCP choice
- o Removed LE-min,LE-strict discussion
- o Added one more potential use case: reporting errors or telemetry data from OSs
- o Added privacy considerations to the security section (not worth an own section I think)
- o Changed IANA considerations section

Changes in Version 02:

- o Applied many editorial suggestions from David Black
- o Added Multicast traffic use case
- o Clarified what is required for deployment in section 1.2 (Deployment Considerations)
- o Added text about implementations using AQMs and ECN usage
- o Updated IANA section according to David Black's suggestions
- o Revised text in the security section
- o Changed copyright Notice to pre5378Trust200902

Changes in Version 01:

- o Now obsoletes RFC 3662.
- o Tried to be more precise in section 1.1 (Applicability) according to R. Geib's suggestions, so rephrased several paragraphs. Added text about congestion control
- o Change section 2 (PHB Description) according to R. Geib's suggestions.
- o Added RFC 2119 language to several sentences.
- o Detailed the description of remarking implications and recommendations in Section 8.
- o Added Section 10 to explicitly list changes with respect to RFC 4594, because this document will update it.

Appendix D. Note to RFC Editor

This section lists actions for the RFC editor during final formatting.

- o Apply the suggested changes of section Section 12 and add a normative reference in draft-ietf-tsvwg-rtcweb-qos to this RFC.
- o Delete Section 12.
- o Please replace the occurrences of RFCXXXX in Section 10 and Section 11 with the assigned RFC number for this document.
- o Delete Appendix C.
- o Delete this section.

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25 October 2021

Stream Control Transmission Protocol (SCTP) Network Address Translation
Support
draft-ietf-tsvwg-natsupp-23

Abstract

The Stream Control Transmission Protocol (SCTP) provides a reliable communications channel between two end-hosts in many ways similar to the Transmission Control Protocol (TCP). With the widespread deployment of Network Address Translators (NAT), specialized code has been added to NAT functions for TCP that allows multiple hosts to reside behind a NAT function and yet share a single IPv4 address, even when two hosts (behind a NAT function) choose the same port numbers for their connection. This additional code is sometimes classified as Network Address and Port Translation (NAPT).

This document describes the protocol extensions needed for the SCTP endpoints and the mechanisms for NAT functions necessary to provide similar features of NAPT in the single point and multipoint traversal scenario.

Finally, a YANG module for SCTP NAT is defined.

Status of This Memo

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1. Introduction

Stream Control Transmission Protocol (SCTP) [RFC4960] provides a reliable communications channel between two end-hosts in many ways similar to TCP [RFC0793]. With the widespread deployment of Network Address Translators (NAT), specialized code has been added to NAT functions for TCP that allows multiple hosts to reside behind a NAT function using private-use addresses (see [RFC6890]) and yet share a single IPv4 address, even when two hosts (behind a NAT function) choose the same port numbers for their connection. This additional code is sometimes classified as Network Address and Port Translation (NAPT). Please note that this document focuses on the case where the NAT function maps a single or multiple internal addresses to a single external address and vice versa.

To date, specialized code for SCTP has not yet been added to most NAT functions so that only a translation of IP addresses is supported. The end result of this is that only one SCTP-capable host can successfully operate behind such a NAT function and this host can only be single-homed. The only alternative for supporting legacy NAT functions is to use UDP encapsulation as specified in [RFC6951].

The NAT function in the document refers to NAPT functions described in Section 2.2 of [RFC3022], NAT64 [RFC6146], or DS-Lite AFTR [RFC6333].

This document specifies procedures allowing a NAT function to support SCTP by providing similar features to those provided by a NAPT for TCP (see [RFC5382] and [RFC7857]), UDP (see [RFC4787] and [RFC7857]), and ICMP (see [RFC5508] and [RFC7857]). This document also specifies a set of data formats for SCTP packets and a set of SCTP endpoint procedures to support NAT traversal. An SCTP implementation supporting these procedures can assure that in both single-homed and multi-homed cases a NAT function will maintain the appropriate state without the NAT function needing to change port numbers.

It is possible and desirable to make these changes for a number of reasons:

- * It is desirable for SCTP internal end-hosts on multiple platforms to be able to share a NAT function's external IP address in the same way that a TCP session can use a NAT function.
- * If a NAT function does not need to change any data within an SCTP packet, it will reduce the processing burden of NAT'ing SCTP by not needing to execute the CRC32c checksum used by SCTP.
- * Not having to touch the IP payload makes the processing of ICMP messages by NAT functions easier.

An SCTP-aware NAT function will need to follow these procedures for generating appropriate SCTP packet formats.

When considering SCTP-aware NAT it is possible to have multiple levels of support. At each level, the Internal Host, Remote Host, and NAT function does or does not support the procedures described in this document. The following table illustrates the results of the various combinations of support and if communications can occur between two endpoints.

Internal Host	NAT Function	Remote Host	Communication
Support	Support	Support	Yes
Support	Support	No Support	Limited
Support	No Support	Support	None
Support	No Support	No Support	None
No Support	Support	Support	Limited
No Support	Support	No Support	Limited
No Support	No Support	Support	None
No Support	No Support	No Support	None

Table 1: Communication possibilities

From the table it can be seen that no communication can occur when a NAT function does not support SCTP-aware NAT. This assumes that the NAT function does not handle SCTP packets at all and all SCTP packets sent from behind a NAT function are discarded by the NAT function. In some cases, where the NAT function supports SCTP-aware NAT, but one of the two hosts does not support the feature, communication can possibly occur in a limited way. For example, only one host can have a connection when a collision case occurs.

2. Conventions

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

3. Terminology

This document uses the following terms, which are depicted in Figure 1. Familiarity with the terminology used in [RFC4960] and [RFC5061] is assumed.

Internal-Address (Int-Addr)

An internal address that is known to the internal host.

Internal-Port (Int-Port)

The port number that is in use by the host holding the Internal-Address.

Internal-VTag (Int-VTag)

The SCTP Verification Tag (VTag) (see Section 3.1 of [RFC4960]) that the internal host has chosen for an association. The VTag is a unique 32-bit tag that accompanies any incoming SCTP packet for this association to the Internal-Address.

Remote-Address (Rem-Addr)

The address that an internal host is attempting to contact.

Remote-Port (Rem-Port)

The port number used by the host holding the Remote-Address.

Remote-VTag (Rem-VTag)

The Verification Tag (VTag) (see Section 3.1 of [RFC4960]) that the host holding the Remote-Address has chosen for an association. The VTag is a unique 32-bit tag that accompanies any outgoing SCTP packet for this association to the Remote-Address.

External-Address (Ext-Addr)

An external address assigned to the NAT function, that it uses as a source address when sending packets towards a Remote-Address.

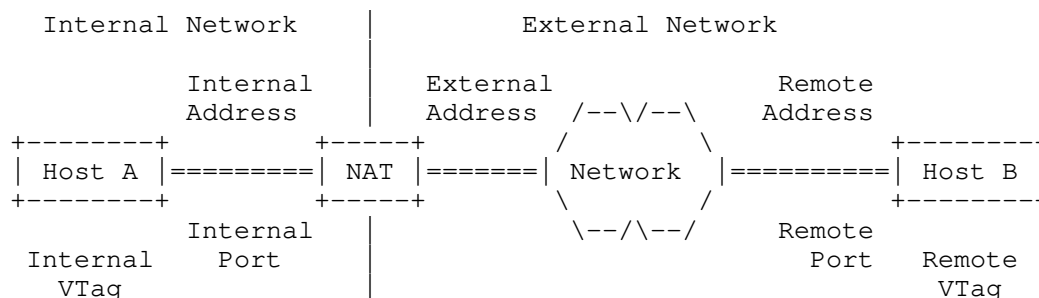


Figure 1: Basic Network Setup

4. Motivation and Overview

4.1. SCTP NAT Traversal Scenarios

This section defines the notion of single and multipoint NAT traversal.

4.1.1. Single Point Traversal

In this case, all packets in the SCTP association go through a single NAT function, as shown in Figure 2.

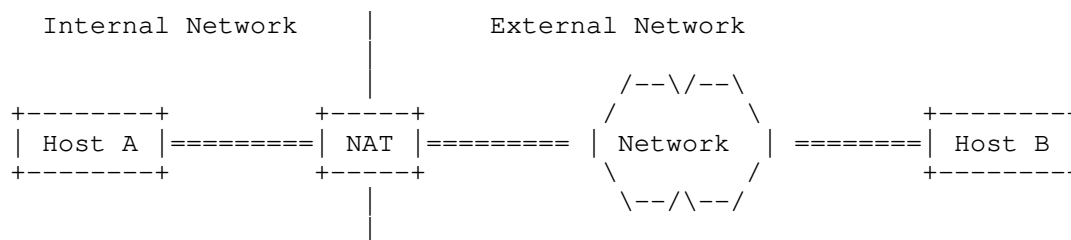


Figure 2: Single NAT Function Scenario

A variation of this case is shown in Figure 3, i.e., multiple NAT functions in the forwarding path between two endpoints.

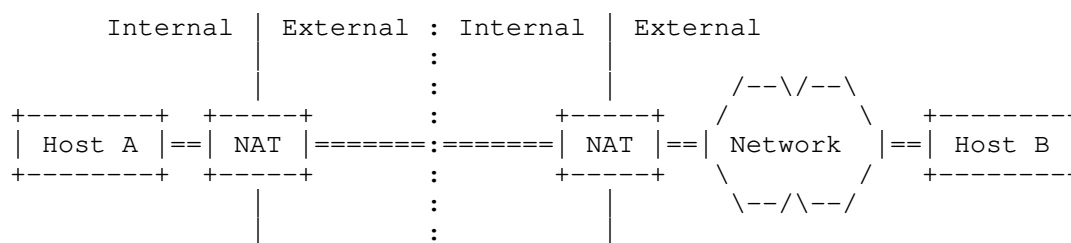


Figure 3: Serial NAT Functions Scenario

Although one of the main benefits of SCTP multi-homing is redundant paths, in the single point traversal scenario the NAT function represents a single point of failure in the path of the SCTP multi-homed association. However, the rest of the path can still benefit from path diversity provided by SCTP multi-homing.

The two SCTP endpoints in this case can be either single-homed or multi-homed. However, the important thing is that the NAT function in this case sees all the packets of the SCTP association.

4.1.2. Multipoint Traversal

This case involves multiple NAT functions and each NAT function only sees some of the packets in the SCTP association. An example is shown in Figure 4.

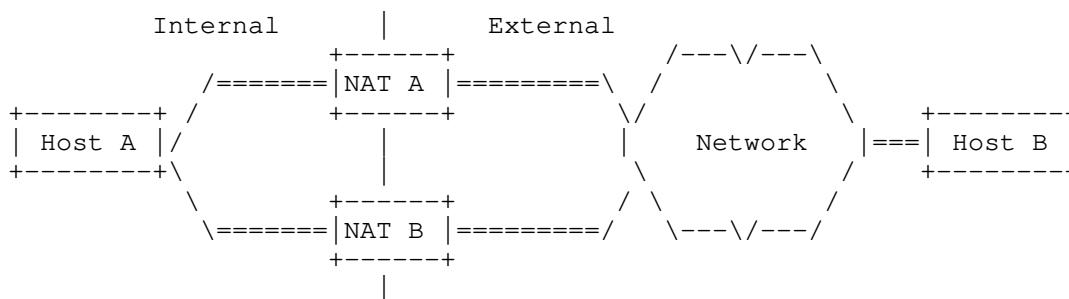


Figure 4: Parallel NAT Functions Scenario

This case does not apply to a single-homed SCTP association (i.e., both endpoints in the association use only one IP address). The advantage here is that the existence of multiple NAT traversal points can preserve the path diversity of a multi-homed association for the entire path. This in turn can improve the robustness of the communication.

4.2. Limitations of Classical NAPT for SCTP

Using classical NAPT possibly results in changing one of the SCTP port numbers during the processing, which requires the recomputation of the transport layer checksum by the NAPT function. Whereas for UDP and TCP this can be done very efficiently, for SCTP the checksum (CRC32c) over the entire packet needs to be recomputed (see Appendix B of [RFC4960] for details of the CRC32c computation). This would considerably add to the NAT computational burden, however hardware support can mitigate this in some implementations.

An SCTP endpoint can have multiple addresses but only has a single port number to use. To make multipoint traversal work, all the NAT functions involved need to recognize the packets they see as belonging to the same SCTP association and perform port number translation in a consistent way. One possible way of doing this is to use a pre-defined table of port numbers and addresses configured within each NAT function. Other mechanisms could make use of NAT to NAT communication. Such mechanisms have not been deployed on a wide scale base and thus are not a preferred solution. Therefore an SCTP variant of NAT function has been developed (see Section 4.3).

4.3. The SCTP-Specific Variant of NAT

In this section it is allowed that there are multiple SCTP capable hosts behind a NAT function that share one External-Address. Furthermore, this section focuses on the single point traversal scenario (see Section 4.1.1).

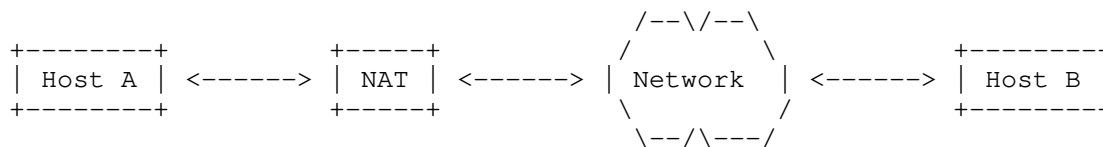
The modification of outgoing SCTP packets sent from an internal host is simple: the source address of the packets has to be replaced with the External-Address. It might also be necessary to establish some state in the NAT function to later handle incoming packets.

Typically, the NAT function has to maintain a NAT binding table of Internal-VTag, Internal-Port, Remote-VTag, Remote-Port, Internal-Address, and whether the restart procedure is disabled or not. An entry in that NAT binding table is called a NAT-State control block. The function Create() obtains the just mentioned parameters and returns a NAT-State control block. A NAT function MAY allow creating NAT-State control blocks via a management interface.

For SCTP packets coming from the external realm of the NAT function the destination address of the packets has to be replaced with the Internal-Address of the host to which the packet has to be delivered, if a NAT state entry is found. The lookup of the Internal-Address is based on the Remote-VTag, Remote-Port, Internal-VTag and the Internal-Port.

The entries in the NAT binding table need to fulfill some uniqueness conditions. There can not be more than one entry NAT binding table with the same pair of Internal-Port and Remote-Port. This rule can be relaxed, if all NAT binding table entries with the same Internal-Port and Remote-Port have the support for the restart procedure disabled (see Section 5.3.1). In this case there can not be no more than one entry with the same Internal-Port, Remote-Port and Remote-VTag and no more than one NAT binding table entry with the same Internal-Port, Remote-Port, and Int-VTag.

The processing of outgoing SCTP packets containing an INIT chunk is illustrated in the following figure. This scenario is valid for all message flows in this section.



```

INIT[Initiate-Tag]
Int-Addr:Int-Port -----> Rem-Addr:Rem-Port
Rem-VTag=0

Create(Initiate-Tag, Int-Port, 0, Rem-Port, Int-Addr,
      IsRestartDisabled)
Returns(NAT-State control block)

```

Translate To:

```

INIT[Initiate-Tag]
Ext-Addr:Int-Port -----> Rem-Addr:Rem-Port
Rem-VTag=0

```

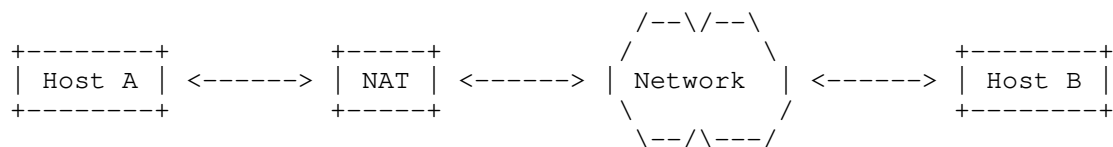
Normally a NAT binding table entry will be created.

However, it is possible that there is already a NAT binding table entry with the same Remote-Port, Internal-Port, and Internal-VTag but different Internal-Address and the restart procedure is disabled. In this case the packet containing the INIT chunk MUST be dropped by the NAT and a packet containing an ABORT chunk SHOULD be sent to the SCTP host that originated the packet with the M bit set and 'VTag and Port Number Collision' error cause (see Section 5.1.1 for the format). The source address of the packet containing the ABORT chunk MUST be the destination address of the packet containing the INIT chunk.

If an outgoing SCTP packet contains an INIT or ASCONF chunk and a matching NAT binding table entry is found, the packet is processed as a normal outgoing packet.

It is also possible that a NAT binding table entry with the same Remote-Port and Internal-Port exists without an Internal-VTag conflict but there exists a NAT binding table entry with the same port numbers but a different Internal-Address and the restart procedure is not disabled. In such a case the packet containing the INIT chunk MUST be dropped by the NAT function and a packet containing an ABORT chunk SHOULD be sent to the SCTP host that originated the packet with the M bit set and 'Port Number Collision' error cause (see Section 5.1.1 for the format).

The processing of outgoing SCTP packets containing no INIT chunks is described in the following figure.

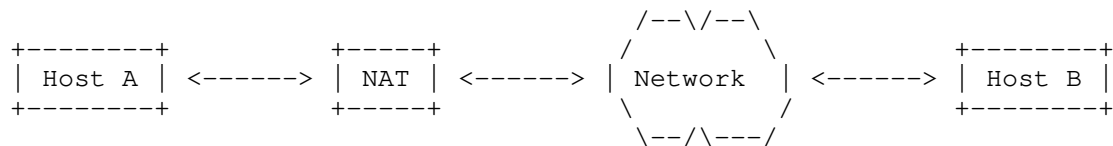


Int-Addr:Int-Port -----> Rem-Addr:Rem-Port
 Rem-VTag

Translate To:

Ext-Addr:Int-Port -----> Rem-Addr:Rem-Port
 Rem-VTag

The processing of incoming SCTP packets containing an INIT ACK chunk is illustrated in the following figure. The Lookup() function has as input the Internal-VTag, Internal-Port, Remote-VTag, and Remote-Port. It returns the corresponding entry of the NAT binding table and updates the Remote-VTag by substituting it with the value of the Initiate-Tag of the INIT ACK chunk. The wildcard character signifies that the parameter's value is not considered in the Lookup() function or changed in the Update() function, respectively.



 INIT ACK[Initiate-Tag]
 Ext-Addr:Int-Port <---- Rem-Addr:Rem-Port
 Int-VTag

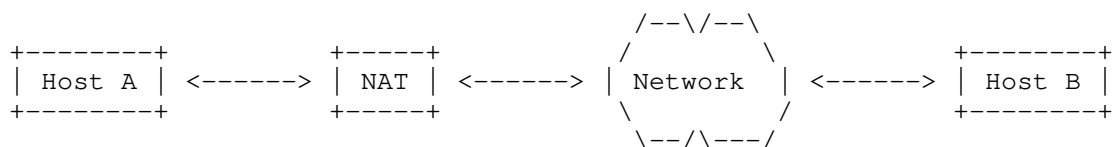
Lookup(Int-VTag, Int-Port, *, Rem-Port)
 Update(*, *, Initiate-Tag, *)

Returns(NAT-State control block containing Int-Addr)

 INIT ACK[Initiate-Tag]
 Int-Addr:Int-Port <----- Rem-Addr:Rem-Port
 Int-VTag

In the case where the Lookup function fails because it does not find an entry, the SCTP packet is dropped. If it succeeds, the Update routine inserts the Remote-VTag (the Initiate-Tag of the INIT ACK chunk) in the NAT-State control block.

The processing of incoming SCTP packets containing an ABORT or SHUTDOWN COMPLETE chunk with the T bit set is illustrated in the following figure.



Ext-Addr:Int-Port <----- Rem-Addr:Rem-Port
Rem-VTag

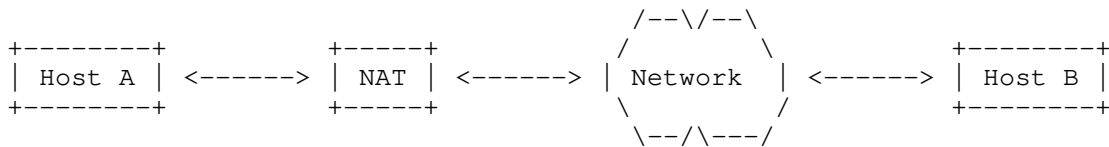
Lookup(*, Int-Port, Rem-VTag, Rem-Port)

Returns (NAT-State control block containing Int-Addr)

Int-Addr:Int-Port <----- Rem-Addr:Rem-Port
Rem-VTag

For an incoming packet containing an INIT chunk a table lookup is made only based on the addresses and port numbers. If an entry with a Remote-VTag of zero is found, it is considered a match and the Remote-VTag is updated. If an entry with a non-matching Remote-VTag is found or no entry is found, the incoming packet is silently dropped. If an entry with a matching Remote-VTag is found, the incoming packet is forwarded. This allows the handling of INIT collision through NAT functions.

The processing of other incoming SCTP packets is described in the following figure.



Ext-Addr:Int-Port <----- Rem-Addr:Rem-Port
Int-VTag

Lookup(Int-VTag, Int-Port, *, Rem-Port)

Returns(NAT-State control block containing Internal-Address)

Int-Addr:Int-Port <----- Rem-Addr:Rem-Port
Int-VTag

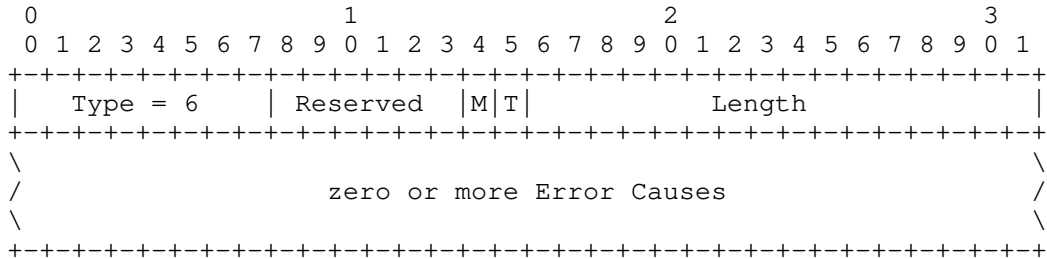
5. Data Formats

This section defines the formats used to support NAT traversal. Section 5.1 and Section 5.2 describe chunks and error causes sent by NAT functions and received by SCTP endpoints. Section 5.3 describes parameters sent by SCTP endpoints and used by NAT functions and SCTP endpoints.

5.1. Modified Chunks

This section presents existing chunks defined in [RFC4960] for which additional flags are specified by this document.

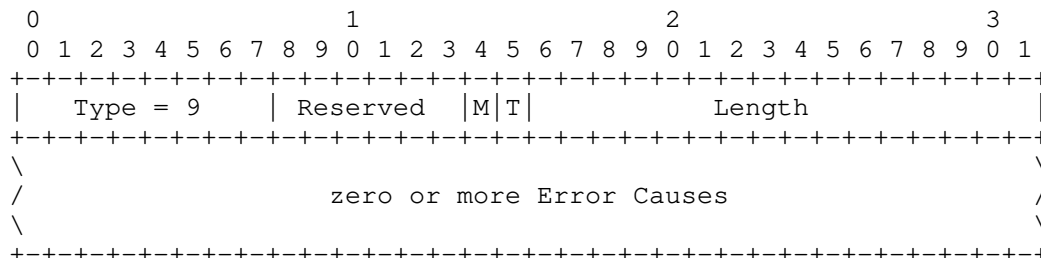
5.1.1. Extended ABORT Chunk



The ABORT chunk is extended to add the new 'M bit'. The M bit indicates to the receiver of the ABORT chunk that the chunk was not generated by the peer SCTP endpoint, but instead by a middle box (e.g., NAT).

[NOTE to RFC-Editor: Assignment of M bit to be confirmed by IANA.]

5.1.2. Extended ERROR Chunk



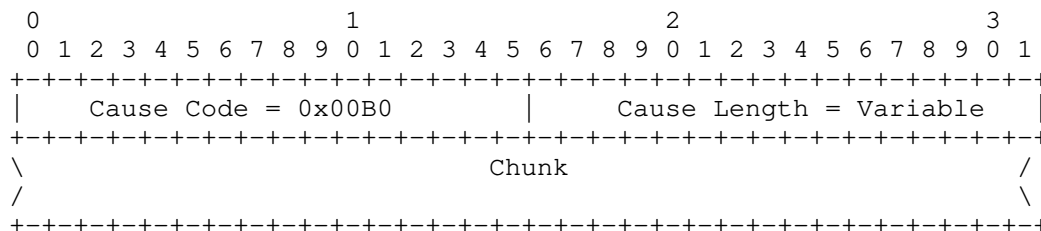
The ERROR chunk defined in [RFC4960] is extended to add the new 'M bit'. The M bit indicates to the receiver of the ERROR chunk that the chunk was not generated by the peer SCTP endpoint, but instead by a middle box.

[NOTE to RFC-Editor: Assignment of M bit to be confirmed by IANA.]

5.2. New Error Causes

This section defines the new error causes added by this document.

5.2.1. VTag and Port Number Collision Error Cause



Cause Code: 2 bytes (unsigned integer)

This field holds the IANA defined cause code for the 'VTag and Port Number Collision' Error Cause. IANA is requested to assign the value 0x00B0 for this cause code.

Cause Length: 2 bytes (unsigned integer)

This field holds the length in bytes of the error cause. The value MUST be the length of the Cause-Specific Information plus 4.

Chunk: variable length

The Cause-Specific Information is filled with the chunk that caused this error. This can be an INIT, INIT ACK, or ASCONF chunk. Note that if the entire chunk will not fit in the ERROR chunk or ABORT chunk being sent then the bytes that do not fit are truncated.

[NOTE to RFC-Editor: Assignment of cause code to be confirmed by IANA.]

5.2.2. Missing State Error Cause

0										1										2										3									
0	1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6	7	8	9	0	1								
Cause Code = 0x00B1										Cause Length = Variable																													
Original Packet																																							

Cause Code: 2 bytes (unsigned integer)

This field holds the IANA defined cause code for the 'Missing State' Error Cause. IANA is requested to assign the value 0x00B1 for this cause code.

Cause Length: 2 bytes (unsigned integer)

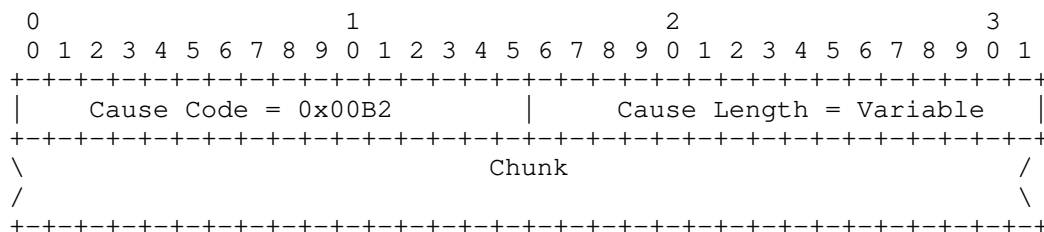
This field holds the length in bytes of the error cause. The value MUST be the length of the Cause-Specific Information plus 4.

Original Packet: variable length

The Cause-Specific Information is filled with the IPv4 or IPv6 packet that caused this error. The IPv4 or IPv6 header MUST be included. Note that if the packet will not fit in the ERROR chunk or ABORT chunk being sent then the bytes that do not fit are truncated.

[NOTE to RFC-Editor: Assignment of cause code to be confirmed by IANA.]

5.2.3. Port Number Collision Error Cause



Cause Code: 2 bytes (unsigned integer)

This field holds the IANA defined cause code for the 'Port Number Collision' Error Cause. IANA is requested to assign the value 0x00B2 for this cause code.

Cause Length: 2 bytes (unsigned integer)

This field holds the length in bytes of the error cause. The value MUST be the length of the Cause-Specific Information plus 4.

Chunk: variable length

The Cause-Specific Information is filled with the chunk that caused this error. This can be an INIT, INIT ACK, or ASCONF chunk. Note that if the entire chunk will not fit in the ERROR chunk or ABORT chunk being sent then the bytes that do not fit are truncated.

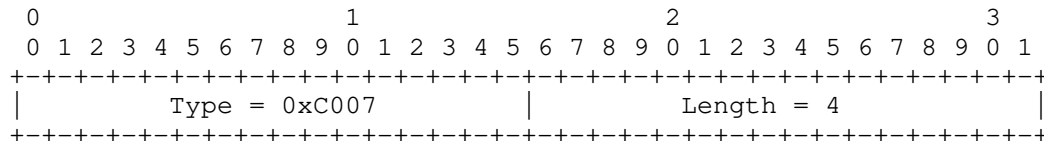
[NOTE to RFC-Editor: Assignment of cause code to be confirmed by IANA.]

5.3. New Parameters

This section defines new parameters and their valid appearance defined by this document.

5.3.1. Disable Restart Parameter

This parameter is used to indicate that the restart procedure is requested to be disabled. Both endpoints of an association MUST include this parameter in the INIT chunk and INIT ACK chunk when establishing an association and MUST include it in the ASCONF chunk when adding an address to successfully disable the restart procedure.



Parameter Type: 2 bytes (unsigned integer)

This field holds the IANA defined parameter type for the Disable Restart Parameter. IANA is requested to assign the value 0xC007 for this parameter type.

Parameter Length: 2 bytes (unsigned integer)

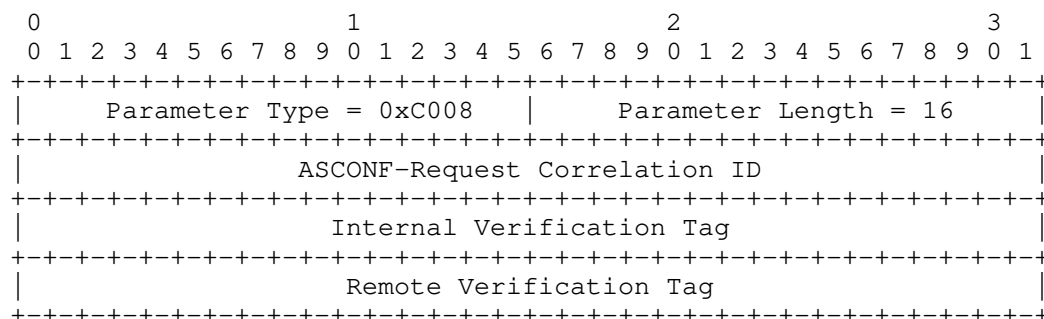
This field holds the length in bytes of the parameter. The value MUST be 4.

[NOTE to RFC-Editor: Assignment of parameter type to be confirmed by IANA.]

The Disable Restart Parameter MAY appear in INIT, INIT ACK and ASCONF chunks and MUST NOT appear in any other chunk.

5.3.2. VTags Parameter

This parameter is used to help a NAT function to recover from state loss.



Parameter Type: 2 bytes (unsigned integer)

This field holds the IANA defined parameter type for the VTags Parameter. IANA is requested to assign the value 0xC008 for this parameter type.

Parameter Length: 2 bytes (unsigned integer)

This field holds the length in bytes of the parameter. The value MUST be 16.

ASCONF-Request Correlation ID: 4 bytes (unsigned integer)

This is an opaque integer assigned by the sender to identify each request parameter. The receiver of the ASCONF Chunk will copy this 32-bit value into the ASCONF Response Correlation ID field of the ASCONF ACK response parameter. The sender of the packet containing the ASCONF chunk can use this same value in the ASCONF ACK chunk to find which request the response is for. The receiver MUST NOT change the value of the ASCONF-Request Correlation ID.

Internal Verification Tag: 4 bytes (unsigned integer)

The Verification Tag that the internal host has chosen for the association. The Verification Tag is a unique 32-bit tag that accompanies any incoming SCTP packet for this association to the Internal-Address.

Remote Verification Tag: 4 bytes (unsigned integer)

The Verification Tag that the host holding the Remote-Address has chosen for the association. The VTag is a unique 32-bit tag that accompanies any outgoing SCTP packet for this association to the Remote-Address.

[NOTE to RFC-Editor: Assignment of parameter type to be confirmed by IANA.]

The VTags Parameter MAY appear in ASCONF chunks and MUST NOT appear in any other chunk.

6. Procedures for SCTP Endpoints and NAT Functions

If an SCTP endpoint is behind an SCTP-aware NAT, a number of problems can arise as it tries to communicate with its peers:

- * IP addresses can not be included in the SCTP packet. This is discussed in Section 6.1.
- * More than one host behind a NAT function could select the same VTag and source port number when communicating with the same peer server. This creates a situation where the NAT function will not be able to tell the two associations apart. This situation is discussed in Section 6.2.
- * If an SCTP endpoint is a server communicating with multiple peers and the peers are behind the same NAT function, then these peers cannot be distinguished by the server. This case is discussed in Section 6.3.
- * A restart of a NAT function during a conversation could cause a loss of its state. This problem and its solution is discussed in Section 6.4.
- * NAT functions need to deal with SCTP packets being fragmented at the IP layer. This is discussed in Section 6.5.
- * An SCTP endpoint can be behind two NAT functions in parallel providing redundancy. The method to set up this scenario is discussed in Section 6.6.

The mechanisms to solve these problems require additional chunks and parameters, defined in this document, and modified handling procedures from those specified in [RFC4960] as described below.

6.1. Association Setup Considerations for Endpoints

The association setup procedure defined in [RFC4960] allows multi-homed SCTP endpoints to exchange its IP-addresses by using IPv4 or IPv6 address parameters in the INIT and INIT ACK chunks. However, this does not work when NAT functions are present.

Every association setup from a host behind a NAT function MUST NOT use multiple internal addresses. The INIT chunk MUST NOT contain an IPv4 Address parameter, IPv6 Address parameter, or Supported Address Types parameter. The INIT ACK chunk MUST NOT contain any IPv4 Address parameter or IPv6 Address parameter using non-global addresses. The INIT chunk and the INIT ACK chunk MUST NOT contain any Host Name parameters.

If the association is intended to be finally multi-homed, the procedure in Section 6.6 MUST be used.

The INIT and INIT ACK chunk SHOULD contain the Disable Restart parameter defined in Section 5.3.1.

6.2. Handling of Internal Port Number and Verification Tag Collisions

Consider the case where two hosts in the Internal-Address space want to set up an SCTP association with the same service provided by some remote hosts. This means that the Remote-Port is the same. If they both choose the same Internal-Port and Internal-VTag, the NAT function cannot distinguish between incoming packets anymore. However, this is unlikely. The Internal-VTags are chosen at random and if the Internal-Ports are also chosen from the ephemeral port range at random (see [RFC6056]) this gives a 46-bit random number that has to match.

The same can happen with the Remote-VTag when a packet containing an INIT ACK chunk or an ASCONF chunk is processed by the NAT function.

6.2.1. NAT Function Considerations

If the NAT function detects a collision of internal port numbers and verification tags, it SHOULD send a packet containing an ABORT chunk with the M bit set if the collision is triggered by a packet containing an INIT or INIT ACK chunk. If such a collision is triggered by a packet containing an ASCONF chunk, it SHOULD send a packet containing an ERROR chunk with the M bit. The M bit is a new

bit defined by this document to express to SCTP that the source of this packet is a "middle" box, not the peer SCTP endpoint (see Section 5.1.1). If a packet containing an INIT ACK chunk triggers the collision, the corresponding packet containing the ABORT chunk MUST contain the same source and destination address and port numbers as the packet containing the INIT ACK chunk. If a packet containing an INIT chunk or an ASCONF chunk, the source and destination address and port numbers MUST be swapped.

The sender of the packet containing an ERROR or ABORT chunk MUST include the error cause with cause code 'VTag and Port Number Collision' (see Section 5.2.1).

6.2.2. Endpoint Considerations

The sender of the packet containing the INIT chunk or the receiver of a packet containing the INIT ACK chunk, upon reception of a packet containing an ABORT chunk with M bit set and the appropriate error cause code for colliding NAT binding table state is included, SHOULD reinitiate the association setup procedure after choosing a new initiate tag, if the association is in COOKIE-WAIT state. In any other state, the SCTP endpoint MUST NOT respond.

The sender of the packet containing the ASCONF chunk, upon reception of a packet containing an ERROR chunk with M bit set, MUST stop adding the path to the association.

6.3. Handling of Internal Port Number Collisions

When two SCTP hosts are behind an SCTP-aware NAT it is possible that two SCTP hosts in the Internal-Address space will want to set up an SCTP association with the same server running on the same remote host. If the two hosts choose the same internal port, this is considered an internal port number collision.

For the NAT function, appropriate tracking can be performed by assuring that the VTags are unique between the two hosts.

6.3.1. NAT Function Considerations

The NAT function, when processing the packet containing the INIT ACK chunk, SHOULD note in its NAT binding table if the association supports the disable restart extension. This note is used when establishing future associations (i.e. when processing a packet containing an INIT chunk from an internal host) to decide if the connection can be allowed. The NAT function does the following when processing a packet containing an INIT chunk:

- * If the packet containing the INIT chunk is originating from an internal port to a remote port for which the NAT function has no matching NAT binding table entry, it MUST allow the packet containing the INIT chunk creating an NAT binding table entry.
- * If the packet containing the INIT chunk matches an existing NAT binding table entry, it MUST validate that the disable restart feature is supported and, if it does, allow the packet containing the INIT chunk to be forwarded.
- * If the disable restart feature is not supported, the NAT function SHOULD send a packet containing an ABORT chunk with the M bit set.

The 'Port Number Collision' error cause (see Section 5.2.3) MUST be included in the ABORT chunk sent in response to the packet containing an INIT chunk.

If the collision is triggered by a packet containing an ASCONF chunk, a packet containing an ERROR chunk with the 'Port Number Collision' error cause SHOULD be sent in response to the packet containing the ASCONF chunk.

6.3.2. Endpoint Considerations

For the remote SCTP server this means that the Remote-Port and the Remote-Address are the same. If they both have chosen the same Internal-Port the server cannot distinguish between both associations based on the address and port numbers. For the server it looks like the association is being restarted. To overcome this limitation the client sends a Disable Restart parameter in the INIT chunk.

When the server receives this parameter it does the following:

- * It MUST include a Disable Restart parameter in the INIT ACK to inform the client that it will support the feature.
- * It MUST disable the restart procedures defined in [RFC4960] for this association.

Servers that support this feature will need to be capable of maintaining multiple connections to what appears to be the same peer (behind the NAT function) differentiated only by the VTags.

6.4. Handling of Missing State

6.4.1. NAT Function Considerations

If the NAT function receives a packet from the internal network for which the lookup procedure does not find an entry in the NAT binding table, a packet containing an ERROR chunk SHOULD be sent back with the M bit set. The source address of the packet containing the ERROR chunk MUST be the destination address of the packet received from the internal network. The verification tag is reflected and the T bit is set. Such a packet containing an ERROR chunk SHOULD NOT be sent if the received packet contains an ASCONF chunk with the VTags parameter or an ABORT, SHUTDOWN COMPLETE or INIT ACK chunk. A packet containing an ERROR chunk MUST NOT be sent if the received packet contains an ERROR chunk with the M bit set. In any case, the packet SHOULD NOT be forwarded to the remote address.

If the NAT function receives a packet from the internal network for which it has no NAT binding table entry and the packet contains an ASCONF chunk with the VTags parameter, the NAT function MUST update its NAT binding table according to the verification tags in the VTags parameter and, if present, the Disable Restart parameter.

When sending a packet containing an ERROR chunk, the error cause 'Missing State' (see Section 5.2.2) MUST be included and the M bit of the ERROR chunk MUST be set (see Section 5.1.2).

6.4.2. Endpoint Considerations

Upon reception of this packet containing the ERROR chunk by an SCTP endpoint the receiver takes the following actions:

- * It SHOULD validate that the verification tag is reflected by looking at the VTag that would have been included in an outgoing packet. If the validation fails, discard the received packet containing the ERROR chunk.
- * It SHOULD validate that the peer of the SCTP association supports the dynamic address extension. If the validation fails, discard the received packet containing the ERROR chunk.
- * It SHOULD generate a packet containing a new ASCONF chunk containing the VTags parameter (see Section 5.3.2) and the Disable Restart parameter (see Section 5.3.1) if the association is using the disable restart feature. By processing this packet the NAT function can recover the appropriate state. The procedures for generating an ASCONF chunk can be found in [RFC5061].

The peer SCTP endpoint receiving such a packet containing an ASCONF chunk SHOULD add the address and respond with an acknowledgment if the address is new to the association (following all procedures defined in [RFC5061]). If the address is already part of the association, the SCTP endpoint MUST NOT respond with an error, but instead SHOULD respond with a packet containing an ASCONF ACK chunk acknowledging the address and take no action (since the address is already in the association).

Note that it is possible that upon receiving a packet containing an ASCONF chunk containing the VTags parameter the NAT function will realize that it has an 'Internal Port Number and Verification Tag collision'. In such a case the NAT function SHOULD send a packet containing an ERROR chunk with the error cause code set to 'VTag and Port Number Collision' (see Section 5.2.1).

If an SCTP endpoint receives a packet containing an ERROR chunk with 'Internal Port Number and Verification Tag collision' as the error cause and the packet in the Error Chunk contains an ASCONF with the VTags parameter, careful examination of the association is necessary. The endpoint does the following:

- * It MUST validate that the verification tag is reflected by looking at the VTag that would have been included in the outgoing packet. If the validation fails, it MUST discard the packet.
- * It MUST validate that the peer of the SCTP association supports the dynamic address extension. If the peer does not support this extension, it MUST discard the received packet containing the ERROR chunk.
- * If the association is attempting to add an address (i.e. following the procedures in Section 6.6) then the endpoint MUST NOT consider the address part of the association and SHOULD make no further attempt to add the address (i.e. cancel any ASCONF timers and remove any record of the path), since the NAT function has a VTag collision and the association cannot easily create a new VTag (as it would if the error occurred when sending a packet containing an INIT chunk).
- * If the endpoint has no other path, i.e. the procedure was executed due to missing a state in the NAT function, then the endpoint MUST abort the association. This would occur only if the local NAT function restarted and accepted a new association before attempting to repair the missing state (Note that this is no different than what happens to all TCP connections when a NAT function loses its state).

6.5. Handling of Fragmented SCTP Packets by NAT Functions

SCTP minimizes the use of IP-level fragmentation. However, it can happen that using IP-level fragmentation is needed to continue an SCTP association. For example, if the path MTU is reduced and there are still some DATA chunk in flight, which require packets larger than the new path MTU. If IP-level fragmentation can not be used, the SCTP association will be terminated in a non-graceful way. See [RFC8900] for more information about IP fragmentation.

Therefore, a NAT function MUST be able to handle IP-level fragmented SCTP packets. The fragments MAY arrive in any order.

When an SCTP packet can not be forwarded by the NAT function due to MTU issues and the IP header forbids fragmentation, the NAT MUST send back a "Fragmentation needed and DF set" ICMPv4 or PTB ICMPv6 message to the internal host. This allows for a faster recovery from this packet drop.

6.6. Multi Point Traversal Considerations for Endpoints

If a multi-homed SCTP endpoint behind a NAT function connects to a peer, it MUST first set up the association single-homed with only one address causing the first NAT function to populate its state. Then it SHOULD add each IP address using packets containing ASCONF chunks sent via their respective NAT functions. The address used in the Add IP address parameter is the wildcard address (0.0.0.0 or ::0) and the address parameter in the ASCONF chunk SHOULD also contain the VTags parameter and optionally the Disable Restart parameter.

7. SCTP NAT YANG Module

This section defines a YANG module for SCTP NAT.

The terminology for describing YANG data models is defined in [RFC7950]. The meaning of the symbols in tree diagrams is defined in [RFC8340].

7.1. Tree Structure

This module augments NAT YANG module [RFC8512] with SCTP specifics. The module supports both classical SCTP NAT (that is, rewrite port numbers) and SCTP-specific variant where the ports numbers are not altered. The YANG "feature" is used to indicate whether SCTP-specific variant is supported.

The tree structure of the SCTP NAT YANG module is provided below:

```

module: ietf-nat-sctp
  augment /nat:nat/nat:instances/nat:instance
    /nat:policy/nat:timers:
      +--rw sctp-timeout?  uint32
  augment /nat:nat/nat:instances/nat:instance
    /nat:mapping-table/nat:mapping-entry:
      +--rw int-VTag?      uint32 {sctp-nat}?
      +--rw rem-VTag?      uint32 {sctp-nat}?

```

Concretely, the SCTP NAT YANG module augments the NAT YANG module (policy, in particular) with the following:

- * The sctp-timeout is used to control the SCTP inactivity timeout. That is, the time an SCTP mapping will stay active without SCTP packets traversing the NAT. This timeout can be set only for SCTP. Hence, `"/nat:nat/nat:instances/nat:instance/nat:policy/nat:transport-protocols/nat:protocol-id"` MUST be set to `'132'` (SCTP).

In addition, the SCTP NAT YANG module augments the mapping entry with the following parameters defined in Section 3. These parameters apply only for SCTP NAT mapping entries (i.e., `"/nat/instances/instance/mapping-table/mapping-entry/transport-protocol"` MUST be set to `'132'`);

- * The Internal Verification Tag (Int-VTag)
- * The Remote Verification Tag (Rem-VTag)

7.2. YANG Module

```

<CODE BEGINS> file "ietf-nat-sctp@2020-11-02.yang"
module ietf-nat-sctp {
  yang-version 1.1;
  namespace "urn:ietf:params:xml:ns:yang:ietf-nat-sctp";
  prefix nat-sctp;

  import ietf-nat {
    prefix nat;
    reference
      "RFC 8512: A YANG Module for Network Address Translation
       (NAT) and Network Prefix Translation (NPT)";
  }

  organization
    "IETF TSVWG Working Group";
  contact
    "WG Web:  <https://datatracker.ietf.org/wg/tsvwg/>

```

WG List: <mailto:tsvwg@ietf.org>

Author: Mohamed Boucadair
<mailto:mohamed.boucadair@orange.com>;

description

"This module augments NAT YANG module with Stream Control Transmission Protocol (SCTP) specifics. The extension supports both a classical SCTP NAT (that is, rewrite port numbers) and a, SCTP-specific variant where the ports numbers are not altered.

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This version of this YANG module is part of RFC XXXX; see the RFC itself for full legal notices.";

```
revision 2019-11-18 {
  description
    "Initial revision.";
  reference
    "RFC XXXX: Stream Control Transmission Protocol (SCTP)
      Network Address Translation Support";
}

feature sctp-nat {
  description
    "This feature means that SCTP-specific variant of NAT
      is supported. That is, avoid rewriting port numbers.";
  reference
    "Section 4.3 of RFC XXXX.";
}

augment "/nat:nat/nat:instances/nat:instance"
  + "/nat:policy/nat:timers" {
  when "/nat:nat/nat:instances/nat:instance"
    + "/nat:policy/nat:transport-protocols"
    + "/nat:protocol-id = 132";
  description
    "Extends NAT policy with a timeout for SCTP mapping
      entries.";
```

```
leaf sctp-timeout {
  type uint32;
  units "seconds";
  description
    "SCTP inactivity timeout. That is, the time an SCTP
    mapping entry will stay active without packets
    traversing the NAT.";
}

augment "/nat:nat/nat:instances/nat:instance"
  + "/nat:mapping-table/nat:mapping-entry" {
  when "nat:transport-protocol = 132";
  if-feature "sctp-nat";
  description
    "Extends the mapping entry with SCTP specifics.";

  leaf int-VTag {
    type uint32;
    description
      "The Internal Verification Tag that the internal
      host has chosen for this communication.";
  }
  leaf rem-VTag {
    type uint32;
    description
      "The Remote Verification Tag that the remote
      peer has chosen for this communication.";
  }
}
}
<CODE ENDS>
```

8. Various Examples of NAT Traversals

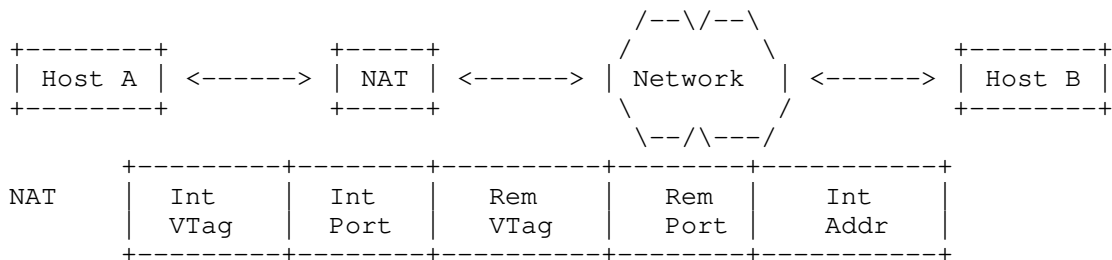
Please note that this section is informational only.

The addresses being used in the following examples are IPv4 addresses for private-use networks and for documentation as specified in [RFC6890]. However, the method described here is not limited to this NAT44 case.

The NAT binding table entries shown in the following examples do not include the flag indicating whether the restart procedure is supported or not. This flag is not relevant for these examples.

8.1. Single-homed Client to Single-homed Server

The internal client starts the association with the remote server via a four-way-handshake. Host A starts by sending a packet containing an INIT chunk.



```
INIT[Initiate-Tag = 1234]
10.0.0.1:1 -----> 203.0.113.1:2
    Rem-VTtag = 0
```

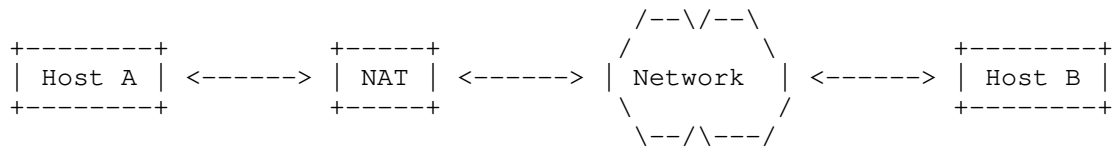
A NAT binding tabled entry is created, the source address is substituted and the packet is sent on:

NAT function creates entry:

NAT	Int VTag	Int Port	Rem VTag	Rem Port	Int Addr
	1234	1	0	2	10.0.0.1

```
INIT[Initiate-Tag = 1234]
192.0.2.1:1 -----> 203.0.113.1:2
    Rem-VTtag = 0
```

Host B receives the packet containing an INIT chunk and sends a packet containing an INIT ACK chunk with the NAT's Remote-address as destination address.



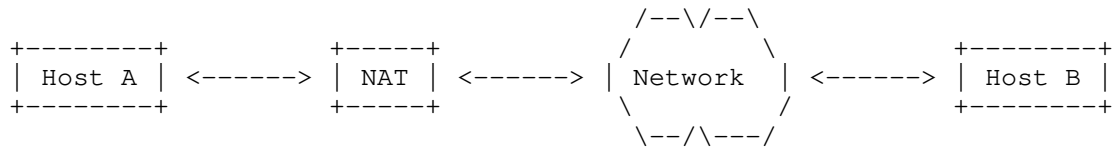
INIT ACK[Initiate-Tag = 5678]
 192.0.2.1:1 <----- 203.0.113.1:2
 Int-VTag = 1234

NAT function updates entry:

NAT	Int VTag	Int Port	Rem VTag	Rem Port	Int Addr
	1234	1	5678	2	10.0.0.1

INIT ACK[Initiate-Tag = 5678]
 10.0.0.1:1 <----- 203.0.113.1:2
 Int-VTag = 1234

The handshake finishes with a COOKIE ECHO acknowledged by a COOKIE ACK.



COOKIE ECHO
 10.0.0.1:1 -----> 203.0.113.1:2
 Rem-VTag = 5678

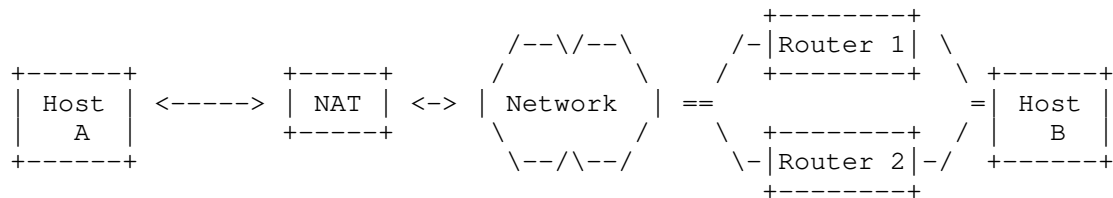
COOKIE ECHO
 192.0.2.1:1 -----> 203.0.113.1:2
 Rem-VTag = 5678

COOKIE ACK
 192.0.2.1:1 <----- 203.0.113.1:2
 Int-VTag = 1234

COOKIE ACK
 10.0.0.1:1 <----- 203.0.113.1:2
 Int-VTag = 1234

8.2. Single-homed Client to Multi-homed Server

The internal client is single-homed whereas the remote server is multi-homed. The client (Host A) sends a packet containing an INIT chunk like in the single-homed case.



NAT	Int VTag	Int Port	Rem VTag	Rem Port	Int Addr
-----	-------------	-------------	-------------	-------------	-------------

```

INIT[Initiate-Tag = 1234]
10.0.0.1:1 ---> 203.0.113.1:2
Rem-VTag = 0
  
```

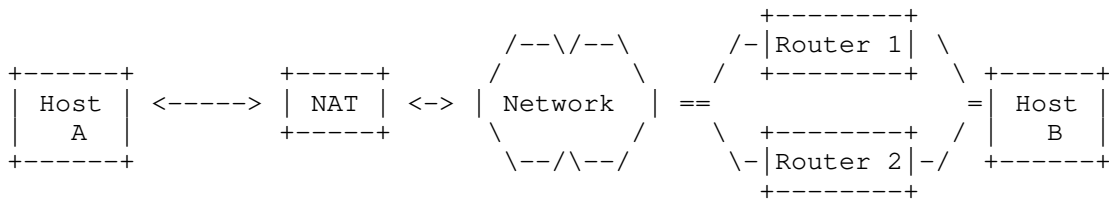
NAT function creates entry:

NAT	Int VTag	Int Port	Rem VTag	Rem Port	Int Addr
	1234	1	0	2	10.0.0.1

```

INIT[Initiate-Tag = 1234]
192.0.2.1:1 -----> 203.0.113.1:2
Rem-VTag = 0
  
```

The server (Host B) includes its two addresses in the INIT ACK chunk.



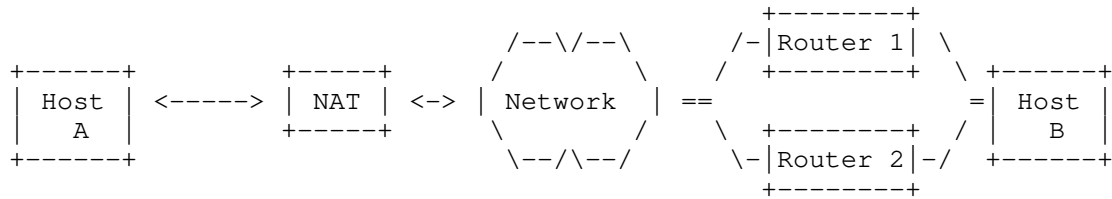
```
INIT ACK[Initiate-tag = 5678, IP-Addr = 203.0.113.129]
192.0.2.1:1 <----- 203.0.113.1:2
                        Int-VTag = 1234
```

The NAT function does not need to change the NAT binding table for the second address:

NAT	Int VTag	Int Port	Rem VTag	Rem Port	Int Addr
	1234	1	5678	2	10.0.0.1

```
INIT ACK[Initiate-Tag = 5678]
10.0.0.1:1 <--- 203.0.113.1:2
      Int-VTag = 1234
```

The handshake finishes with a COOKIE ECHO acknowledged by a COOKIE ACK.



COOKIE ECHO
 10.0.0.1:1 ---> 203.0.113.1:2
 Rem-VTag = 5678

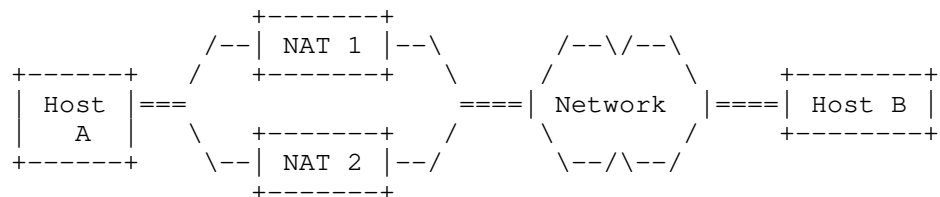
COOKIE ECHO
 192.0.2.1:1 -----> 203.0.113.1:2
 Rem-VTag = 5678

COOKIE ACK
 192.0.2.1:1 <----- 203.0.113.1:2
 Int-VTag = 1234

COOKIE ACK
 10.0.0.1:1 <--- 203.0.113.1:2
 Int-VTag = 1234

8.3. Multihomed Client and Server

The client (Host A) sends a packet containing an INIT chunk to the server (Host B), but does not include the second address.



NAT 1	Int VTag	Int Port	Rem VTag	Rem Port	Int Addr

INIT[Initiate-Tag = 1234]
 10.0.0.1:1 -----> 203.0.113.1:2
 Rem-VTag = 0

NAT function 1 creates entry:

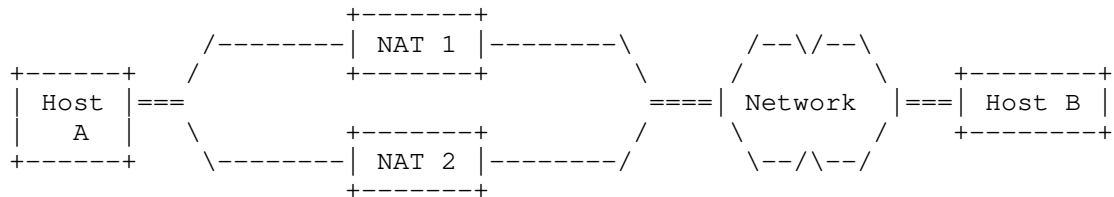
NAT 1	Int VTag	Int Port	Rem VTag	Rem Port	Int Addr
	1234	1	0	2	10.0.0.1

```

                                INIT[Initiate-Tag = 1234]
192.0.2.1:1 -----> 203.0.113.1:2
                                Rem-VTag = 0

```

Host B includes its second address in the INIT ACK.



```

INIT ACK[Initiate-Tag = 5678, IP-Addr = 203.0.113.129]
192.0.2.1:1 <----- 203.0.113.1:2
                                Int-VTag = 1234

```

NAT function 1 does not need to update the NAT binding table for the second address:

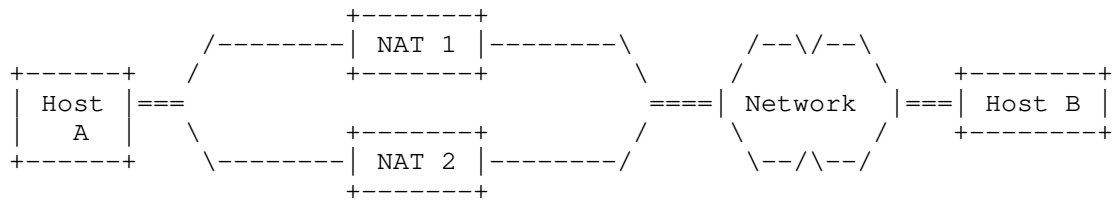
NAT 1	Int VTag	Int Port	Rem VTag	Rem Port	Int Addr
	1234	1	5678	2	10.0.0.1

```

INIT ACK[Initiate-Tag = 5678]
10.0.0.1:1 <----- 203.0.113.1:2
                                Int-VTag = 1234

```

The handshake finishes with a COOKIE ECHO acknowledged by a COOKIE ACK.



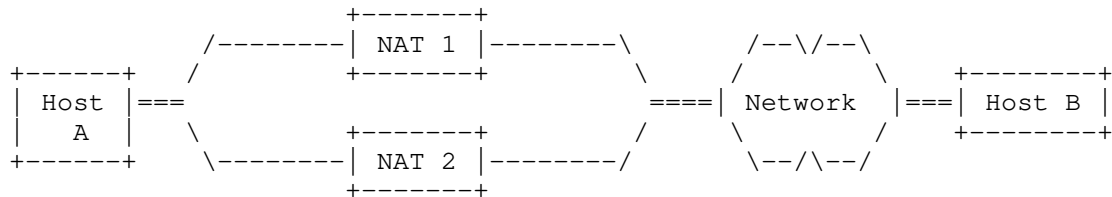
COOKIE ECHO
 10.0.0.1:1 -----> 203.0.113.1:2
 Rem-VTag = 5678

COOKIE ECHO
 192.0.2.1:1 -----> 203.0.113.1:2
 Rem-VTag = 5678

COOKIE ACK
 192.0.2.1:1 <----- 203.0.113.1:2
 Int-VTag = 1234

COOKIE ACK
 10.0.0.1:1 <----- 203.0.113.1:2
 Int-VTag = 1234

Host A announces its second address in an ASCONF chunk. The address parameter contains a wildcard address (0.0.0.0 or ::0) to indicate that the source address has to be added. The address parameter within the ASCONF chunk will also contain the pair of VTags (remote and internal) so that the NAT function can populate its NAT binding table entry completely with this single packet.



ASCONF [ADD-IP=0.0.0.0, INT-VTag=1234, Rem-VTag = 5678]
 10.1.0.1:1 -----> 203.0.113.129:2
 Rem-VTag = 5678

NAT function 2 creates a complete entry:

NAT 2	+-----+				
	Int	Int	Rem	Rem	Int
	VTag	Port	VTag	Port	Addr
	1234	1	5678	2	10.1.0.1
	+-----+				

```

ASCONF [ADD-IP, Int-VTag=1234, Rem-VTag = 5678]
192.0.2.129:1 -----> 203.0.113.129:2
                        Rem-VTag = 5678

```

```

                        ASCONF ACK
192.0.2.129:1 <----- 203.0.113.129:2
                        Int-VTag = 1234

```

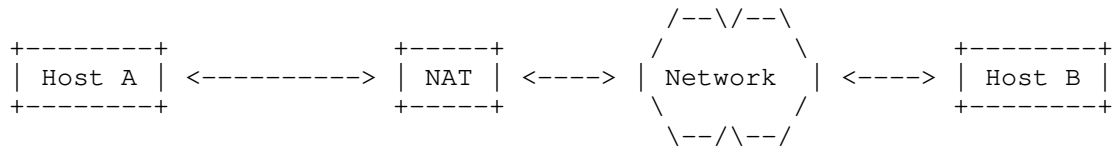
```

                        ASCONF ACK
10.1.0.1:1 <----- 203.0.113.129:2
                        Int-VTag = 1234

```

8.4. NAT Function Loses Its State

Association is already established between Host A and Host B, when the NAT function loses its state and obtains a new external address. Host A sends a DATA chunk to Host B.



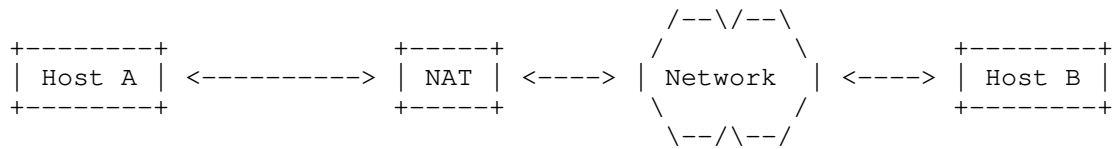
NAT	+-----+				
	Int	Int	Rem	Rem	Int
	VTag	Port	VTag	Port	Addr
	+-----+				

```

                        DATA
10.0.0.1:1 -----> 203.0.113.1:2
                        Rem-VTag = 5678

```

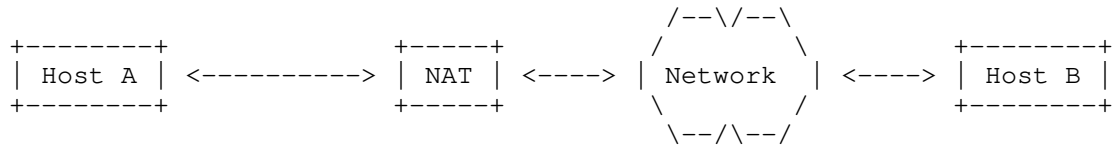
The NAT function cannot find an entry in the NAT binding table for the association. It sends a packet containing an ERROR chunk with the M bit set and the cause "NAT state missing".



```

ERROR [M bit, NAT state missing]
10.0.0.1:1 <----- 203.0.113.1:2
      Rem-VTag = 5678
  
```

On reception of the packet containing the ERROR chunk, Host A sends a packet containing an ASCONF chunk indicating that the former information has to be deleted and the source address of the actual packet added.



```

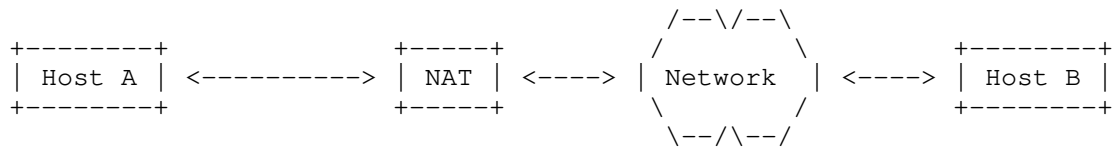
ASCONF [ADD-IP, DELETE-IP, Int-VTag=1234, Rem-VTag = 5678]
10.0.0.1:1 -----> 203.0.113.129:2
      Rem-VTag = 5678
  
```

NAT	Int VTag	Int Port	Rem VTag	Rem Port	Int Addr
	1234	1	5678	2	10.0.0.1

```

ASCONF [ADD-IP, DELETE-IP, Int-VTag=1234, Rem-VTag = 5678]
      192.0.2.2:1 -----> 203.0.113.129:2
          Rem-VTag = 5678
  
```

Host B adds the new source address to this association and deletes all other addresses from this association.



ASCONF ACK
 192.0.2.2:1 <----- 203.0.113.129:2
 Int-VTag = 1234

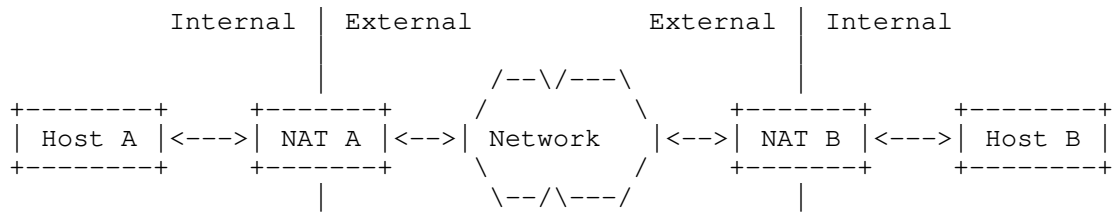
ASCONF ACK
 10.1.0.1:1 <----- 203.0.113.129:2
 Int-VTag = 1234

DATA
 10.0.0.1:1 -----> 203.0.113.1:2
 Rem-VTag = 5678

DATA
 192.0.2.2:1 -----> 203.0.113.129:2
 Rem-VTag = 5678

8.5. Peer-to-Peer Communications

If two hosts, each of them behind a NAT function, want to communicate with each other, they have to get knowledge of the peer's external address. This can be achieved with a so-called rendezvous server. Afterwards the destination addresses are external, and the association is set up with the help of the INIT collision. The NAT functions create their entries according to their internal peer's point of view. Therefore, NAT function A's Internal-VTag and Internal-Port are NAT function B's Remote-VTag and Remote-Port, respectively. The naming (internal/remote) of the verification tag in the packet flow is done from the sending host's point of view.



NAT Binding Tables

NAT A	Int VTag	Int Port	Rem VTag	Rem Port	Int Addr
-------	-------------	-------------	-------------	-------------	-------------

NAT B	Int v-tag	Int port	Rem v-tag	Rem port	Int Addr
-------	--------------	-------------	--------------	-------------	-------------

```

INIT[Initiate-Tag = 1234]
10.0.0.1:1 --> 203.0.113.1:2
    Rem-VTag = 0
  
```

NAT function A creates entry:

NAT A	Int VTag	Int Port	Rem VTag	Rem Port	Int Addr
	1234	1	0	2	10.0.0.1

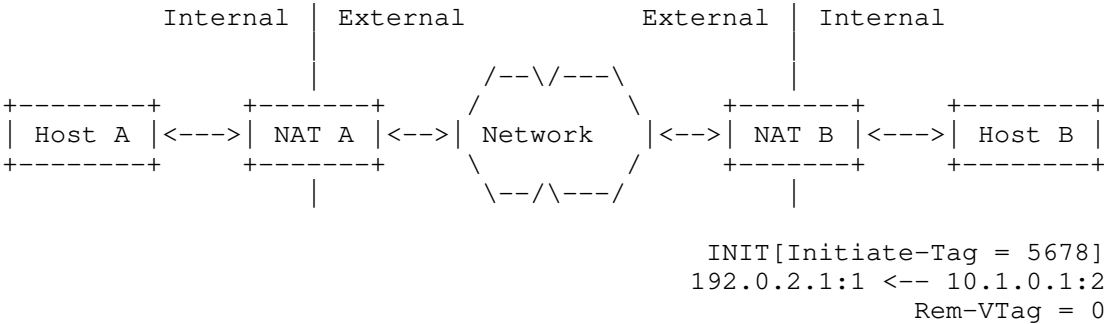
```

INIT[Initiate-Tag = 1234]
192.0.2.1:1 -----> 203.0.113.1:2
    Rem-VTag = 0
  
```

NAT function B processes the packet containing the INIT chunk, but cannot find an entry. The SCTP packet is silently discarded and leaves the NAT binding table of NAT function B unchanged.

NAT B	Int VTag	Int Port	Rem VTag	Rem Port	Int Addr
-------	-------------	-------------	-------------	-------------	-------------

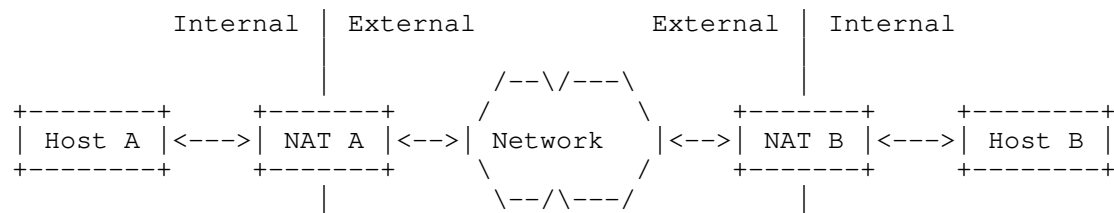
Now Host B sends a packet containing an INIT chunk, which is processed by NAT function B. Its parameters are used to create an entry.



NAT B	Int VTag	Int Port	Rem VTag	Rem Port	Int Addr
	5678	2	0	1	10.1.0.1

```
INIT[Initiate-Tag = 5678]
192.0.2.1:1 <----- 203.0.113.1:2
Rem-VTag = 0
```

NAT function A processes the packet containing the INIT chunk. As the outgoing packet containing an INIT chunk of Host A has already created an entry, the entry is found and updated:

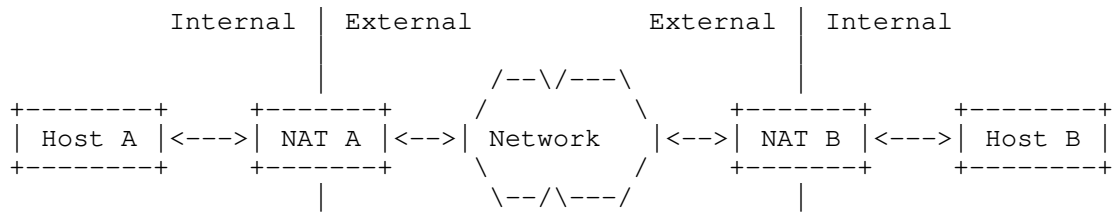


VTag != Int-VTag, but Rem-VTag == 0, find entry.

NAT A	Int VTag	Int Port	Rem VTag	Rem Port	Int Addr
	1234	1	5678	2	10.0.0.1

```
INIT[Initiate-tag = 5678]
10.0.0.1:1 <-- 203.0.113.1:2
    Rem-VTag = 0
```

Host A sends a packet containing an INIT ACK chunk, which can pass through NAT function B:



```

INIT ACK[Initiate-Tag = 1234]
10.0.0.1:1 --> 203.0.113.1:2
    Rem-VTag = 5678

```

```

    INIT ACK[Initiate-Tag = 1234]
    192.0.2.1:1 -----> 203.0.113.1:2
        Rem-VTag = 5678

```

NAT function B updates entry:

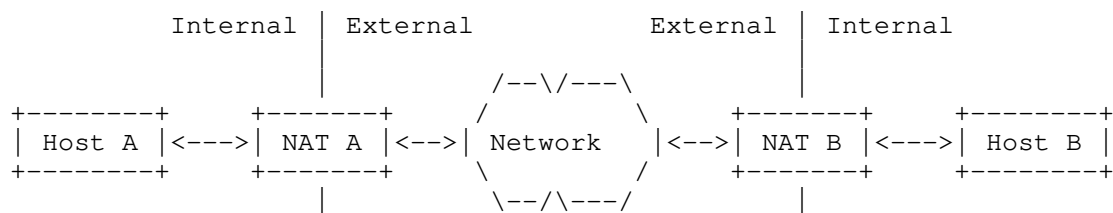
NAT B	Int	Int	Rem	Rem	Int
	VTag	Port	VTag	Port	Addr
	5678	2	1234	1	10.1.0.1

```

INIT ACK[Initiate-Tag = 1234]
    192.0.2.1:1 --> 10.1.0.1:2
        Rem-VTag = 5678

```

The lookup for COOKIE ECHO and COOKIE ACK is successful.



COOKIE ECHO
 192.0.2.1:1 <-- 10.1.0.1:2
 Rem-VTag = 1234

COOKIE ECHO
 192.0.2.1:1 <----- 203.0.113.1:2
 Rem-VTag = 1234

COOKIE ECHO
 10.0.0.1:1 <-- 203.0.113.1:2
 Rem-VTag = 1234

COOKIE ACK
 10.0.0.1:1 --> 203.0.113.1:2
 Rem-VTag = 5678

COOKIE ACK
 192.0.2.1:1 -----> 203.0.113.1:2
 Rem-VTag = 5678

COOKIE ACK
 192.0.2.1:1 --> 10.1.0.1:2
 Rem-VTag = 5678

9. Socket API Considerations

This section describes how the socket API defined in [RFC6458] is extended to provide a way for the application to control NAT friendliness.

Please note that this section is informational only.

A socket API implementation based on [RFC6458] is extended by supporting one new read/write socket option.

9.1. Get or Set the NAT Friendliness (SCTP_NAT_FRIENDLY)

This socket option uses the option_level IPPROTO_SCTP and the option_name SCTP_NAT_FRIENDLY. It can be used to enable/disable the NAT friendliness for future associations and retrieve the value for future and specific ones.

```
struct sctp_assoc_value {  
    sctp_assoc_t assoc_id;  
    uint32_t assoc_value;  
};
```

assoc_id

This parameter is ignored for one-to-one style sockets. For one-to-many style sockets the application can fill in an association identifier or SCTP_FUTURE_ASSOC for this query. It is an error to use SCTP_{CURRENT|ALL}_ASSOC in assoc_id.

assoc_value

A non-zero value indicates a NAT-friendly mode.

10. IANA Considerations

[NOTE to RFC-Editor: "RFCXXXX" is to be replaced by the RFC number you assign this document.]

[NOTE to RFC-Editor: The requested values for the chunk type and the chunk parameter types are tentative and to be confirmed by IANA.]

This document (RFCXXXX) is the reference for all registrations described in this section. The requested changes are described below.

10.1. New Chunk Flags for Two Existing Chunk Types

As defined in [RFC6096] two chunk flags have to be assigned by IANA for the ERROR chunk. The requested value for the T bit is 0x01 and for the M bit is 0x02.

This requires an update of the "ERROR Chunk Flags" registry for SCTP:

ERROR Chunk Flags

Chunk Flag Value	Chunk Flag Name	Reference
0x01	T bit	[RFCXXXX]
0x02	M bit	[RFCXXXX]
0x04	Unassigned	
0x08	Unassigned	
0x10	Unassigned	
0x20	Unassigned	
0x40	Unassigned	
0x80	Unassigned	

Table 2

As defined in [RFC6096] one chunk flag has to be assigned by IANA for the ABORT chunk. The requested value of the M bit is 0x02.

This requires an update of the "ABORT Chunk Flags" registry for SCTP:

ABORT Chunk Flags

Chunk Flag Value	Chunk Flag Name	Reference
0x01	T bit	[RFC4960]
0x02	M bit	[RFCXXXX]
0x04	Unassigned	
0x08	Unassigned	
0x10	Unassigned	
0x20	Unassigned	
0x40	Unassigned	
0x80	Unassigned	

Table 3

10.2. Three New Error Causes

Three error causes have to be assigned by IANA. It is requested to use the values given below.

This requires three additional lines in the "Error Cause Codes" registry for SCTP:

Error Cause Codes

Value	Cause Code	Reference
176	VTag and Port Number Collision	[RFCXXXX]
177	Missing State	[RFCXXXX]
178	Port Number Collision	[RFCXXXX]

Table 4

10.3. Two New Chunk Parameter Types

Two chunk parameter types have to be assigned by IANA. IANA is requested to assign these values from the pool of parameters with the upper two bits set to '11' and to use the values given below.

This requires two additional lines in the "Chunk Parameter Types" registry for SCTP:

Chunk Parameter Types

ID Value	Chunk Parameter Type	Reference
49159	Disable Restart (0xC007)	[RFCXXXX]
49160	VTags (0xC008)	[RFCXXXX]

Table 5

10.4. One New URI

An URI in the "ns" subregistry within the "IETF XML" registry has to be assigned by IANA ([RFC3688]):

URI: urn:ietf:params:xml:ns:yang:ietf-nat-sctp
 Registrant Contact: The IESG.
 XML: N/A; the requested URI is an XML namespace.

10.5. One New YANG Module

An YANG module in the "YANG Module Names" subregistry within the "YANG Parameters" registry has to be assigned by IANA ([RFC6020]):

Name: ietf-nat-sctp
 Namespace: urn:ietf:params:xml:ns:yang:ietf-nat-sctp
 Maintained by IANA: N
 Prefix: nat-sctp
 Reference: RFCXXXX

11. Security Considerations

State maintenance within a NAT function is always a subject of possible Denial Of Service attacks. This document recommends that at a minimum a NAT function runs a timer on any SCTP state so that old association state can be cleaned up.

Generic issues related to address sharing are discussed in [RFC6269] and apply to SCTP as well.

For SCTP endpoints not disabling the restart procedure, this document does not add any additional security considerations to the ones given in [RFC4960], [RFC4895], and [RFC5061].

SCTP endpoints disabling the restart procedure, need to monitor the status of all associations to mitigate resource exhaustion attacks by establishing a lot of associations sharing the same IP addresses and port numbers.

In any case, SCTP is protected by the verification tags and the usage of [RFC4895] against off-path attackers.

For IP-level fragmentation and reassembly related issues see [RFC4963].

The YANG module specified in this document defines a schema for data that is designed to be accessed via network management protocols such as NETCONF [RFC6241] or RESTCONF [RFC8040]. The lowest NETCONF layer is the secure transport layer, and the mandatory-to-implement secure transport is Secure Shell (SSH) [RFC6242]. The lowest RESTCONF layer is HTTPS, and the mandatory-to-implement secure transport is TLS [RFC8446].

The Network Configuration Access Control Model (NACM) [RFC8341] provides the means to restrict access for particular NETCONF or RESTCONF users to a preconfigured subset of all available NETCONF or RESTCONF protocol operations and content.

All data nodes defined in the YANG module that can be created, modified, and deleted (i.e., config true, which is the default) are considered sensitive. Write operations (e.g., edit-config) applied to these data nodes without proper protection can negatively affect network operations. An attacker who is able to access the SCTP NAT function can undertake various attacks, such as:

- * Setting a low timeout for SCTP mapping entries to cause failures to deliver incoming SCTP packets.
- * Instantiating mapping entries to cause NAT collision.

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A Non-Queue-Building Per-Hop Behavior (NQB PHB) for Differentiated
Services
draft-ietf-tsvwg-nqb-10

Abstract

This document specifies properties and characteristics of a Non-Queue-Building Per-Hop Behavior (NQB PHB). The purpose of this NQB PHB is to provide a separate queue that enables smooth, low-data-rate, application-limited traffic flows, which would ordinarily share a queue with bursty and capacity-seeking traffic, to avoid the latency, latency variation and loss caused by such traffic. This PHB is implemented without prioritization and without rate policing, making it suitable for environments where the use of either these features may be restricted. The NQB PHB has been developed primarily for use by access network segments, where queuing delays and queuing loss caused by Queue-Building protocols are manifested, but its use is not limited to such segments. In particular, applications to cable broadband links, Wi-Fi links, and mobile network radio and core segments are discussed. This document recommends a specific Differentiated Services Code Point (DSCP) to identify Non-Queue-Building flows.

Status of This Memo

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1. Introduction

This document defines a Differentiated Services per-hop behavior (PHB) called "Non-Queue-Building Per-Hop Behavior" (NQB PHB), which isolates traffic flows that are relatively low data rate and that do not themselves materially contribute to queueing delay and loss, allowing them to avoid the queueing delays and losses caused by other traffic. Such Non-Queue-Building flows (for example: interactive voice, gaming, machine-to-machine applications) are application limited flows that are distinguished from traffic flows managed by an end-to-end congestion control algorithm.

The vast majority of packets that are carried by broadband access networks are managed by an end-to-end congestion control algorithm, such as Reno, Cubic or BBR. These congestion control algorithms attempt to seek the available capacity of the end-to-end path (which can frequently be the access network link capacity), and in doing so generally overshoot the available capacity, causing a queue to build-up at the bottleneck link. This queue build up results in queueing delay (variable latency) and possibly packet loss that can affect all of the applications that are sharing the bottleneck link.

In contrast to traditional congestion-controlled applications, there are a variety of relatively low data rate applications that do not materially contribute to queueing delay and loss, but are nonetheless subjected to it by sharing the same bottleneck link in the access network. Many of these applications may be sensitive to latency or latency variation, as well as packet loss, and thus produce a poor quality of experience in such conditions.

Active Queue Management (AQM) mechanisms (such as PIE [RFC8033], DOCSIS-PIE [RFC8034], or CoDel [RFC8289]) can improve the quality of experience for latency sensitive applications, but there are practical limits to the amount of improvement that can be achieved without impacting the throughput of capacity-seeking applications. For example, AQMs generally allow a significant amount of queue depth variation in order to accommodate the behaviors of congestion control algorithms such as Reno and Cubic. If the AQM attempted to control the queue much more tightly, applications using those algorithms would not perform well. Alternatively, flow queueing systems, such as fq_codel [RFC8290] can be employed to isolate flows from one another, but these are not appropriate for all bottleneck links, due to complexity or other reasons.

The NQB PHB supports differentiating between these two classes of traffic in bottleneck links and queueing them separately in order that both classes can deliver satisfactory quality of experience for their applications.

To be clear, a network implementing the NQB PHB solely provides isolation for traffic classified as behaving in conformance with the NQB DSCP (and optionally enforces that behavior). It is the NQB senders' behavior itself which results in low latency and low loss.

2. Requirements Language

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

3. Context

3.1. Non-Queue-Building Behavior

There are many applications that send traffic at relatively low data rates and/or in a fairly smooth and consistent manner such that they are highly unlikely to exceed the available capacity of the network path between source and sink. These applications may themselves only cause very small, transient queues to form in network buffers, but nonetheless they can be subjected to packet delay and delay variation as a result of sharing a network buffer with applications that tend to cause large and/or standing queues to form. Many of these applications are negatively affected by excessive packet delay and delay variation. Such applications are ideal candidates to be queued separately from the applications that are the cause of queue buildup, latency and loss.

In contrast, Queue-building (QB) flows include those that use TCP or QUIC, with Cubic, Reno or other TCP congestion control algorithms that probe for the link capacity and induce latency and loss as a result. Other types of QB flows include those that frequently send at a high burst rate (e.g. several consecutive packets sent well in excess of 1 Mbps) even if the long-term average data rate is much lower.

3.2. Relationship to the Diffserv Architecture

The IETF has defined the Differentiated Services architecture [RFC2475] with the intention that it allows traffic to be marked in a manner that conveys the performance requirements of that traffic either quantitatively or in a relative sense (i.e. priority). The architecture defines the use of the Diffserv field [RFC2474] for this purpose, and numerous RFCs have been written that describe recommended interpretations of the values (Diffserv Code Points) of the field, and standardized treatments (traffic conditioning and per-

hop-behaviors) that can be implemented to satisfy the performance requirements of traffic so marked.

While this architecture is powerful, and can be configured to meet the performance requirements of a variety of applications and traffic categories, or to achieve differentiated service offerings, it has proven problematic to enable its use for these purposes end-to-end across the Internet.

This difficulty is in part due to the fact that meeting (in an end-to-end context) the performance requirements of an application involves all of the networks in the path agreeing on what those requirements are, and sharing an interest in meeting them. In many cases this is made more difficult due to the fact that the performance "requirements" are not strict ones (e.g. applications will degrade in some manner as loss/latency/jitter increase), so the importance of meeting them for any particular application in some cases involves a judgment as to the value of avoiding some amount of degradation in quality for that application in exchange for an increase in the degradation of another application.

Further, in many cases the implementation of Diffserv PHBs has historically involved prioritization of service classes with respect to one another, which sets up the zero-sum game alluded to in the previous paragraph, and results in the need to limit access to higher priority classes via mechanisms such as access control, admission control, traffic conditioning and rate policing, and/or to meter and bill for carriage of such traffic. These mechanisms can be difficult or impossible to implement in an end-to-end context.

Finally, some jurisdictions impose regulations that limit the ability of networks to provide differentiation of services, in large part based on the belief that doing so necessarily involves prioritization or privileged access to bandwidth, and thus a benefit to one class of traffic always comes at the expense of another.

In contrast, the NQB PHB has been designed with the goal that it avoids many of these issues, and thus could conceivably be deployed end-to-end across the Internet. The intent of the NQB DSCP is that it signals verifiable behavior that permits the sender to request differentiated treatment. Also, the NQB traffic is to be given a separate queue with priority equal to default traffic, and given no reserved bandwidth other than the bandwidth that it shares with default traffic. As a result, the NQB PHB does not aim to meet specific application performance requirements. Instead the goal of the NQB PHB is to provide statistically better loss, latency, and jitter performance for traffic that is itself only an insignificant contributor to those degradations. The PHB is also designed to

minimize any incentives for a sender to mismark its traffic, since neither higher priority nor reserved bandwidth are being offered. These attributes eliminate many of the tradeoffs that underlie the handling of differentiated service classes in the Diffserv architecture as it has traditionally been defined. They also significantly simplify access control and admission control functions, reducing them to simple verification of behavior.

3.3. Relationship to L4S

The NQB DSCP and PHB described in this draft have been defined to operate independently of the experimental L4S Architecture [I-D.ietf-tsvwg-l4s-arch]. Nonetheless, the NQB traffic flows are intended to be compatible with [I-D.ietf-tsvwg-l4s-arch], with the result being that NQB traffic and L4S traffic can share the low-latency queue in an L4S DualQ node [I-D.ietf-tsvwg-aqm-dualq-coupled]. Compliance with the DualQ Coupled AQM requirements (Section 2.5 of [I-D.ietf-tsvwg-aqm-dualq-coupled]) is considered sufficient to support the NQB PHB requirement of fair allocation of bandwidth between the QB and NQB queues (Section 5).

4. DSCP Marking of NQB Traffic

4.1. Non-Queue-Building Sender Requirements

Non-queue-building (NQB) flows are typically UDP flows that don't seek the maximum capacity of the link (examples: online games, voice chat, DNS lookups, real-time IoT analytics data). Here the data rate is limited by the application itself rather than by network capacity - these applications send, at most, the equivalent of a few well-spaced packets per RTT, even if the packets are not actually RTT-clocked. In today's network this corresponds to an instantaneous data rate (packet size divided by packet inter-arrival time) of no more than about 1 Mbps (e.g. no more than one 1250 B packet every 10 ms), but there is no precise bound since it depends on the conditions in which the application is operating.

Note that, while such flows ordinarily don't implement a traditional congestion control mechanism, they nonetheless are expected to comply with existing guidance for safe deployment on the Internet, for example the requirements in [RFC8085] and Section 2 of [RFC3551] (also see the circuit breaker limits in Section 4.3 of [RFC8083] and the description of inelastic pseudowires in Section 4 of [RFC7893]). To be clear, the description of NQB flows in this document should not be interpreted as suggesting that such flows are in any way exempt from this responsibility.

Applications that align with the description of NQB behavior in the preceding paragraphs SHOULD identify themselves to the network using a Diffserv Code Point (DSCP) of 45 (decimal) so that their packets can be queued separately from QB flows. The choice of the value 45 is motivated in part by the desire to achieve separate queuing in existing WiFi networks (see Section 8.3) and by the desire to make implementation of the PHB simpler in network gear that has the ability to classify traffic based on ranges of DSCP value (see Section 4.2 for further discussion). In networks where another (e.g. a local-use) codepoint is designated for NQB traffic, or where specialized PHBs are available that can meet specific application requirements (e.g. a guaranteed-latency path for voice traffic), it may be preferred to use another DSCP. In end systems where the choice of using DSCP 45 is not available to the application, the CS5 DSCP (40 decimal) could be used as a fallback. See Section 4.2 for rationale as to why this choice could be fruitful.

If the application's traffic exceeds more than a few packets per RTT, or exceeds a fairly small fraction of the expected path capacity on an instantaneous (e.g. inter-packet or a suitably short time interval) basis, the application SHOULD NOT mark its traffic with the NQB DSCP. In such a case, the application could instead consider implementing a low latency congestion control mechanism as described in [I-D.ietf-tsvwg-ecn-l4s-id]. At the time of writing, it is believed that 1 Mbps is a reasonable upper bound on instantaneous traffic rate for an NQB-marked application, but this value is of course subject to the context in which the application is expected to be deployed.

An application that marks its traffic as NQB but happens to exceed the available path capacity (even on an instantaneous basis) runs the risk of being subjected to a Traffic Protection algorithm (see Section 5.2), which could result in the excess traffic being discarded or queued separately as default traffic (and thus potentially delivered out of order). As a result, applications that aren't clearly beneath the threshold described above would need to weigh the risk of additional loss or out-of-order delivery against the expected latency benefits of NQB treatment in determining whether or not to mark their packets as NQB.

4.2. Aggregation of the NQB DSCP with other Diffserv PHBs

It is RECOMMENDED that networks and nodes that do not support the NQB PHB be configured to treat NQB marked traffic the same as traffic marked "Default". It is additionally RECOMMENDED that such networks and nodes simply classify the NQB DSCP into the same treatment aggregate as Default traffic, or encapsulate the NQB marked packet, rather than re-marking NQB traffic as Default. This preservation of

the NQB marking enables hops further along the path to provide the NQB PHB successfully. Section 4.3 discusses re-marking of NQB traffic to an alternate DSCP value of 5 within core networks as a means to facilitate this for networks where this behavior can be more readily implemented for such a value.

In backbone and core network switches (particularly if shallow-buffered), and nodes that do not typically experience congestion, treating NQB marked traffic the same as Default may be sufficient to preserve loss/latency/jitter performance for NQB traffic. In other nodes, treating NQB marked traffic as Default could result in degradation of loss/latency/jitter performance but is recommended nonetheless in order to preserve the incentives described in Section 5. An alternative, in controlled environments where there is no risk of mismarking of traffic, would be to aggregate NQB marked traffic with real-time, latency sensitive traffic. Similarly, networks and nodes that aggregate service classes as discussed in [RFC5127] and [RFC8100] may not be able to provide a PDB/PHB that meets the requirements of this document. In these cases it is RECOMMENDED that NQB-marked traffic be aggregated into the Elastic Treatment Aggregate (for [RFC5127] networks) or the Default / Elastic Treatment Aggregate (for [RFC8100] networks), although in some cases a network operator may instead choose to aggregate NQB traffic into the (Bulk) Real-Time Treatment Aggregate. Either approach comes with trade-offs: when the aggregated traffic encounters a bottleneck, aggregating with Default/Elastic traffic could result in a degradation of loss/latency/jitter performance for NQB traffic, while aggregating with Real-Time (assuming such traffic is provided a prioritized PHB) risks creating an incentive for mismarking of non-compliant traffic as NQB (except in controlled environments). In either case, the NQB DSCP SHOULD be preserved (possibly via encapsulation) in order to limit the negative impact that such networks would have on end-to-end performance for NQB traffic. This aligns with recommendations in [RFC5127].

Nodes that support the NQB PHB may choose to aggregate other service classes into the NQB queue. This is particularly useful in cases where specialized PHBs for these other service classes are not provided. Candidate service classes for this aggregation would include those that carry inelastic traffic that has low to very-low tolerance for loss, latency and/or jitter as discussed in [RFC4594]. These could include Telephony (EF/VA), Signaling (CS5), Real-Time Interactive (CS4) and Broadcast Video (CS3). Or, in some networks, equipment limitations may necessitate aggregating all traffic marked with DSCPs 40-47 (i.e., whose three MSBs are 0b101). As noted in Section 4.1, the choice of the value 45 is motivated in part by the desire to make this aggregation simpler in network equipment that can classify packets via comparing the DSCP value to a range of configured values.

4.3. End-to-end usage and DSCP Re-marking

In contrast to some existing standard PHBs, many of which are typically only meaningful within a Diffserv Domain (e.g. an AS or an enterprise network), this PHB is expected to be used end-to-end across the Internet, wherever suitable operator agreements apply. Under the [RFC2474] model, this requires that the corresponding DSCP is recognized by all operators and mapped across their boundaries accordingly.

To support NQB, networks MUST preserve a DSCP marking distinction between NQB traffic and Default traffic when forwarding via an interconnect from or to another network. To facilitate the default treatment of NQB traffic in backbones and core networks discussed in the previous section (where IP Precedence may be deployed), networks that support NQB SHOULD NOT use the value 45 for NQB at network interconnects unless that usage is explicitly documented in the TCA (Traffic Conditioning Agreement, see [RFC2475]) for that interconnection. Rather, networks SHOULD remap NQB traffic to DSCP 5 prior to interconnection, unless agreed otherwise between the interconnecting partners. To be clear, interconnecting networks are not precluded from negotiating (via an SLA, TCA, or some other agreement) a different DSCP to use to signal NQB across an interconnect. Additionally, the fact that this PHB is intended for end-to-end usage does not preclude networks from mapping the NQB DSCP to a value other than 45 or 5 for internal usage, as long as the appropriate NQB DSCP is restored when forwarding to another network.

Furthermore, in other network environments where IP Precedence ([RFC0791]) is deployed, it is RECOMMENDED that the network operator re-mark NQB traffic to DSCP 5 in order to ensure that it is aggregated with Default traffic.

In order to enable interoperability with WiFi equipment as described in Section 8.3.1, networks SHOULD re-mark NQB traffic (e.g. DSCP 5) to DSCP 45 prior to a customer access link, subject to the safeguards described below and in that section.

Thus, this document recommends two DSCPs to designate NQB, the value 45 for use by hosts and in WiFi networks, and the value 5 for use across network interconnections.

4.3.1. Unmanaged Networks

In cases where a network operator is delivering traffic into an unmanaged network outside of their control (e.g. a residential ISP delivering traffic to a customer's home network), the network operator should presume that the existing network equipment in the user network supports the WiFi default DSCP mapping (see Section 8.3) and/or IP Precedence, and does not support the safeguards that are provided by the NQB PHB requirements. As a result, the network operator should take precautions to prevent issues. When the data rate of the access network segment is less than the expected data rate of the user network, this is unlikely to be an issue. However, if the access network rate exceeds the expected rate of the user network, the operator SHOULD deploy a policing function on NQB marked traffic that minimizes the potential for negative impacts on traffic marked Default, for example by limiting the rate of such traffic to a set fraction of the customer's service rate, with excess traffic either dropped or re-marked as Default.

As an additional safeguard, and to prevent the inadvertent introduction of problematic traffic into unmanaged user networks, network equipment that is intended to deliver traffic into unmanaged user networks (e.g. an access network gateway for a residential ISP) MUST by default ensure that NQB traffic is marked with a DSCP that selects the WiFi "Best Effort" Access Category and the "Routine" IP Precedence level (i.e. DSCP 0-7). Such equipment MUST support the ability to configure the remapping, so that (when appropriate safeguards are in place) traffic can be delivered as NQB-marked.

4.4. The NQB DSCP and Tunnels

[RFC2983] discusses tunnel models that support Diffserv. It describes a "uniform model" in which the inner DSCP is copied to the outer header at encapsulation, and the outer DSCP is copied to the inner header at decapsulation. It also describes a "pipe model" in which the outer DSCP is not copied to the inner header at decapsulation. Both models can be used in conjunction with the NQB PHB. In the case of the pipe model, any DSCP manipulation (re-marking) of the outer header by intermediate nodes would be discarded

at tunnel egress, potentially improving the possibility of achieving NQB treatment in subsequent nodes.

As is discussed in [RFC2983], tunnel protocols that are sensitive to reordering can result in undesirable interactions if multiple DSCP PHBs are signaled for traffic within a tunnel instance. This is true for NQB marked traffic as well. If a tunnel contains a mix of QB and NQB traffic, and this is reflected in the outer DSCP in a network that supports the NQB PHB, it would be necessary to avoid a reordering-sensitive tunnel protocol.

5. Non-Queue-Building PHB Requirements

It is worthwhile to note again that the NQB designation and marking is intended to convey verifiable traffic behavior, as opposed to simply a desire for differentiated treatment. Also, it is important that incentives are aligned correctly, i.e. that there is a benefit to the application in marking its packets correctly, and a disadvantage (or at least no benefit) to an application in intentionally mismarking its traffic. Thus, a useful property of nodes (i.e. network switches and routers) that support separate queues for NQB and QB flows is that for NQB flows, the NQB queue provides better performance than the QB queue; and for QB flows, the QB queue provides better performance than the NQB queue (this is discussed further in this section and Section 11). By adhering to these principles, there is no incentive for senders to mismark their traffic as NQB, and further, any mismarking can be identified by the network.

5.1. Primary Requirements

A node supporting the NQB PHB makes no guarantees on latency or data rate for NQB marked flows, but instead aims to provide a bound on queuing delay for as many such marked flows as it can, and shed load when needed.

A node supporting the NQB PHB **MUST** provide a queue for non-queue-building traffic separate from any queue used for queue-building traffic.

NQB traffic, in aggregate, **SHOULD NOT** be rate limited or rate policed separately from queue-building traffic of equivalent importance.

The NQB queue **SHOULD** be given equivalent forwarding preference compared to queue-building traffic of equivalent importance. The node **SHOULD** provide a scheduler that allows QB and NQB traffic of equivalent importance to share the link in a fair manner, e.g. a deficit round-robin scheduler with equal weights. Compliance with

these recommendations helps to ensure that there are no incentives for QB traffic to be mismarked as NQB. In environments where mismarking is not a potential issue (e.g. a network where a marking policy is enforced by other means), these requirements may not be necessary.

A node supporting the NQB PHB SHOULD treat traffic marked as Default (DSCP=0) as QB traffic having equivalent importance to the NQB marked traffic. A node supporting the NQB DSCP MUST support the ability to configure the classification criteria that are used to identify QB and NQB traffic of equivalent importance.

The NQB queue SHOULD have a buffer size that is significantly smaller than the buffer provided for QB traffic (e.g. single-digit milliseconds). It is expected that most QB traffic is engineered to work well when the network provides a relatively deep buffer (e.g. on the order of tens or hundreds of ms) in nodes where support for the NQB PHB is advantageous (i.e. bottleneck nodes). Providing a similarly deep buffer for the NQB queue would be at cross purposes to providing very low queueing delay, and would erode the incentives for QB traffic to be marked correctly.

5.2. Traffic Protection

It is possible that due to an implementation error or misconfiguration, a QB flow would end up getting mismarked as NQB, or vice versa. In the case of an NQB flow that isn't marked as NQB and ends up in the QB queue, it would only impact its own quality of service, and so it seems to be of lesser concern. However, a QB flow that is mismarked as NQB would cause queueing delays and/or loss for all of the other flows that are sharing the NQB queue.

To prevent this situation from harming the performance of the real NQB flows, network elements that support differentiating NQB traffic SHOULD support a "traffic protection" function that can identify QB flows that are mismarked as NQB, and either reclassify those flows/packets to the QB queue or discard the offending traffic. Such a function SHOULD be implemented in an objective and verifiable manner, basing its decisions upon the behavior of the flow rather than on application-layer constructs. It is RECOMMENDED that traffic protection algorithms base their decisions on the detection of actual queueing, as opposed to simply packet arrival rates. It may be advantageous for a traffic protection function to employ hysteresis to prevent borderline flows from being reclassified capriciously.

One example traffic protection algorithm can be found in [I-D.briscoe-docsis-q-protection].

There are some situations where such function may not be necessary. For example, a network element designed for use in controlled environments (e.g. enterprise LAN) may not require a traffic protection function. Additionally, some networks may prefer to police the application of the NQB DSCP at the ingress edge, so that in-network traffic protection is not needed.

5.3. Guidance for Very Low Rate Links

The NQB sender requirements in Section 4.1 place responsibility in the hands of the application developer to determine the likelihood that the application's sending behavior could result in a queue forming along the path. These requirements rely on application developers having a reasonable sense for the network context in which their application is to be deployed. Even so, there will undoubtedly be networks that contain links having a data rate that is below the lower end of what is considered "typical", and some of these links may even be below the instantaneous sending rate of some NQB-marked applications.

To limit the consequences of this scenario, operators of such networks SHOULD utilize a traffic protection function that is more tolerant of burstiness (i.e. a temporary queue). Alternatively, operators of such networks MAY choose to disable NQB support on these low speed links. In particular, for links that are far below "typical" path rates, it is RECOMMENDED that NQB support be disabled.

6. Impact on Higher Layer Protocols

Network elements that support the NQB PHB and that support traffic protection as discussed in the previous section introduce the possibility that flows classified into the NQB queue could experience out of order delivery or packet loss if their behavior is not consistent with NQB. This is particularly true if the traffic protection algorithm makes decisions on a packet-by-packet basis. In this scenario, a flow that is (mis)marked as NQB and that causes a queue to form in this bottleneck link could see some of its packets forwarded by the NQB queue, and some of them either discarded or redirected to the QB queue. In the case of redirection, depending on the queueing latency and scheduling within the network element, this could result in packets being delivered out of order. As a result, the use of the NQB DSCP by a higher layer protocol carries some risk that an increased amount of out of order delivery or packet loss will be experienced. This characteristic provides one disincentive for mis-marking of traffic.

7. Configuration and Management

As required above, nodes supporting the NQB PHB provide for the configuration of classifiers that can be used to differentiate between QB and NQB traffic of equivalent importance. The default for such classifiers is recommended to be the assigned NQB DSCP (to identify NQB traffic) and the Default (0) DSCP (to identify QB traffic).

8. Example Use Cases

8.1. DOCSIS Access Networks

Residential cable broadband Internet services are commonly configured with a single bottleneck link (the access network link) upon which the service definition is applied. The service definition, typically an upstream/downstream data rate tuple, is implemented as a configured pair of rate shapers that are applied to the user's traffic. In such networks, the quality of service that each application receives, and as a result, the quality of experience that it generates for the user is influenced by the characteristics of the access network link.

To support the NQB PHB, cable broadband services **MUST** be configured to provide a separate queue for NQB marked traffic. The NQB queue **MUST** be configured to share the service's rate shaped bandwidth with the queue for QB traffic.

8.2. Mobile Networks

Historically, 3GPP mobile networks have utilised "bearers" to encapsulate each user's user plane traffic through the radio and core networks. A "dedicated bearer" may be allocated a Quality of Service (QoS) to apply any prioritisation to its flows at queues and radio schedulers. Typically an LTE operator provides a dedicated bearer for IMS VoLTE (Voice over LTE) traffic, which is prioritised in order to meet regulatory obligations for call completion rates; and a "best effort" default bearer, for Internet traffic. The "best effort" bearer provides no guarantees, and hence its buffering characteristics are not compatible with low-latency traffic. The 5G radio and core systems offer more flexibility over bearer allocation, meaning bearers can be allocated per traffic type (e.g. loss-tolerant, low-latency etc.) and hence support more suitable treatment of Internet real-time flows.

To support the NQB PHB, the mobile network SHOULD be configured to give UEs a dedicated, low-latency, non-GBR, EPS bearer, e.g. one with QCI 7, in addition to the default EPS bearer; or a Data Radio Bearer with 5QI 7 in a 5G system (see Table 5.7.4-1: Standardized 5QI to QoS characteristics mapping in [SA-5G]).

A packet carrying the NQB DSCP SHOULD be routed through the dedicated low-latency EPS bearer. A packet that has no associated NQB marking SHOULD NOT be routed through the dedicated low-latency EPS bearer.

8.3. WiFi Networks

WiFi networking equipment compliant with 802.11e/n/ac/ax [IEEE802-11] generally supports either four or eight transmit queues and four sets of associated Enhanced Multimedia Distributed Control Access (EDCA) parameters (corresponding to the four WiFi Multimedia (WMM) Access Categories) that are used to enable differentiated media access characteristics. As discussed in [RFC8325], most existing WiFi implementations use a default DSCP to User Priority mapping that utilizes the most significant three bits of the Diffserv Field to select "User Priority" which is then mapped to the four WMM Access Categories. [RFC8325] also provides an alternative mapping that more closely aligns with the DSCP recommendations provided by the IETF.

In addition to the requirements provided in other sections of this document, to support the NQB PHB, WiFi equipment (including equipment compliant with [RFC8325]) SHOULD map the NQB codepoint 45 into a separate queue in the same Access Category as the queue that carries default traffic (i.e. the Best Effort Access Category).

8.3.1. Interoperability with Existing WiFi Networks

While some existing WiFi equipment may be capable (in some cases via firmware update) of supporting the NQB PHB requirements, many currently deployed devices cannot be configured in this way. As a result the remainder of this section discusses interoperability with these existing WiFi networks, as opposed to PHB compliance.

In order to increase the likelihood that NQB traffic is provided a separate queue from QB traffic in existing WiFi equipment that uses the default mapping, the 45 code point is recommended for NQB. This maps NQB to UP_5 which is in the "Video" Access Category. While this DSCP to User Priority mapping enables these WiFi systems to support the NQB PHB requirement for segregated queuing, it does not support the remaining NQB PHB requirements in Section 5. The ramifications of, and remedies for this are discussed further below.

Existing WiFi devices are unlikely to support a traffic protection algorithm, so traffic mismarked as NQB is not likely to be detected and remedied by such devices.

Furthermore, in their default configuration, existing WiFi devices utilize EDCA parameters that result in statistical prioritization of the "Video" Access Category above the "Best Effort" Access Category. If left unchanged, this would violate the NQB PHB requirement for equal prioritization, and could erode the principle of alignment of incentives. In order to preserve the incentives principle for NQB, WiFi systems SHOULD be configured such that the EDCA parameters for the Video Access Category match those of the Best Effort Access Category. This recommendation is primarily applicable to WiFi systems deployed in a managed environment or those deployed by an ISP, and is intended for situations when the vast majority of traffic that would use AC_VI is NQB. In other situations (e.g. consumer-grade WiFi gear deployed by an ISP's customer) this configuration may not be possible, and the requirements and recommendations in Section 4.3.1 would apply.

Similarly, systems that utilize [RFC8325] but that are unable to fully support the PHB requirements, SHOULD map the recommended NQB code point 45 (or the locally determined alternative) to UP_5 in the "Video" Access Category.

9. Acknowledgements

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10. IANA Considerations

This document requests that IANA assign the Differentiated Services Field Codepoint (DSCP) 45 ('0b101101', 0x2D) from the "Differentiated Services Field Codepoints (DSCP)" registry (<https://www.iana.org/assignments/dscp-registry/>) ("DSCP Pool 3 Codepoints", Codepoint Space xxxx01, Standards Action) as the RECOMMENDED codepoint for Non-Queue-Building behavior for end-systems and in edge networks.

IANA should update this registry as follows:

* Name: NQB-EDGE

- * Value (Binary): 101101
- * Value (Decimal): 45
- * Reference: this document

This document requests that IANA assign the Differentiated Services Field Codepoint (DSCP) 5 ('0b000101', 0x05) from the "Differentiated Services Field Codepoints (DSCP)" registry (<https://www.iana.org/assignments/dscp-registry/>) ("DSCP Pool 3 Codepoints", Codepoint Space xxxx01, Standards Action) as the RECOMMENDED codepoint for Non-Queue-Building behavior in core networks and interconnections.

IANA should update this registry as follows:

- * Name: NQB-CORE
- * Value (Binary): 000101
- * Value (Decimal): 5
- * Reference: this document

11. Security Considerations

When the NQB PHB is fully supported in bottleneck links, there is no incentive for a queue-building application to mismark its packets as NQB (or vice versa). If a queue-building flow were to mark its packets as NQB, it would be unlikely to receive a benefit by doing so, and it could experience excessive packet loss, excessive latency variation and/or excessive out-of-order delivery (depending on the nature of the traffic protection function). If a non-queue-building flow were to fail to mark its packets as NQB, it could suffer the latency and loss typical of sharing a queue with capacity seeking traffic.

In order to preserve low latency performance for NQB traffic, networks that support the NQB PHB will need to ensure that mechanisms are in place to prevent malicious NQB-marked traffic from causing excessive queue delays. This document recommends the implementation of a traffic protection mechanism to achieve this goal, but recognizes that other options may be more desirable in certain situations.

Notwithstanding the above, the choice of DSCP for NQB does allow existing WiFi networks to readily (and by default) support some of the PHB requirements, but without a traffic protection function, and

(when left in the default state) by giving NQB traffic higher priority than QB traffic. This does open up the NQB marking to potential abuse on these WiFi links, but since these existing WiFi networks already give one quarter of the DSCP space this same treatment, and further they give another quarter of the DSCP space even higher priority, the NQB DSCP does not seem to be of any greater risk for abuse than these others.

The NQB signal is not integrity protected and could be flipped by an on-path attacker. This might negatively affect the QoS of the tampered flow.

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Appendix A. DSCP Remarking Pathologies

Some network operators typically bleach (zero out) the Diffserv field on ingress into their network

[I-D.ietf-tsvwg-dscp-considerations][Custura][Barik], and in some cases apply their own DSCP for internal usage. Bleaching the NQB DSCP is not expected to cause harm to default traffic, but it will severely limit the ability to provide NQB treatment end-to-end. Reports on existing deployments of DSCP manipulation [Custura][Barik] categorize the re-marking behaviors into the following six policies: bleach all traffic (set DSCP to zero), set the top three bits (the former Precedence bits) on all traffic to 0b000, 0b001, or 0b010, set the low three bits on all traffic to 0b000, or remark all traffic to a particular (non-zero) DSCP value.

Regarding the DSCP values of 5 & 45, there were no observations of DSCP manipulation reported in which traffic was marked 5 or 45 by any of these policies. Thus it appears that these re-marking policies would be unlikely to result in QB traffic being marked as NQB (45). In terms of the fate of NQB-marked traffic that is subjected to one of these policies, the result would be that NQB marked traffic would be indistinguishable from some subset (possibly all) of other traffic. In the policies where all traffic is remarked using the same (zero or non-zero) DSCP, the ability for a subsequent network hop to differentiate NQB traffic via DSCP would clearly be lost entirely.

In the policies where the top three bits are overwritten, both NQB values (5 & 45) would receive the same marking as would the currently unassigned Pool 3 DSCPs 13,21,29,37,53,61, with all of these code points getting mapped to DSCP=5, 13 or 21 (depending on the overwrite value used). Since none of the DSCPs in the preceding lists are currently assigned by IANA, and they all are set aside for Standards Action, it is believed that they are not widely used currently, but this may vary based on local-usage.

For the policy in which the low three bits are set to 0b000, the NQB (45) value would be mapped to CS5 and would be indistinguishable from CS5, VA, EF (and the unassigned DSCPs 41, 42, 43). Traffic marked using the existing standardized DSCPs in this list are likely to share the same general properties as NQB traffic (non capacity-seeking, very low data rate or relatively low and consistent data rate). Similarly, any future recommended usage for DSCPs 41, 42, 43 would likely be somewhat compatible with NQB treatment, assuming that IP Precedence compatibility (see Section 1.5.4 of [RFC4594]) is maintained in the future. Here there may be an opportunity for a node to provide the NQB PHB or the CS5 PHB to CS5-marked traffic and retain some of the benefits of NQB marking. This could be another motivation to (as discussed in Section 4.2) classify CS5-marked traffic into NQB queue. For this same re-marking policy, the NQB (5) value would be mapped to CS0/default and would be indistinguishable from CS0, LE (and the unassigned DSCPs 2,3,4,6,7). In this case, NQB traffic is likely to be given default treatment in all subsequent nodes, which would eliminate the ability to provide NQB treatment in those nodes, but would be relatively harmless otherwise.

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Stream Control Transmission Protocol
draft-ietf-tsvwg-rfc4960-bis-19

Abstract

This document obsoletes RFC 4960, if approved. It describes the Stream Control Transmission Protocol (SCTP) and incorporates the specification of the chunk flags registry from RFC 6096 and the specification of the I bit of DATA chunks from RFC 7053. Therefore, RFC 6096 and RFC 7053 are also obsoleted by this document, if approved. In addition to that, the Errata documents RFC 4460 and RFC 8540 are also obsoleted by this document, if approved.

SCTP was originally designed to transport Public Switched Telephone Network (PSTN) signaling messages over IP networks. It is also suited to be used for other applications, for example WebRTC.

SCTP is a reliable transport protocol operating on top of a connectionless packet network such as IP. It offers the following services to its users:

- * acknowledged error-free non-duplicated transfer of user data,
- * data fragmentation to conform to discovered path maximum transmission unit (PMTU) size,
- * sequenced delivery of user messages within multiple streams, with an option for order-of-arrival delivery of individual user messages,
- * optional bundling of multiple user messages into a single SCTP packet, and
- * network-level fault tolerance through supporting of multi-homing at either or both ends of an association.

The design of SCTP includes appropriate congestion avoidance behavior and resistance to flooding and masquerade attacks.

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1. Conventions

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

2. Introduction

This section explains the reasoning behind the development of the Stream Control Transmission Protocol (SCTP), the services it offers, and the basic concepts needed to understand the detailed description of the protocol.

This document obsoletes [RFC4960], if approved. In addition to that, it incorporates the specification of the chunk flags registry from [RFC6096] and the specification of the I bit of DATA chunks from [RFC7053]. Therefore, [RFC6096] and [RFC7053] are also obsoleted by this document, if approved.

2.1. Motivation

TCP [RFC0793] has performed immense service as the primary means of reliable data transfer in IP networks. However, an increasing number of recent applications have found TCP too limiting, and have incorporated their own reliable data transfer protocol on top of UDP [RFC0768]. The limitations that users have wished to bypass include the following:

- * TCP provides both reliable data transfer and strict order-of-transmission delivery of data. Some applications need reliable transfer without sequence maintenance, while others would be satisfied with partial ordering of the data. In both of these cases, the head-of-line blocking offered by TCP causes unnecessary delay.
- * The stream-oriented nature of TCP is often an inconvenience. Applications add their own record marking to delineate their messages, and make explicit use of the push facility to ensure that a complete message is transferred in a reasonable time.
- * The limited scope of TCP sockets complicates the task of providing highly-available data transfer capability using multi-homed hosts.
- * TCP is relatively vulnerable to denial-of-service attacks, such as SYN attacks.

Transport of PSTN signaling across the IP network is an application for which all of these limitations of TCP are relevant. While this application directly motivated the development of SCTP, other applications might find SCTP a good match to their requirements. One example of this is the use of datachannels in the WebRTC infrastructure.

2.2. Architectural View of SCTP

SCTP is viewed as a layer between the SCTP user application ("SCTP user" for short) and a connectionless packet network service such as IP. The remainder of this document assumes SCTP runs on top of IP. The basic service offered by SCTP is the reliable transfer of user messages between peer SCTP users. It performs this service within the context of an association between two SCTP endpoints. Section 11 of this document sketches the API that exists at the boundary between the SCTP and the SCTP upper layers.

SCTP is connection-oriented in nature, but the SCTP association is a broader concept than the TCP connection. SCTP provides the means for each SCTP endpoint (Section 2.3) to provide the other endpoint

(during association startup) with a list of transport addresses (i.e., multiple IP addresses in combination with an SCTP port) through which that endpoint can be reached and from which it will originate SCTP packets. The association spans transfers over all of the possible source/destination combinations that can be generated from each endpoint's lists.

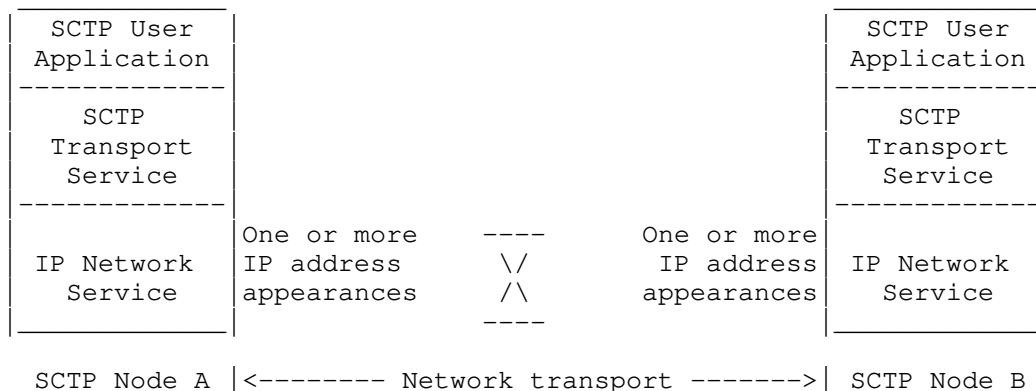


Figure 1: An SCTP Association

In addition to encapsulating SCTP packets in IPv4 or IPv6, it is also possible to encapsulate SCTP packets in UDP as specified in [RFC6951] or encapsulate them in DTLS as specified in [RFC8261].

2.3. Key Terms

Some of the language used to describe SCTP has been introduced in the previous sections. This section provides a consolidated list of the key terms and their definitions.

Active Destination Transport Address: A transport address on a peer endpoint that a transmitting endpoint considers available for receiving user messages.

Association Maximum DATA Chunk Size (AMDCS): The smallest Path Maximum DATA Chunk Size (PMDCS) of all destination addresses.

Bundling Of Chunks: An optional multiplexing operation, whereby more than one chunk can be carried in the same SCTP packet.

Bundling Of User Messages: An optional multiplexing operation, whereby more than one user message can be carried in the same SCTP packet. Each user message occupies its own DATA chunk.

Chunk: A unit of information within an SCTP packet, consisting of a chunk header and chunk-specific content.

Congestion Window (cwnd): An SCTP variable that limits outstanding data, in number of bytes, that a sender can send to a particular destination transport address before receiving an acknowledgement.

Control Chunk: A chunk not being used for transmitting user data, i.e. every chunk which is not a DATA chunk.

Cumulative TSN Ack Point: The Transmission Sequence Number (TSN) of the last DATA chunk acknowledged via the Cumulative TSN Ack field of a SACK chunk.

Flightsize: The number of bytes of outstanding data to a particular destination transport address at any given time.

Idle Destination Address: An address that has not had user messages sent to it within some length of time, normally the 'HB.interval' or greater.

Inactive Destination Transport Address: An address that is considered inactive due to errors and unavailable to transport user messages.

Message (or User Message): Data submitted to SCTP by the Upper Layer Protocol (ULP).

Network Byte Order: Most significant byte first, a.k.a., big endian.

Ordered Message: A user message that is delivered in order with respect to all previous user messages sent within the stream on which the message was sent.

Outstanding Data (or Data Outstanding or Data In Flight): The total size of the DATA chunks associated with outstanding TSNs. A retransmitted DATA chunk is counted once in outstanding data. A DATA chunk that is classified as lost but that has not yet been retransmitted is not in outstanding data.

Outstanding TSN (at an SCTP Endpoint): A TSN (and the associated DATA chunk) that has been sent by the endpoint but for which it has not yet received an acknowledgement.

Out Of The Blue (OOTB) Packet: A correctly formed packet, for which the receiver can not identify the association it belongs to. See Section 8.4.

Path: The route taken by the SCTP packets sent by one SCTP endpoint to a specific destination transport address of its peer SCTP endpoint. Sending to different destination transport addresses does not necessarily guarantee getting separate paths. Within this specification, a path is identified by the destination transport address, since the routing is assumed to be stable. This includes in particular the source address being selected when sending packets to the destination address.

Path Maximum DATA Chunk Size (PMDCS): The maximum size (including the DATA chunk header) of a DATA chunk which fits into an SCTP packet not exceeding the PMTU of a particular destination address.

Path Maximum Transmission Unit (PMTU): The maximum size (including the SCTP common header and all chunks including their paddings) of an SCTP packet which can be sent to a particular destination address without using IP level fragmentation.

Primary Path: The primary path is the destination and source address that will be put into a packet outbound to the peer endpoint by default. The definition includes the source address since an implementation MAY wish to specify both destination and source address to better control the return path taken by reply chunks and on which interface the packet is transmitted when the data sender is multi-homed.

Receiver Window (rwnd): An SCTP variable a data sender uses to store the most recently calculated receiver window of its peer, in number of bytes. This gives the sender an indication of the space available in the receiver's inbound buffer.

SCTP Association: A protocol relationship between SCTP endpoints, composed of the two SCTP endpoints and protocol state information including Verification Tags and the currently active set of Transmission Sequence Numbers (TSNs), etc. An association can be uniquely identified by the transport addresses used by the endpoints in the association. Two SCTP endpoints MUST NOT have more than one SCTP association between them at any given time.

SCTP Endpoint: The logical sender/receiver of SCTP packets. On a

multi-homed host, an SCTP endpoint is represented to its peers as a combination of a set of eligible destination transport addresses to which SCTP packets can be sent and a set of eligible source transport addresses from which SCTP packets can be received. All transport addresses used by an SCTP endpoint MUST use the same port number, but can use multiple IP addresses. A transport address used by an SCTP endpoint MUST NOT be used by another SCTP endpoint. In other words, a transport address is unique to an SCTP endpoint.

SCTP Packet (or Packet): The unit of data delivery across the interface between SCTP and the connectionless packet network (e.g., IP). An SCTP packet includes the common SCTP header, possible SCTP control chunks, and user data encapsulated within SCTP DATA chunks.

SCTP User Application (or SCTP User): The logical higher-layer application entity which uses the services of SCTP, also called the Upper-Layer Protocol (ULP).

Slow-Start Threshold (ssthresh): An SCTP variable. This is the threshold that the endpoint will use to determine whether to perform slow start or congestion avoidance on a particular destination transport address. Ssthresh is in number of bytes.

State Cookie: A container of all information needed to establish an association.

Stream: A unidirectional logical channel established from one to another associated SCTP endpoint, within which all user messages are delivered in sequence except for those submitted to the unordered delivery service.

Note: The relationship between stream numbers in opposite directions is strictly a matter of how the applications use them. It is the responsibility of the SCTP user to create and manage these correlations if they are so desired.

Stream Sequence Number: A 16-bit sequence number used internally by SCTP to ensure sequenced delivery of the user messages within a given stream. One Stream Sequence Number is attached to each ordered user message.

Tie-Tags: Two 32-bit random numbers that together make a 64-bit nonce. These tags are used within a State Cookie and TCB so that a newly restarting association can be linked to the original association within the endpoint that did not restart and yet not reveal the true Verification Tags of an existing association.

Transmission Control Block (TCB): An internal data structure created by an SCTP endpoint for each of its existing SCTP associations to other SCTP endpoints. TCB contains all the status and operational information for the endpoint to maintain and manage the corresponding association.

Transmission Sequence Number (TSN): A 32-bit sequence number used internally by SCTP. One TSN is attached to each chunk containing user data to permit the receiving SCTP endpoint to acknowledge its receipt and detect duplicate deliveries.

Transport Address: A transport address is traditionally defined by a network-layer address, a transport-layer protocol, and a transport-layer port number. In the case of SCTP running over IP, a transport address is defined by the combination of an IP address and an SCTP port number (where SCTP is the transport protocol).

Unordered Message: Unordered messages are "unordered" with respect to any other message; this includes both other unordered messages as well as other ordered messages. An unordered message might be delivered prior to or later than ordered messages sent on the same stream.

User Message: The unit of data delivery across the interface between SCTP and its user.

Verification Tag: A 32-bit unsigned integer that is randomly generated. The Verification Tag provides a key that allows a receiver to verify that the SCTP packet belongs to the current association and is not an old or stale packet from a previous association.

2.4. Abbreviations

MAC Message Authentication Code [RFC2104]
RTO Retransmission Timeout
RTT Round-Trip Time
RTTVAR Round-Trip Time Variation
SCTP Stream Control Transmission Protocol
SRTT Smoothed RTT
TCB Transmission Control Block
TLV Type-Length-Value coding format
TSN Transmission Sequence Number
ULP Upper-Layer Protocol

2.5. Functional View of SCTP

The SCTP transport service can be decomposed into a number of functions. These are depicted in Figure 2 and explained in the remainder of this section.

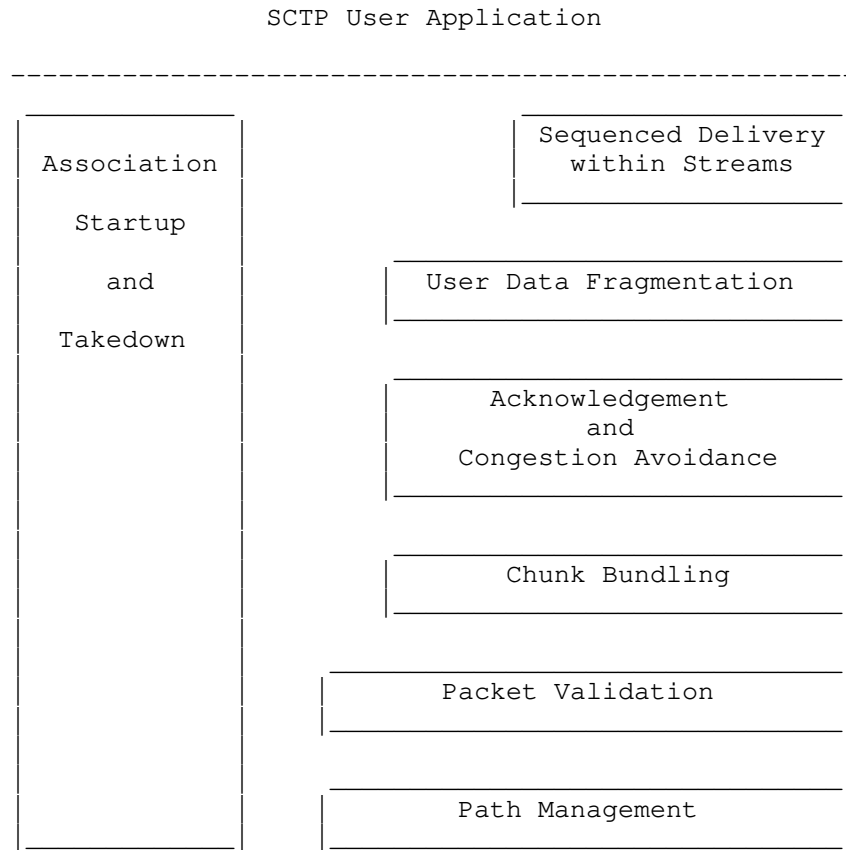


Figure 2: Functional View of the SCTP Transport Service

2.5.1. Association Startup and Takedown

An association is initiated by a request from the SCTP user (see the description of the ASSOCIATE (or SEND) primitive in Section 11).

A cookie mechanism, similar to one described by Karn and Simpson in [RFC2522], is employed during the initialization to provide protection against synchronization attacks. The cookie mechanism uses a four-way handshake, the last two legs of which are allowed to carry user data for fast setup. The startup sequence is described in Section 5 of this document.

SCTP provides for graceful close (i.e., shutdown) of an active association on request from the SCTP user. See the description of the SHUTDOWN primitive in Section 11. SCTP also allows ungraceful close (i.e., abort), either on request from the user (ABORT primitive) or as a result of an error condition detected within the SCTP layer. Section 9 describes both the graceful and the ungraceful close procedures.

SCTP does not support a half-open state (like TCP) wherein one side continues sending data while the other end is closed. When either endpoint performs a shutdown, the association on each peer will stop accepting new data from its user and only deliver data in queue at the time of the graceful close (see Section 9).

2.5.2. Sequenced Delivery within Streams

The term "stream" is used in SCTP to refer to a sequence of user messages that are to be delivered to the upper-layer protocol in order with respect to other messages within the same stream. This is in contrast to its usage in TCP, where it refers to a sequence of bytes (in this document, a byte is assumed to be 8 bits).

The SCTP user can specify at association startup time the number of streams to be supported by the association. This number is negotiated with the remote end (see Section 5.1.1). User messages are associated with stream numbers (SEND, RECEIVE primitives, Section 11). Internally, SCTP assigns a Stream Sequence Number to each message passed to it by the SCTP user. On the receiving side, SCTP ensures that messages are delivered to the SCTP user in sequence within a given stream. However, while one stream might be blocked waiting for the next in-sequence user message, delivery from other streams might proceed.

SCTP provides a mechanism for bypassing the sequenced delivery service. User messages sent using this mechanism are delivered to the SCTP user as soon as they are received.

2.5.3. User Data Fragmentation

When needed, SCTP fragments user messages to ensure that the size of the SCTP packet passed to the lower layer does not exceed the PMTU. Once a user message has been fragmented, this fragmentation cannot be changed anymore. On receipt, fragments are reassembled into complete messages before being passed to the SCTP user.

2.5.4. Acknowledgement and Congestion Avoidance

SCTP assigns a Transmission Sequence Number (TSN) to each user data fragment or unfragmented message. The TSN is independent of any Stream Sequence Number assigned at the stream level. The receiving end acknowledges all TSNs received, even if there are gaps in the sequence. If a user data fragment or unfragmented message needs to be retransmitted, the TSN assigned to it is used. In this way, reliable delivery is kept functionally separate from sequenced stream delivery.

The acknowledgement and congestion avoidance function is responsible for packet retransmission when timely acknowledgement has not been received. Packet retransmission is conditioned by congestion avoidance procedures similar to those used for TCP. See Section 6 and Section 7 for a detailed description of the protocol procedures associated with this function.

2.5.5. Chunk Bundling

As described in Section 3, the SCTP packet as delivered to the lower layer consists of a common header followed by one or more chunks. Each chunk contains either user data or SCTP control information. An SCTP implementation supporting bundling on the sender side might delay the sending of user messages to allow the corresponding DATA chunks to be bundled.

The SCTP user has the option to request that an SCTP implementation does not delay the sending of a user message just for this purpose. However, even if the SCTP user has chosen this option, the SCTP implementation might delay the sending due to other reasons, for example due to congestion control or flow control, and might also bundle multiple DATA chunks, if possible.

2.5.6. Packet Validation

A mandatory Verification Tag field and a 32-bit checksum field (see Appendix A for a description of the CRC32c checksum) are included in the SCTP common header. The Verification Tag value is chosen by each end of the association during association startup. Packets received without the expected Verification Tag value are discarded, as a protection against blind masquerade attacks and against stale SCTP packets from a previous association. The CRC32c checksum is set by the sender of each SCTP packet to provide additional protection against data corruption in the network. The receiver of an SCTP packet with an invalid CRC32c checksum silently discards the packet.

2.5.7. Path Management

The sending SCTP user is able to manipulate the set of transport addresses used as destinations for SCTP packets through the primitives described in Section 11. The SCTP path management function monitors reachability through heartbeats when other packet traffic is inadequate to provide this information and advises the SCTP user when reachability of any transport address of the peer endpoint changes. The path management function chooses the destination transport address for each outgoing SCTP packet based on the SCTP user's instructions and the currently perceived reachability status of the eligible destination set. The path management function is also responsible for reporting the eligible set of local transport addresses to the peer endpoint during association startup, and for reporting the transport addresses returned from the peer endpoint to the SCTP user.

At association startup, a primary path is defined for each SCTP endpoint, and is used for normal sending of SCTP packets.

On the receiving end, the path management is responsible for verifying the existence of a valid SCTP association to which the inbound SCTP packet belongs before passing it for further processing.

Note: Path Management and Packet Validation are done at the same time, so although described separately above, in reality they cannot be performed as separate items.

2.6. Serial Number Arithmetic

It is essential to remember that the actual Transmission Sequence Number space is finite, though very large. This space ranges from 0 to $2^{32} - 1$. Since the space is finite, all arithmetic dealing with Transmission Sequence Numbers MUST be performed modulo 2^{32} . This unsigned arithmetic preserves the relationship of sequence numbers as they cycle from $2^{32} - 1$ to 0 again. There are some subtleties to computer modulo arithmetic, so great care has to be taken in programming the comparison of such values. When referring to TSNs, the symbol " \leq " means "less than or equal" (modulo 2^{32}).

Comparisons and arithmetic on TSNs in this document SHOULD use Serial Number Arithmetic as defined in [RFC1982] where SERIAL_BITS = 32.

An endpoint SHOULD NOT transmit a DATA chunk with a TSN that is more than $2^{31} - 1$ above the beginning TSN of its current send window. Doing so will cause problems in comparing TSNs.

Transmission Sequence Numbers wrap around when they reach $2^{32} - 1$. That is, the next TSN a DATA chunk MUST use after transmitting TSN = $2^{32} - 1$ is TSN = 0.

Any arithmetic done on Stream Sequence Numbers SHOULD use Serial Number Arithmetic as defined in [RFC1982] where SERIAL_BITS = 16. All other arithmetic and comparisons in this document use normal arithmetic.

2.7. Changes from RFC 4960

SCTP was originally defined in [RFC4960], which this document obsoletes, if approved. Readers interested in the details of the various changes that this document incorporates are asked to consult [RFC8540].

In addition to these and further editorial changes, the following changes have been incorporated in this document:

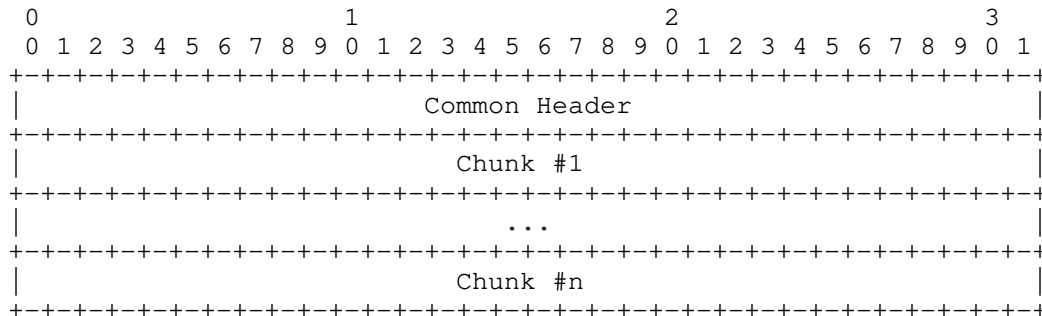
- * Update references.
- * Improve the language related to requirements levels.
- * Allow the ASSOCIATE primitive to take multiple remote addresses; also refer to the Socket API specification.
- * Refer to the PLPMTUD specification for path MTU discovery.

- * Move the description of ICMP handling from an Appendix to the main text.
- * Remove the Appendix describing ECN handling from the document.
- * Describe the packet size handling more precisely by introducing PMTU, PMDCS and AMDCS.
- * Add the definition of control chunk.
- * Improve the description of the handling of INIT and INIT ACK chunks with invalid mandatory parameters.
- * Allow using $L > 1$ for Appropriate Byte Counting (ABC) during slow start.
- * Explicitly describe the reinitialization of the congestion controller on route changes.
- * Improve the terminology to make clear that this specification does not describe a full mesh architecture.
- * Improve the description of sequence number generation (Transmission Sequence Number and Stream Sequence Number).
- * Improve the description of reneging.
- * Don't require the change of the cumulative TSN ACK anymore for increasing the congestion window. This improves the consistency with the handling in congestion avoidance.
- * Improve the description of the State Cookie.
- * Fix the API for retrieving messages in case of association failures.

3. SCTP Packet Format

An SCTP packet is composed of a common header and chunks. A chunk contains either control information or user data.

The SCTP packet format is shown below:

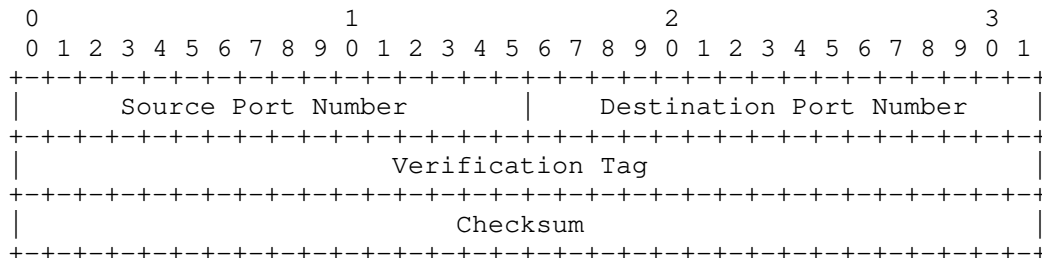


INIT, INIT ACK and SHUTDOWN COMPLETE chunks MUST NOT be bundled with any other chunk into an SCTP packet. All other chunks MAY be bundled to form an SCTP packet that does not exceed the PMTU. See Section 6.10 for more details on chunk bundling.

If a user data message does not fit into one SCTP packet it can be fragmented into multiple chunks using the procedure defined in Section 6.9.

All integer fields in an SCTP packet MUST be transmitted in network byte order, unless otherwise stated.

3.1. SCTP Common Header Field Descriptions



Source Port Number: 16 bits (unsigned integer)

This is the SCTP sender's port number. It can be used by the receiver in combination with the source IP address, the SCTP destination port, and possibly the destination IP address to identify the association to which this packet belongs. The source port number 0 MUST NOT be used.

Destination Port Number: 16 bits (unsigned integer)

This is the SCTP port number to which this packet is destined. The receiving host will use this port number to de-multiplex the SCTP packet to the correct receiving endpoint/application. The destination port number 0 MUST NOT be used.

Verification Tag: 32 bits (unsigned integer)

The receiver of an SCTP packet uses the Verification Tag to validate the sender of this packet. On transmit, the value of the Verification Tag MUST be set to the value of the Initiate Tag received from the peer endpoint during the association initialization, with the following exceptions:

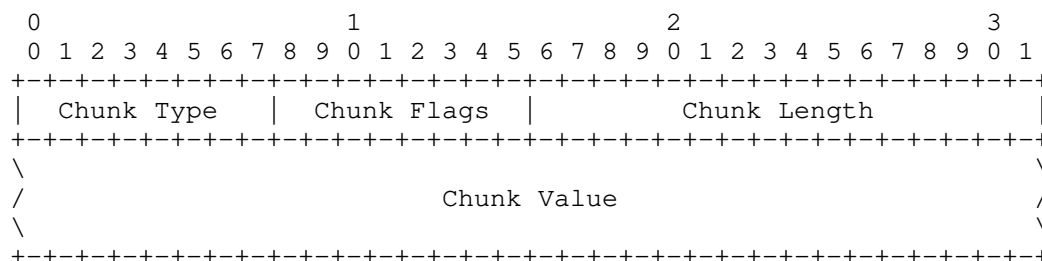
- * A packet containing an INIT chunk MUST have a zero Verification Tag.
- * A packet containing a SHUTDOWN COMPLETE chunk with the T bit set MUST have the Verification Tag copied from the packet with the SHUTDOWN ACK chunk.
- * A packet containing an ABORT chunk MAY have the verification tag copied from the packet that caused the ABORT chunk to be sent. For details see Section 8.4 and Section 8.5.

Checksum: 32 bits (unsigned integer)

This field contains the checksum of the SCTP packet. Its calculation is discussed in Section 6.8. SCTP uses the CRC32c algorithm as described in Appendix A for calculating the checksum.

3.2. Chunk Field Descriptions

The figure below illustrates the field format for the chunks to be transmitted in the SCTP packet. Each chunk is formatted with a Chunk Type field, a chunk-specific Flag field, a Chunk Length field, and a Value field.



Chunk Type: 8 bits (unsigned integer)

This field identifies the type of information contained in the Chunk Value field. It takes a value from 0 to 254. The value of 255 is reserved for future use as an extension field.

The values of Chunk Types are defined as follows:

ID Value	Chunk Type
0	Payload Data (DATA)
1	Initiation (INIT)
2	Initiation Acknowledgement (INIT ACK)
3	Selective Acknowledgement (SACK)
4	Heartbeat Request (HEARTBEAT)
5	Heartbeat Acknowledgement (HEARTBEAT ACK)
6	Abort (ABORT)
7	Shutdown (SHUTDOWN)
8	Shutdown Acknowledgement (SHUTDOWN ACK)
9	Operation Error (ERROR)
10	State Cookie (COOKIE ECHO)
11	Cookie Acknowledgement (COOKIE ACK)
12	Reserved for Explicit Congestion Notification Echo (ECNE)
13	Reserved for Congestion Window Reduced (CWR)
14	Shutdown Complete (SHUTDOWN COMPLETE)
15 to 62	available
63	reserved for IETF-defined Chunk Extensions
64 to 126	available
127	reserved for IETF-defined Chunk Extensions
128 to 190	available

191	reserved for IETF-defined Chunk Extensions
192 to 254	available
255	reserved for IETF-defined Chunk Extensions

Table 1: Chunk Types

Note: The ECNE and CWR chunk types are reserved for future use of Explicit Congestion Notification (ECN).

Chunk Types are encoded such that the highest-order 2 bits specify the action that is taken if the processing endpoint does not recognize the Chunk Type.

00	Stop processing this SCTP packet; discard the unrecognized chunk and all further chunks.
01	Stop processing this SCTP packet, discard the unrecognized chunk and all further chunks, and report the unrecognized chunk in an ERROR chunk using the 'Unrecognized Chunk Type' error cause.
10	Skip this chunk and continue processing.
11	Skip this chunk and continue processing, but report it in an ERROR chunk using the 'Unrecognized Chunk Type' error cause.

Table 2: Processing of Unknown Chunks

Chunk Flags: 8 bits

The usage of these bits depends on the Chunk type as given by the Chunk Type field. Unless otherwise specified, they are set to 0 on transmit and are ignored on receipt.

Chunk Length: 16 bits (unsigned integer)

This value represents the size of the chunk in bytes, including the Chunk Type, Chunk Flags, Chunk Length, and Chunk Value fields. Therefore, if the Chunk Value field is zero-length, the Length field will be set to 4. The Chunk Length field does not count any chunk padding. However, it does include any padding of variable-length parameters other than the last parameter in the chunk.

Note: A robust implementation is expected to accept the chunk whether or not the final padding has been included in the Chunk Length.

Chunk Value: variable length

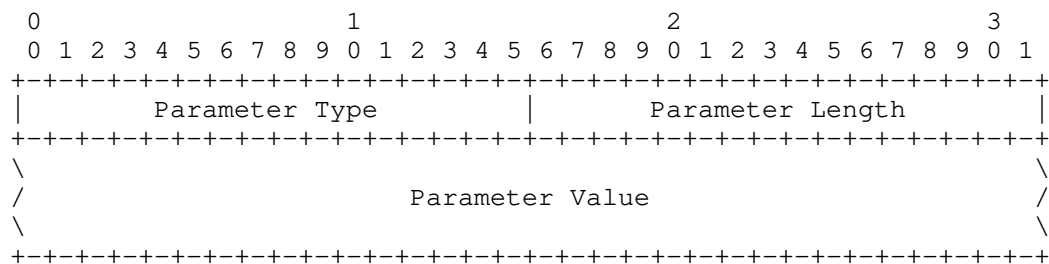
The Chunk Value field contains the actual information to be transferred in the chunk. The usage and format of this field is dependent on the Chunk Type.

The total length of a chunk (including Type, Length, and Value fields) MUST be a multiple of 4 bytes. If the length of the chunk is not a multiple of 4 bytes, the sender MUST pad the chunk with all zero bytes, and this padding is not included in the Chunk Length field. The sender MUST NOT pad with more than 3 bytes. The receiver MUST ignore the padding bytes.

SCTP-defined chunks are described in detail in Section 3.3. The guidelines for IETF-defined chunk extensions can be found in Section 15.1 of this document.

3.2.1. Optional/Variable-Length Parameter Format

Chunk values of SCTP control chunks consist of a chunk-type-specific header of required fields, followed by zero or more parameters. The optional and variable-length parameters contained in a chunk are defined in a Type-Length-Value format as shown below.



Parameter Type: 16 bits (unsigned integer)

The Type field is a 16-bit identifier of the type of parameter. It takes a value of 0 to 65534.

The value of 65535 is reserved for IETF-defined extensions. Values other than those defined in specific SCTP chunk descriptions are reserved for use by IETF.

Parameter Length: 16 bits (unsigned integer)

The Parameter Length field contains the size of the parameter in bytes, including the Parameter Type, Parameter Length, and Parameter Value fields. Thus, a parameter with a zero-length Parameter Value field would have a Parameter Length field of 4. The Parameter Length does not include any padding bytes.

Parameter Value: variable length

The Parameter Value field contains the actual information to be transferred in the parameter.

The total length of a parameter (including Parameter Type, Parameter Length, and Parameter Value fields) MUST be a multiple of 4 bytes. If the length of the parameter is not a multiple of 4 bytes, the sender pads the parameter at the end (i.e., after the Parameter Value field) with all zero bytes. The length of the padding is not included in the Parameter Length field. A sender MUST NOT pad with more than 3 bytes. The receiver MUST ignore the padding bytes.

The Parameter Types are encoded such that the highest-order 2 bits specify the action that is taken if the processing endpoint does not recognize the Parameter Type.

00	Stop processing this parameter; do not process any further parameters within this chunk.
01	Stop processing this parameter, do not process any further parameters within this chunk, and report the unrecognized parameter as described in Section 3.2.2.
10	Skip this parameter and continue processing.
11	Skip this parameter and continue processing but report the unrecognized parameter as described in Section 3.2.2.

Table 3: Processing of Unknown Parameters

Please note that, when an INIT or INIT ACK chunk is received, in all four cases, an INIT ACK or COOKIE ECHO chunk is sent in response, respectively. In the 00 or 01 case, the processing of the parameters after the unknown parameter is canceled, but no processing already done is rolled back.

The actual SCTP parameters are defined in the specific SCTP chunk sections. The rules for IETF-defined parameter extensions are defined in Section 15.3. Parameter types MUST be unique across all chunks. For example, the parameter type '5' is used to represent an IPv4 address (see Section 3.3.2.1). The value '5' then is reserved across all chunks to represent an IPv4 address and MUST NOT be reused with a different meaning in any other chunk.

3.2.2. Reporting of Unrecognized Parameters

If the receiver of an INIT chunk detects unrecognized parameters and has to report them according to Section 3.2.1, it MUST put the "Unrecognized Parameter" parameter(s) in the INIT ACK chunk sent in response to the INIT chunk. Note that if the receiver of the INIT chunk is not going to establish an association (e.g., due to lack of resources), an "Unrecognized Parameter" error cause would not be included with any ABORT chunk being sent to the sender of the INIT chunk.

If the receiver of any other chunk (e.g., INIT ACK) detects unrecognized parameters and has to report them according to Section 3.2.1, it SHOULD bundle the ERROR chunk containing the "Unrecognized Parameters" error cause with the chunk sent in response (e.g., COOKIE ECHO). If the receiver of an INIT ACK chunk cannot bundle the COOKIE ECHO chunk with the ERROR chunk, the ERROR chunk MAY be sent separately but not before the COOKIE ACK chunk has been received.

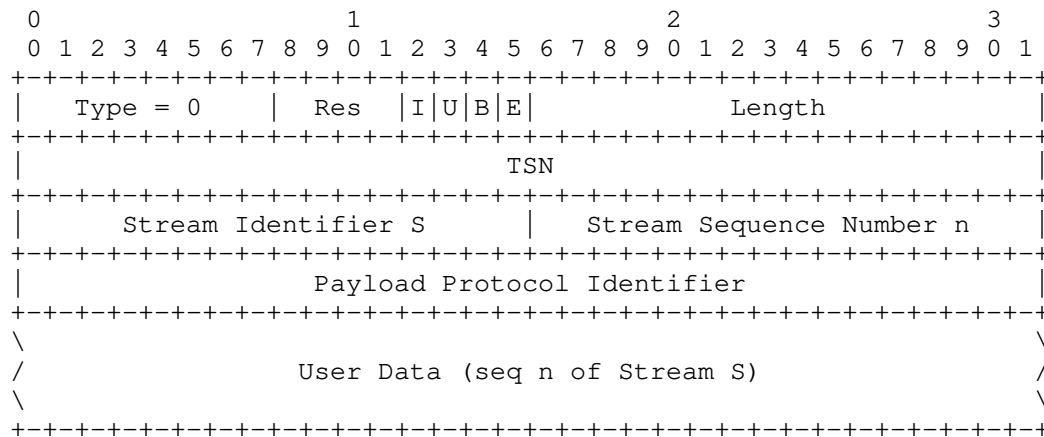
Any time a COOKIE ECHO chunk is sent in a packet, it MUST be the first chunk.

3.3. SCTP Chunk Definitions

This section defines the format of the different SCTP chunk types.

3.3.1. Payload Data (DATA) (0)

The following format MUST be used for the DATA chunk:



Res: 4 bits

All set to 0 on transmit and ignored on receipt.

I bit: 1 bit

The (I)mmEDIATE bit MAY be set by the sender whenever the sender of a DATA chunk can benefit from the corresponding SACK chunk being sent back without delay. See Section 4 of [RFC7053] for a discussion of the benefits.

U bit: 1 bit

The (U)NORDERED bit, if set to 1, indicates that this is an unordered DATA chunk, and there is no Stream Sequence Number assigned to this DATA chunk. Therefore, the receiver MUST ignore the Stream Sequence Number field.

After reassembly (if necessary), unordered DATA chunks MUST be dispatched to the upper layer by the receiver without any attempt to reorder.

If an unordered user message is fragmented, each fragment of the message MUST have its U bit set to 1.

B bit: 1 bit

The (B)EGINNING fragment bit, if set, indicates the first fragment of a user message.

E bit: 1 bit

The (E)NDING fragment bit, if set, indicates the last fragment of a user message.

Length: 16 bits (unsigned integer)

This field indicates the length of the DATA chunk in bytes from the beginning of the type field to the end of the User Data field excluding any padding. A DATA chunk with one byte of user data will have Length set to 17 (indicating 17 bytes).

A DATA chunk with a User Data field of length L will have the Length field set to $(16 + L)$ (indicating $16 + L$ bytes) where L MUST be greater than 0.

TSN: 32 bits (unsigned integer)

This value represents the TSN for this DATA chunk. The valid range of TSN is from 0 to 4294967295 ($2^{32} - 1$). TSN wraps back to 0 after reaching 4294967295.

Stream Identifier S: 16 bits (unsigned integer)

Identifies the stream to which the following user data belongs.

Stream Sequence Number n: 16 bits (unsigned integer)

This value represents the Stream Sequence Number of the following user data within the stream S. Valid range is 0 to 65535.

When a user message is fragmented by SCTP for transport, the same Stream Sequence Number MUST be carried in each of the fragments of the message.

Payload Protocol Identifier: 32 bits (unsigned integer)

This value represents an application (or upper layer) specified protocol identifier. This value is passed to SCTP by its upper layer and sent to its peer. This identifier is not used by SCTP but can be used by certain network entities, as well as by the peer application, to identify the type of information being carried in this DATA chunk. This field MUST be sent even in fragmented DATA chunks (to make sure it is available for agents in the middle of the network). Note that this field is not touched by an SCTP implementation; The upper layer is responsible for the host to network byte order conversion of this field.

The value 0 indicates that no application identifier is specified by the upper layer for this payload data.

User Data: variable length

This is the payload user data. The implementation MUST pad the end of the data to a 4-byte boundary with all-zero bytes. Any padding MUST NOT be included in the Length field. A sender MUST never add more than 3 bytes of padding.

An unfragmented user message **MUST** have both the B and E bits set to 1. Setting both B and E bits to 0 indicates a middle fragment of a multi-fragment user message, as summarized in the following table:

B	E	Description
1	0	First piece of a fragmented user message
0	0	Middle piece of a fragmented user message
0	1	Last piece of a fragmented user message
1	1	Unfragmented message

Table 4: Fragment Description Flags

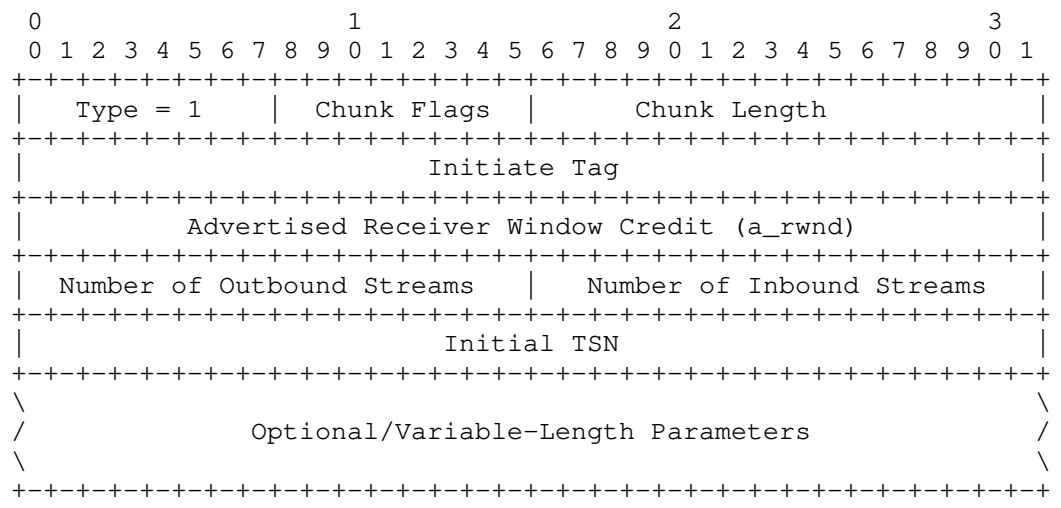
When a user message is fragmented into multiple chunks, the TSNs are used by the receiver to reassemble the message. This means that the TSNs for each fragment of a fragmented user message **MUST** be strictly sequential.

The TSNs of DATA chunks sent **SHOULD** be strictly sequential.

Note: The extension described in [RFC8260] can be used to mitigate the head of line blocking when transferring large user messages.

3.3.2. Initiation (INIT) (1)

This chunk is used to initiate an SCTP association between two endpoints. The format of the INIT chunk is shown below:



The following parameters are specified for the INIT chunk. Unless otherwise noted, each parameter MUST only be included once in the INIT chunk.

Fixed Length Parameter	Status
Initiate Tag	Mandatory
Advertised Receiver Window Credit	Mandatory
Number of Outbound Streams	Mandatory
Number of Inbound Streams	Mandatory
Initial TSN	Mandatory

Table 5: Fixed Length Parameters of INIT Chunks

Variable Length Parameter	Status	Type Value
IPv4 Address (Note 1)	Optional	5
IPv6 Address (Note 1)	Optional	6
Cookie Preservative	Optional	9
Reserved for ECN Capable (Note 2)	Optional	32768 (0x8000)
Host Name Address (Note 3)	Deprecated	11
Supported Address Types (Note 4)	Optional	12

Table 6: Variable Length Parameters of INIT Chunks

Note 1: The INIT chunks can contain multiple addresses that can be IPv4 and/or IPv6 in any combination.

Note 2: The ECN Capable field is reserved for future use of Explicit Congestion Notification.

Note 3: An INIT chunk MUST NOT contain the Host Name Address parameter. The receiver of an INIT chunk containing a Host Name Address parameter MUST send an ABORT chunk and MAY include an "Unresolvable Address" error cause.

Note 4: This parameter, when present, specifies all the address types the sending endpoint can support. The absence of this parameter indicates that the sending endpoint can support any address type.

If an INIT chunk is received with all mandatory parameters that are specified for the INIT chunk, then the receiver SHOULD process the INIT chunk and send back an INIT ACK. The receiver of the INIT chunk MAY bundle an ERROR chunk with the COOKIE ACK chunk later. However, restrictive implementations MAY send back an ABORT chunk in response to the INIT chunk.

The Chunk Flags field in INIT chunks is reserved, and all bits in it SHOULD be set to 0 by the sender and ignored by the receiver.

Initiate Tag: 32 bits (unsigned integer)

The receiver of the INIT chunk (the responding end) records the value of the Initiate Tag parameter. This value MUST be placed into the Verification Tag field of every SCTP packet that the receiver of the INIT chunk transmits within this association.

The Initiate Tag is allowed to have any value except 0. See Section 5.3.1 for more on the selection of the tag value.

If the value of the Initiate Tag in a received INIT chunk is found to be 0, the receiver MUST silently discard the packet.

Advertised Receiver Window Credit (a_rwnd): 32 bits (unsigned integer)

This value represents the dedicated buffer space, in number of bytes, the sender of the INIT chunk has reserved in association with this window.

The Advertised Receiver Window Credit MUST NOT be smaller than 1500.

A receiver of an INIT chunk with the a_rwnd value set to a value smaller than 1500 MUST discard the packet, SHOULD send a packet in response containing an ABORT chunk and using the Initiate Tag as the Verification Tag, and MUST NOT change the state of any existing association.

During the life of the association, this buffer space SHOULD NOT be reduced (i.e., dedicated buffers ought not to be taken away from this association); however, an endpoint MAY change the value of a_rwnd it sends in SACK chunks.

Number of Outbound Streams (OS): 16 bits (unsigned integer)

Defines the number of outbound streams the sender of this INIT chunk wishes to create in this association. The value of 0 MUST NOT be used.

A receiver of an INIT chunk with the OS value set to 0 MUST discard the packet, SHOULD send a packet in response containing an ABORT chunk and using the Initiate Tag as the Verification Tag, and MUST NOT change the state of any existing association.

Number of Inbound Streams (MIS): 16 bits (unsigned integer)

Defines the maximum number of streams the sender of this INIT chunk allows the peer end to create in this association. The value 0 MUST NOT be used.

Note: There is no negotiation of the actual number of streams but instead the two endpoints will use the min(requested, offered). See Section 5.1.1 for details.

A receiver of an INIT chunk with the MIS value set to 0 MUST discard the packet, SHOULD send a packet in response containing an ABORT chunk and using the Initiate Tag as the Verification Tag, and MUST NOT change the state of any existing association.

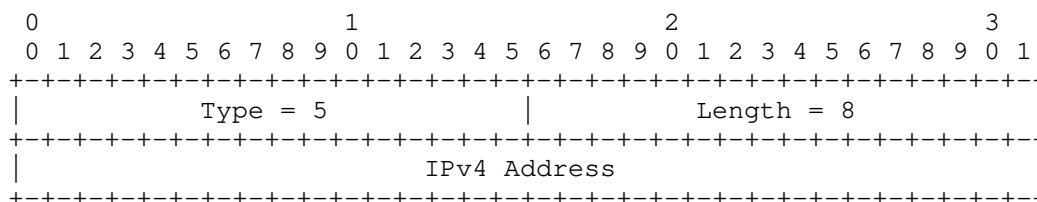
Initial TSN (I-TSN): 32 bits (unsigned integer)

Defines the initial TSN that the sender of the INIT chunk will use. The valid range is from 0 to 4294967295 and the Initial TSN SHOULD be set to a random value in that range. The methods described in [RFC4086] can be used for the Initial TSN randomization.

3.3.2.1. Optional or Variable-Length Parameters in INIT chunks

The following parameters follow the Type-Length-Value format as defined in Section 3.2.1. Any Type-Length-Value fields MUST be placed after the fixed-length fields. (The fixed-length fields are defined in the previous section.)

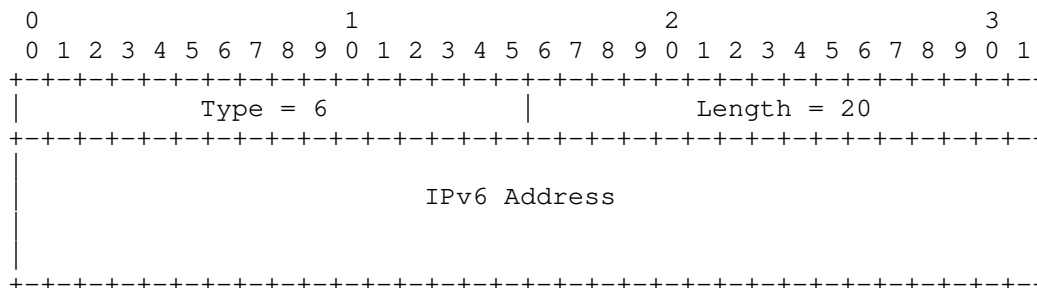
3.3.2.1.1. IPv4 Address (5)



IPv4 Address: 32 bits (unsigned integer)

Contains an IPv4 address of the sending endpoint. It is binary encoded.

3.3.2.1.2. IPv6 Address (6)



IPv6 Address: 128 bits (unsigned integer)

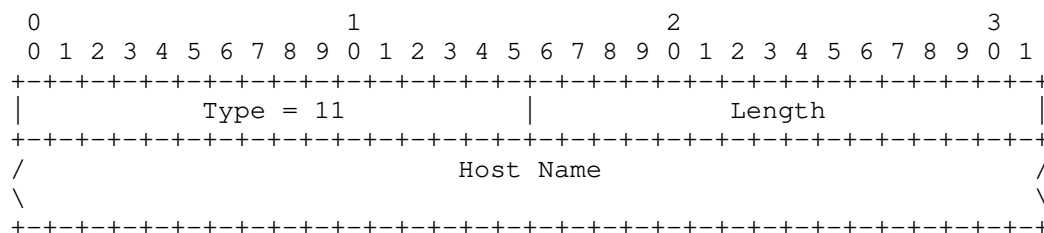
A sender **MUST NOT** use an IPv4-mapped IPv6 address [RFC4291], but **SHOULD** instead use an IPv4 Address parameter for an IPv4 address.

This parameter indicates to the receiver how much increment in milliseconds the sender wishes the receiver to add to its default cookie life-span.

This optional parameter MAY be added to the INIT chunk by the sender when it reattempts establishing an association with a peer to which its previous attempt of establishing the association failed due to a stale cookie operation error. The receiver MAY choose to ignore the suggested cookie life-span increase for its own security reasons.

3.3.2.1.4. Host Name Address (11)

The sender of an INIT chunk or INIT ACK chunk MUST NOT include this parameter. The usage of the Host Name Address parameter is deprecated. The receiver of an INIT chunk or an INIT ACK containing a Host Name Address parameter MUST send an ABORT chunk and MAY include an "Unresolvable Address" error cause.



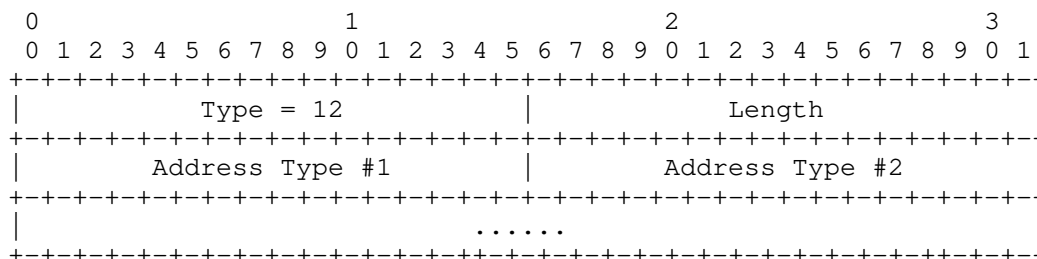
Host Name: variable length

This field contains a host name in "host name syntax" per Section 2.1 of [RFC1123]. The method for resolving the host name is out of scope of SCTP.

At least one null terminator is included in the Host Name string and MUST be included in the length.

3.3.2.1.5. Supported Address Types (12)

The sender of INIT chunk uses this parameter to list all the address types it can support.

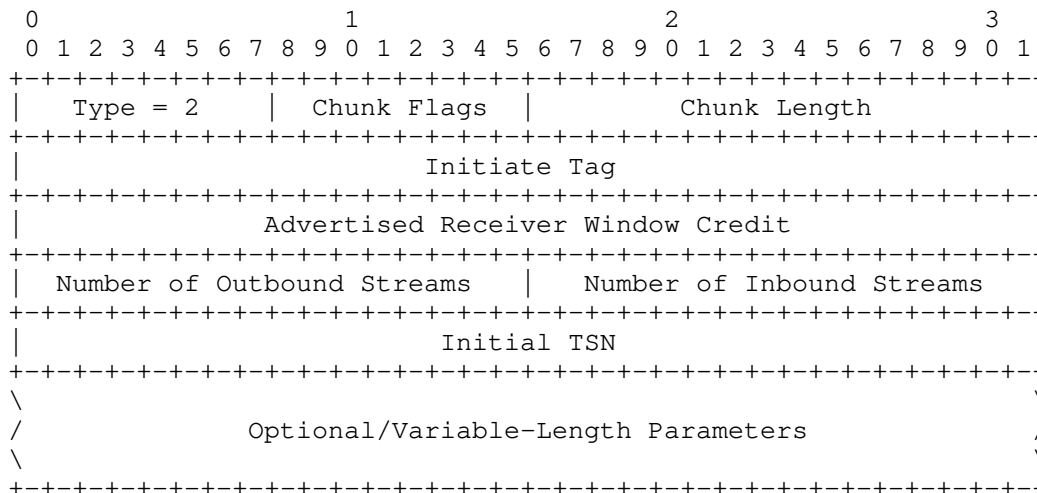


Address Type: 16 bits (unsigned integer)

This is filled with the type value of the corresponding address TLV (e.g., 5 for indicating IPv4, 6 for indicating IPv6). The value indicating the Host Name Address parameter MUST NOT be used when sending this parameter and MUST be ignored when receiving this parameter.

3.3.3. Initiation Acknowledgement (INIT ACK) (2)

The INIT ACK chunk is used to acknowledge the initiation of an SCTP association. The format of the INIT ACK chunk is shown below:



The parameter part of INIT ACK is formatted similarly to the INIT chunk. The following parameters are specified for the INIT ACK chunk:

Fixed Length Parameter	Status
Initiate Tag	Mandatory
Advertised Receiver Window Credit	Mandatory
Number of Outbound Streams	Mandatory
Number of Inbound Streams	Mandatory
Initial TSN	Mandatory

Table 7: Fixed Length Parameters of INIT ACK Chunks

It uses two extra variable parameters: The State Cookie and the Unrecognized Parameter:

Variable Length Parameter	Status	Type Value
State Cookie	Mandatory	7
IPv4 Address (Note 1)	Optional	5
IPv6 Address (Note 1)	Optional	6
Unrecognized Parameter	Optional	8
Reserved for ECN Capable (Note 2)	Optional	32768 (0x8000)
Host Name Address (Note 3)	Deprecated	11

Table 8: Variable Length Parameters of INIT ACK Chunks

Note 1: The INIT ACK chunks can contain any number of IP address parameters that can be IPv4 and/or IPv6 in any combination.

Note 2: The ECN Capable field is reserved for future use of Explicit Congestion Notification.

Note 3: An INIT ACK chunk MUST NOT contain the Host Name Address parameter. The receiver of INIT ACK chunks containing a Host Name Address parameter MUST send an ABORT chunk and MAY include an "Unresolvable Address" error cause.

The Chunk Flags field in INIT ACK chunks is reserved, and all bits in it SHOULD be set to 0 by the sender and ignored by the receiver.

Initiate Tag: 32 bits (unsigned integer)

The receiver of the INIT ACK chunk records the value of the Initiate Tag parameter. This value MUST be placed into the Verification Tag field of every SCTP packet that the receiver of the INIT ACK chunk transmits within this association.

The Initiate Tag MUST NOT take the value 0. See Section 5.3.1 for more on the selection of the Initiate Tag value.

If an endpoint in the COOKIE-WAIT state receives an INIT ACK chunk with the Initiate Tag set to 0, it MUST destroy the TCB and SHOULD send an ABORT chunk with the T bit set. If such an INIT-ACK chunk is received in any state other than CLOSED or COOKIE-WAIT, it SHOULD be discarded silently (see Section 5.2.3).

Advertised Receiver Window Credit (a_rwnd): 32 bits (unsigned integer)

This value represents the dedicated buffer space, in number of bytes, the sender of the INIT ACK chunk has reserved in association with this window.

The Advertised Receiver Window Credit MUST NOT be smaller than 1500.

A receiver of an INIT ACK chunk with the a_rwnd value set to a value smaller than 1500 MUST discard the packet, SHOULD send a packet in response containing an ABORT chunk and using the Initiate Tag as the Verification Tag, and MUST NOT change the state of any existing association.

During the life of the association, this buffer space SHOULD NOT be reduced (i.e., dedicated buffers ought not to be taken away from this association); however, an endpoint MAY change the value of a_rwnd it sends in SACK chunks.

Number of Outbound Streams (OS): 16 bits (unsigned integer)

Defines the number of outbound streams the sender of this INIT ACK chunk wishes to create in this association. The value of 0 MUST NOT be used, and the value MUST NOT be greater than the MIS value sent in the INIT chunk.

If an endpoint in the COOKIE-WAIT state receives an INIT ACK chunk with the OS value set to 0, it MUST destroy the TCB and SHOULD send an ABORT chunk. If such an INIT-ACK chunk is received in any state other than CLOSED or COOKIE-WAIT, it SHOULD be discarded silently (see Section 5.2.3).

Number of Inbound Streams (MIS): 16 bits (unsigned integer)

Defines the maximum number of streams the sender of this INIT ACK chunk allows the peer end to create in this association. The value 0 MUST NOT be used.

Note: There is no negotiation of the actual number of streams but instead the two endpoints will use the min(requested, offered). See Section 5.1.1 for details.

If an endpoint in the COOKIE-WAIT state receives an INIT ACK chunk with the MIS value set to 0, it MUST destroy the TCB and SHOULD send an ABORT chunk. If such an INIT-ACK chunk is received in any state other than CLOSED or COOKIE-WAIT, it SHOULD be discarded silently (see Section 5.2.3).

Initial TSN (I-TSN): 32 bits (unsigned integer)

Defines the initial TSN that the sender of the INIT ACK chunk will use. The valid range is from 0 to 4294967295 and the Initial TSN SHOULD be set to a random value in that range. The methods described in [RFC4086] can be used for the Initial TSN randomization.

Implementation Note: An implementation MUST be prepared to receive an INIT ACK chunk that is quite large (more than 1500 bytes) due to the variable size of the State Cookie and the variable address list. For example if a responder to the INIT chunk has 1000 IPv4 addresses it wishes to send, it would need at least 8,000 bytes to encode this in the INIT ACK chunk.

If an INIT ACK chunk is received with all mandatory parameters that are specified for the INIT ACK chunk, then the receiver SHOULD process the INIT ACK chunk and send back a COOKIE ECHO chunk. The receiver of the INIT ACK chunk MAY bundle an ERROR chunk with the COOKIE ECHO chunk. However, restrictive implementations MAY send back an ABORT chunk in response to the INIT ACK chunk.

In combination with the Source Port carried in the SCTP common header, each IP Address parameter in the INIT ACK chunk indicates to the receiver of the INIT ACK chunk a valid transport address supported by the sender of the INIT ACK chunk for the life time of the association being initiated.

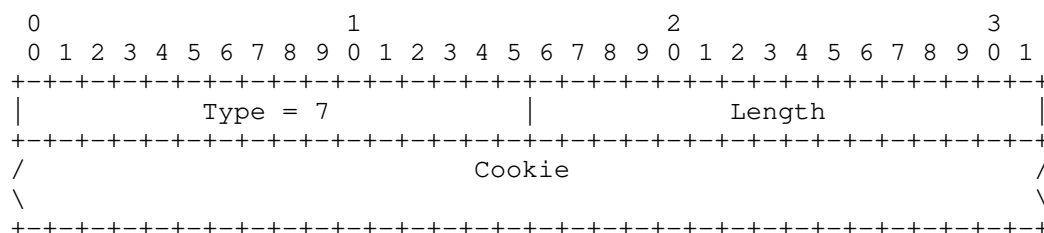
If the INIT ACK chunk contains at least one IP Address parameter, then the source address of the IP datagram containing the INIT ACK chunk and any additional address(es) provided within the INIT ACK chunk MAY be used as destinations by the receiver of the INIT ACK chunk. If the INIT ACK chunk does not contain any IP Address parameters, the receiver of the INIT ACK chunk MUST use the source address associated with the received IP datagram as its sole destination address for the association.

The State Cookie and Unrecognized Parameters use the Type-Length-Value format as defined in Section 3.2.1 and are described below. The other fields are defined the same as their counterparts in the INIT chunk.

3.3.3.1. Optional or Variable-Length Parameters in INIT ACK chunks

The State Cookie and Unrecognized Parameters use the Type-Length-Value format as defined in Section 3.2.1 and are described below. The IPv4 Address Parameter is described in Section 3.3.2.1.1, and the IPv6 Address Parameter is described in Section 3.3.2.1.2. The Host Name Address Parameter is described in Section 3.3.2.1.4 and MUST NOT be included in an INIT ACK chunk. Any Type-Length-Value fields MUST be placed after the fixed-length fields. (The fixed-length fields are defined in the previous section.)

3.3.3.1.1. State Cookie (7)

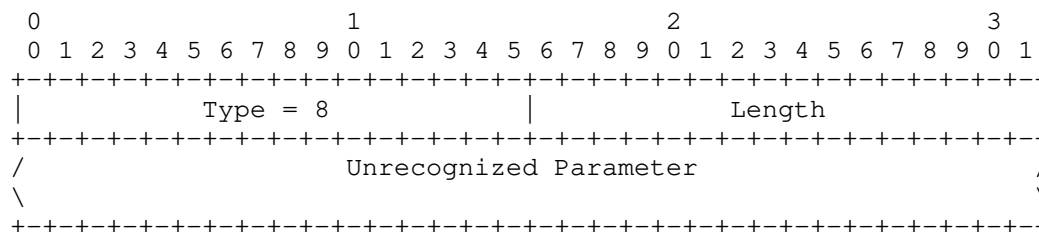


Cookie: variable length

This parameter value MUST contain all the necessary state and parameter information required for the sender of this INIT ACK chunk to create the association, along with a Message Authentication Code (MAC). See Section 5.1.3 for details on State Cookie definition.

3.3.3.1.2. Unrecognized Parameter (8)

This parameter is returned to the originator of the INIT chunk when the INIT chunk contains an unrecognized parameter that has a type that indicates it SHOULD be reported to the sender.



Unrecognized Parameter: variable length

The parameter value field will contain an unrecognized parameter copied from the INIT chunk complete with Parameter Type, Length, and Value fields.

3.3.4. Selective Acknowledgement (SACK) (3)

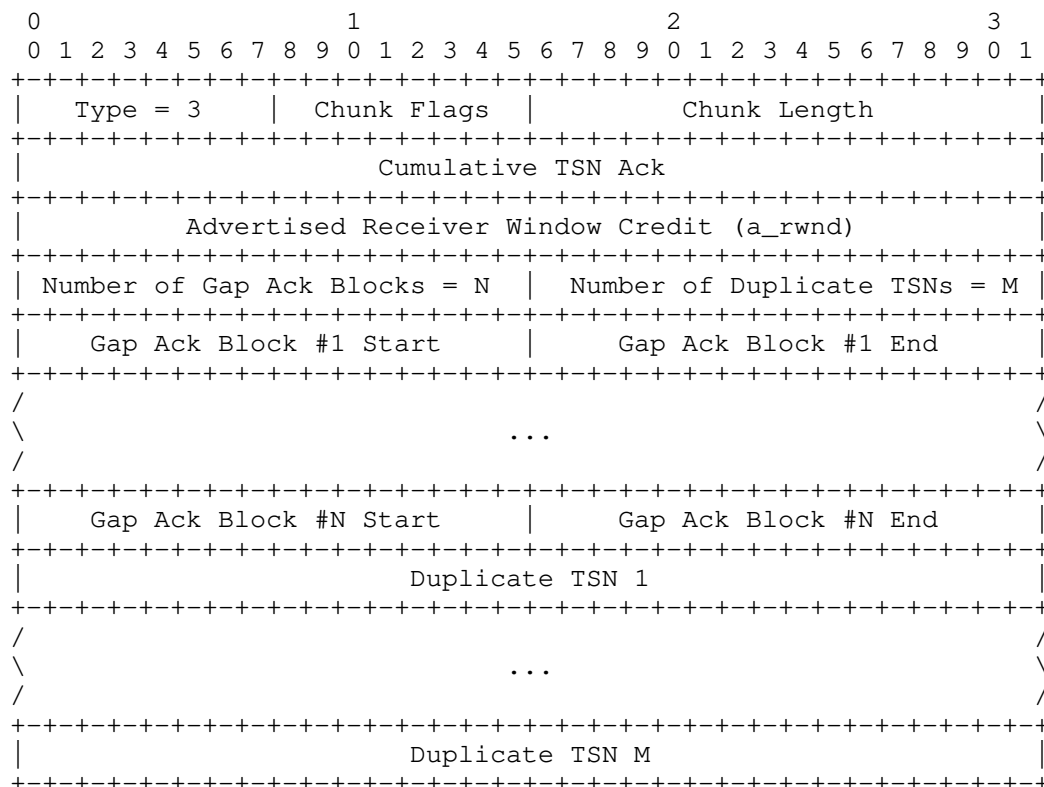
This chunk is sent to the peer endpoint to acknowledge received DATA chunks and to inform the peer endpoint of gaps in the received subsequences of DATA chunks as represented by their TSNs.

The SACK chunk MUST contain the Cumulative TSN Ack, Advertised Receiver Window Credit (*a_rwnd*), Number of Gap Ack Blocks, and Number of Duplicate TSNs fields.

By definition, the value of the Cumulative TSN Ack parameter is the last TSN received before a break in the sequence of received TSNs occurs; the next TSN value following this one has not yet been received at the endpoint sending the SACK chunk. This parameter therefore acknowledges receipt of all TSNs less than or equal to its value.

The handling of *a_rwnd* by the receiver of the SACK chunk is discussed in detail in Section 6.2.1.

The SACK chunk also contains zero or more Gap Ack Blocks. Each Gap Ack Block acknowledges a subsequence of TSNs received following a break in the sequence of received TSNs. The Gap Ack Blocks SHOULD be isolated. This means that the TSN just before each Gap Ack Block and the TSN just after each Gap Ack Block have not been received. By definition, all TSNs acknowledged by Gap Ack Blocks are greater than the value of the Cumulative TSN Ack.



Chunk Flags: 8 bits

All set to 0 on transmit and ignored on receipt.

Cumulative TSN Ack: 32 bits (unsigned integer)

The largest TSN, such that all TSNs smaller than or equal to it have been received and the next one has not been received. In the case where no DATA chunk has been received, this value is set to the peer's Initial TSN minus one.

Advertised Receiver Window Credit (a_rwnd): 32 bits (unsigned integer)

This field indicates the updated receive buffer space in bytes of the sender of this SACK chunk; see Section 6.2.1 for details.

Number of Gap Ack Blocks: 16 bits (unsigned integer)

Indicates the number of Gap Ack Blocks included in this SACK chunk.

Number of Duplicate TSNs: 16 bit

This field contains the number of duplicate TSNs the endpoint has received. Each duplicate TSN is listed following the Gap Ack Block list.

Gap Ack Blocks:

These fields contain the Gap Ack Blocks. They are repeated for each Gap Ack Block up to the number of Gap Ack Blocks defined in the Number of Gap Ack Blocks field. All DATA chunks with TSNs greater than or equal to (Cumulative TSN Ack + Gap Ack Block Start) and less than or equal to (Cumulative TSN Ack + Gap Ack Block End) of each Gap Ack Block are assumed to have been received correctly.

Gap Ack Block Start: 16 bits (unsigned integer)

Indicates the Start offset TSN for this Gap Ack Block. To calculate the actual TSN number the Cumulative TSN Ack is added to this offset number. This calculated TSN identifies the lowest TSN in this Gap Ack Block that has been received.

Gap Ack Block End: 16 bits (unsigned integer)

Indicates the End offset TSN for this Gap Ack Block. To calculate the actual TSN number, the Cumulative TSN Ack is added to this offset number. This calculated TSN identifies the highest TSN in this Gap Ack Block that has been received.

For example, assume that the receiver has the following DATA chunks newly arrived at the time when it decides to send a Selective ACK,

```

-----
| TSN = 17 |
-----
|           | <- still missing
-----
| TSN = 15 |
-----
| TSN = 14 |
-----
|           | <- still missing
-----
| TSN = 12 |
-----
| TSN = 11 |
-----
| TSN = 10 |
-----

```

then the parameter part of the SACK chunk MUST be constructed as follows (assuming the new a_rwnd is set to 4660 by the sender):

Cumulative TSN Ack = 12															
a_rwnd = 4660															
num of block = 2								num of dup = 0							
block #1 start = 2								block #1 end = 3							
block #2 start = 5								block #2 end = 5							

Duplicate TSN: 32 bits (unsigned integer)

Indicates the number of times a TSN was received in duplicate since the last SACK chunk was sent. Every time a receiver gets a duplicate TSN (before sending the SACK chunk), it adds it to the list of duplicates. The duplicate count is reinitialized to zero after sending each SACK chunk.

For example, if a receiver were to get the TSN 19 three times it would list 19 twice in the outbound SACK chunk. After sending the SACK chunk, if it received yet one more TSN 19 it would list 19 as a duplicate once in the next outgoing SACK chunk.

3.3.5. Heartbeat Request (HEARTBEAT) (4)

An endpoint SHOULD send a HEARTBEAT (HB) chunk to its peer endpoint to probe the reachability of a particular destination transport address defined in the present association.

The parameter field contains the Heartbeat Information, which is a variable-length opaque data structure understood only by the sender.

0																1																2																3															
0	1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6	7	8	9	0	1																																
Type = 4																Chunk Flags																Heartbeat Length																															
Heartbeat Information TLV (Variable-Length)																																																															

Chunk Flags: 8 bits

Set to 0 on transmit and ignored on receipt.

Heartbeat Length: 16 bits (unsigned integer)
Set to the size of the chunk in bytes, including the chunk header and the Heartbeat Information field.

Heartbeat Information: variable length
Defined as a variable-length parameter using the format described in Section 3.2.1, i.e.:

Variable Parameters	Status	Type Value
Heartbeat Info	Mandatory	1

Table 9: Variable Length Parameters of HEARTBEAT Chunks

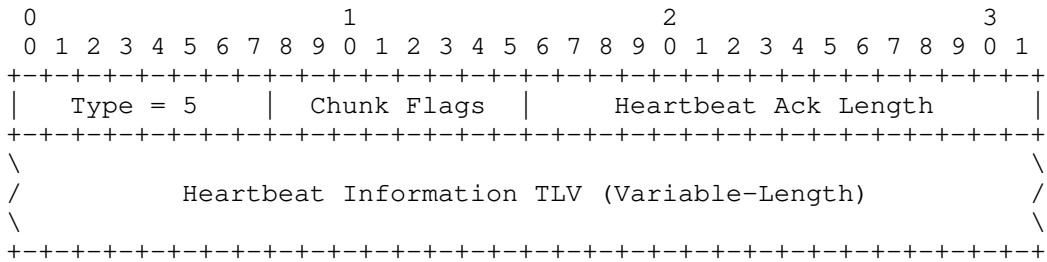
0										1										2										3																													
0	1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6	7	8	9	0	1																												
+										+										+										+																													
										Heartbeat Info Type = 1																				HB Info Length																													
+										+										+										+										+										+									
/										Sender-Specific Heartbeat Info										/										\																													
+										+										+										+										+										+									

The Sender-Specific Heartbeat Info field SHOULD include information about the sender’s current time when this HEARTBEAT chunk is sent and the destination transport address to which this HEARTBEAT chunk is sent (see Section 8.3). This information is simply reflected back by the receiver in the HEARTBEAT ACK chunk (see Section 3.3.6). Note also that the HEARTBEAT chunk is both for reachability checking and for path verification (see Section 5.4). When a HEARTBEAT chunk is being used for path verification purposes, it MUST include a random nonce of length 64-bit or longer ([RFC4086] provides some information on randomness guidelines).

3.3.6. Heartbeat Acknowledgement (HEARTBEAT ACK) (5)

An endpoint MUST send this chunk to its peer endpoint as a response to a HEARTBEAT chunk (see Section 8.3). A packet containing the HEARTBEAT ACK chunk is always sent to the source IP address of the IP datagram containing the HEARTBEAT chunk to which this HEARTBEAT ACK chunk is responding.

The parameter field contains a variable-length opaque data structure.



Chunk Flags: 8 bits
Set to 0 on transmit and ignored on receipt.

Heartbeat Ack Length: 16 bits (unsigned integer)
Set to the size of the chunk in bytes, including the chunk header and the Heartbeat Information field.

Heartbeat Information: variable length
This field MUST contain the Heartbeat Info parameter (as defined in Section 3.3.5) of the Heartbeat Request to which this Heartbeat Acknowledgement is responding.

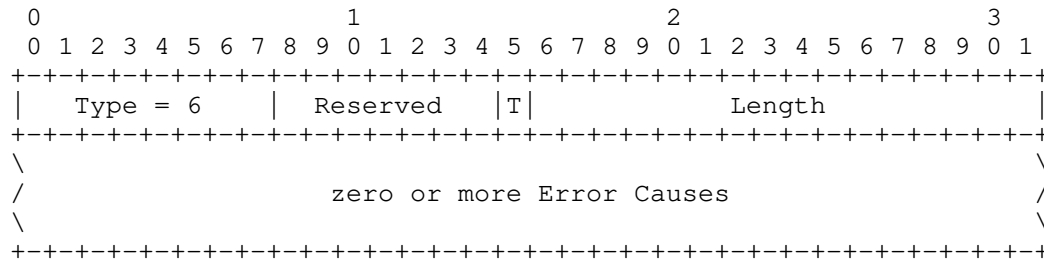
Variable Parameters	Status	Type Value
Heartbeat Info	Mandatory	1

Table 10: Variable Length Parameters of HEARTBEAT ACK Chunks

3.3.7. Abort Association (ABORT) (6)

The ABORT chunk is sent to the peer of an association to close the association. The ABORT chunk MAY contain Cause Parameters to inform the receiver about the reason of the abort. DATA chunks MUST NOT be bundled with ABORT chunks. Control chunks (except for INIT, INIT ACK, and SHUTDOWN COMPLETE) MAY be bundled with an ABORT chunk, but they MUST be placed before the ABORT chunk in the SCTP packet, otherwise they will be ignored by the receiver.

If an endpoint receives an ABORT chunk with a format error or no TCB is found, it MUST silently discard it. Moreover, under any circumstances, an endpoint that receives an ABORT chunk MUST NOT respond to that ABORT chunk by sending an ABORT chunk of its own.



Chunk Flags: 8 bits

Reserved: 7 bits

Set to 0 on transmit and ignored on receipt.

T bit: 1 bit

The T bit is set to 0 if the sender filled in the Verification Tag expected by the peer. If the Verification Tag is reflected, the T bit MUST be set to 1. Reflecting means that the sent Verification Tag is the same as the received one.

Length: 16 bits (unsigned integer)

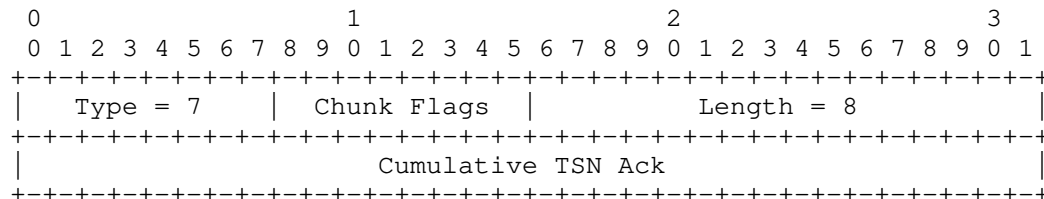
Set to the size of the chunk in bytes, including the chunk header and all the Error Cause fields present.

See Section 3.3.10 for Error Cause definitions.

Note: Special rules apply to this chunk for verification; please see Section 8.5.1 for details.

3.3.8. Shutdown Association (SHUTDOWN) (7)

An endpoint in an association MUST use this chunk to initiate a graceful close of the association with its peer. This chunk has the following format.



Chunk Flags: 8 bits

Set to 0 on transmit and ignored on receipt.

Length: 16 bits (unsigned integer)

Indicates the length of the parameter. Set to 8.

Cumulative TSN Ack: 32 bits (unsigned integer)

The largest TSN, such that all TSNs smaller than or equal to it have been received and the next one has not been received.

Note: Since the SHUTDOWN chunk does not contain Gap Ack Blocks, it cannot be used to acknowledge TSNs received out of order. In a SACK chunk, lack of Gap Ack Blocks that were previously included indicates that the data receiver reneged on the associated DATA chunks.

Since the SHUTDOWN chunk does not contain Gap Ack Blocks, the receiver of the SHUTDOWN chunk MUST NOT interpret the lack of a Gap Ack Block as a renege. (See Section 6.2 for information on reneging.)

The sender of the SHUTDOWN chunk MAY bundle a SACK chunk to indicate any gaps in the received TSNs.

3.3.9. Shutdown Acknowledgement (SHUTDOWN ACK) (8)

This chunk MUST be used to acknowledge the receipt of the SHUTDOWN chunk at the completion of the shutdown process; see Section 9.2 for details.

The SHUTDOWN ACK chunk has no parameters.

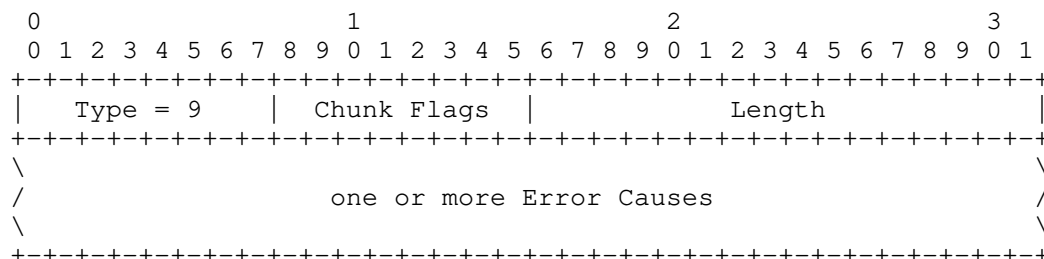
0								1								2								3															
0	1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6	7	8	9
Type = 8								Chunk Flags								Length = 4																							

Chunk Flags: 8 bits

Set to 0 on transmit and ignored on receipt.

3.3.10. Operation Error (ERROR) (9)

An endpoint sends this chunk to its peer endpoint to notify it of certain error conditions. It contains one or more error causes. An Operation Error is not considered fatal in and of itself, but the corresponding error cause MAY be used with an ABORT chunk to report a fatal condition. An ERROR chunk has the following format:



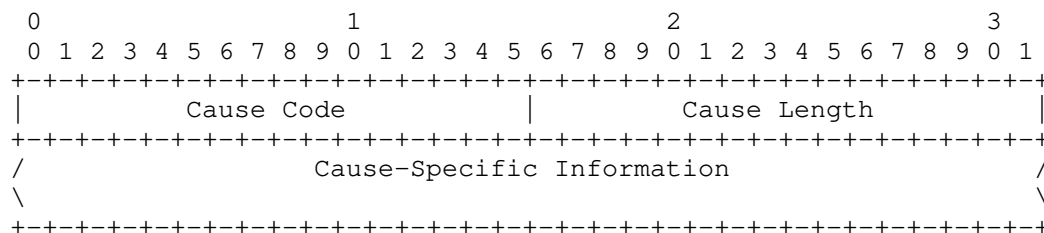
Chunk Flags: 8 bits

Set to 0 on transmit and ignored on receipt.

Length: 16 bits (unsigned integer)

Set to the size of the chunk in bytes, including the chunk header and all the Error Cause fields present.

Error causes are defined as variable-length parameters using the format described in Section 3.2.1, that is:



Cause Code: 16 bits (unsigned integer)

Defines the type of error conditions being reported.

Value	Cause Code
1	Invalid Stream Identifier
2	Missing Mandatory Parameter
3	Stale Cookie Error
4	Out of Resource
5	Unresolvable Address
6	Unrecognized Chunk Type
7	Invalid Mandatory Parameter
8	Unrecognized Parameters
9	No User Data
10	Cookie Received While Shutting Down
11	Restart of an Association with New Addresses
12	User Initiated Abort
13	Protocol Violation

Table 11: Cause Code

Cause Length: 16 bits (unsigned integer)

Set to the size of the parameter in bytes, including the Cause Code, Cause Length, and Cause-Specific Information fields.

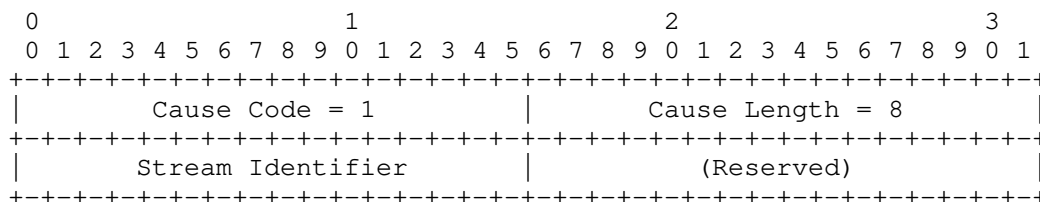
Cause-Specific Information: variable length

This field carries the details of the error condition.

Section 3.3.10.1 – Section 3.3.10.13 define error causes for SCTP. Guidelines for the IETF to define new error cause values are discussed in Section 15.4.

3.3.10.1. Invalid Stream Identifier (1)

Indicates that the endpoint received a DATA chunk sent using a nonexistent stream.



Stream Identifier: 16 bits (unsigned integer)

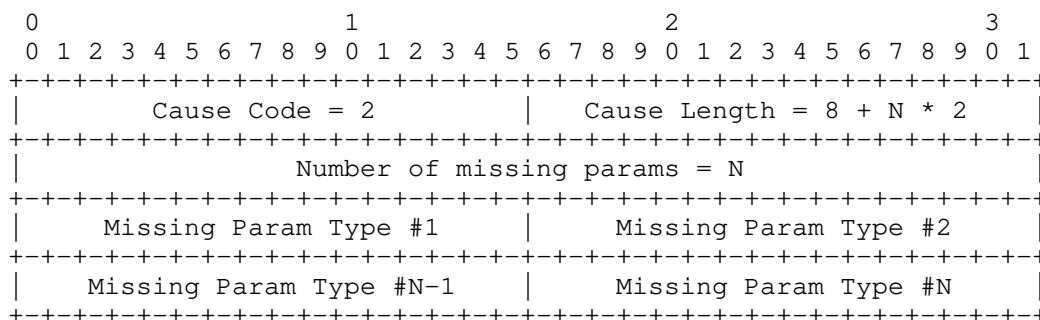
Contains the Stream Identifier of the DATA chunk received in error.

Reserved: 16 bits

This field is reserved. It is set to all 0's on transmit and ignored on receipt.

3.3.10.2. Missing Mandatory Parameter (2)

Indicates that one or more mandatory TLV parameters are missing in a received INIT or INIT ACK chunk.



Number of Missing params: 32 bits (unsigned integer)

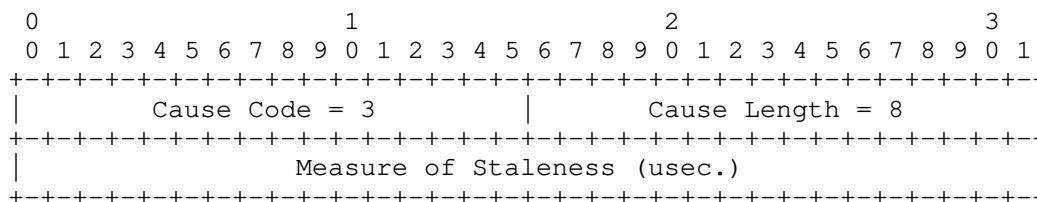
This field contains the number of parameters contained in the Cause-Specific Information field.

Missing Param Type: 16 bits (unsigned integer)

Each field will contain the missing mandatory parameter number.

3.3.10.3. Stale Cookie Error (3)

Indicates the receipt of a valid State Cookie that has expired.



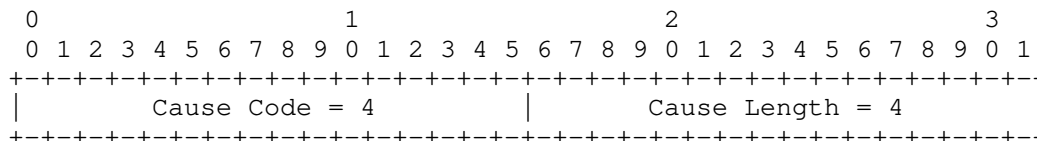
Measure of Staleness: 32 bits (unsigned integer)

This field contains the difference, rounded up in microseconds, between the current time and the time the State Cookie expired.

The sender of this error cause MAY choose to report how long past expiration the State Cookie is by including a non-zero value in the Measure of Staleness field. If the sender does not wish to provide the Measure of Staleness, it SHOULD set this field to the value of zero.

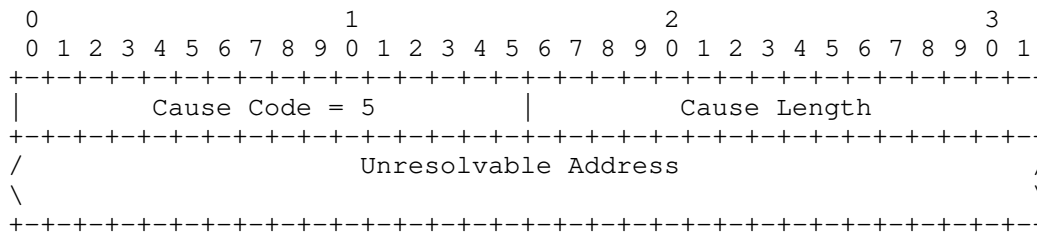
3.3.10.4. Out of Resource (4)

Indicates that the sender is out of resource. This is usually sent in combination with or within an ABORT chunk.



3.3.10.5. Unresolvable Address (5)

Indicates that the sender is not able to resolve the specified address parameter (e.g., type of address is not supported by the sender). This is usually sent in combination with or within an ABORT chunk.

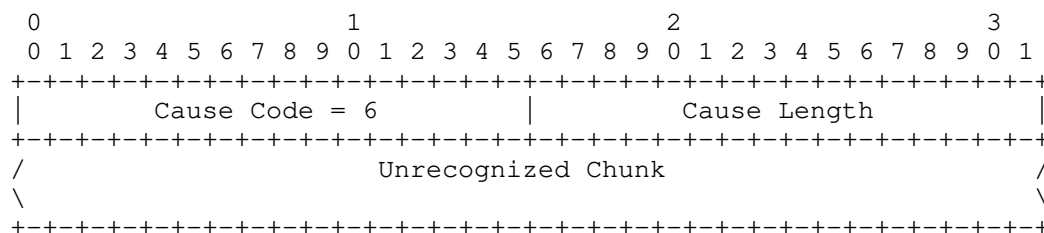


Unresolvable Address: variable length

The Unresolvable Address field contains the complete Type, Length, and Value of the address parameter (or Host Name parameter) that contains the unresolvable address or host name.

3.3.10.6. Unrecognized Chunk Type (6)

This error cause is returned to the originator of the chunk if the receiver does not understand the chunk and the upper bits of the 'Chunk Type' are set to 01 or 11.

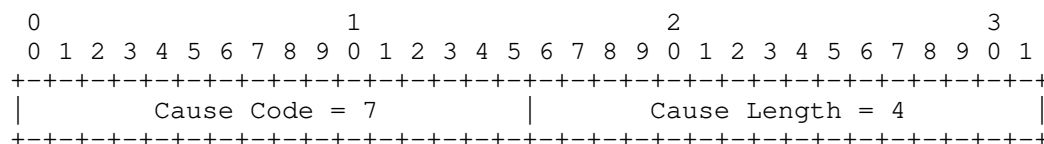


Unrecognized Chunk: variable length

The Unrecognized Chunk field contains the unrecognized chunk from the SCTP packet complete with Chunk Type, Chunk Flags, and Chunk Length.

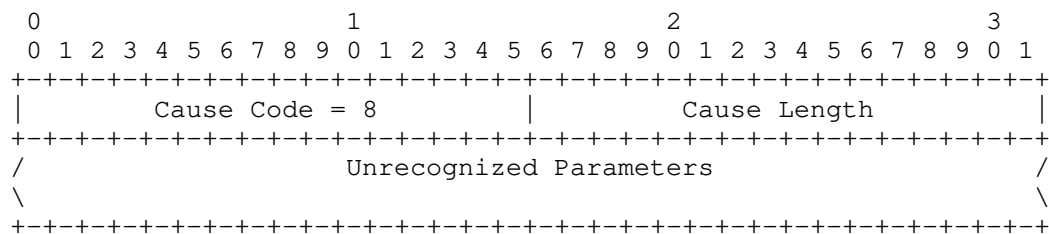
3.3.10.7. Invalid Mandatory Parameter (7)

This error cause is returned to the originator of an INIT or INIT ACK chunk when one of the mandatory parameters is set to an invalid value.



3.3.10.8. Unrecognized Parameters (8)

This error cause is returned to the originator of the INIT ACK chunk if the receiver does not recognize one or more Optional TLV parameters in the INIT ACK chunk.

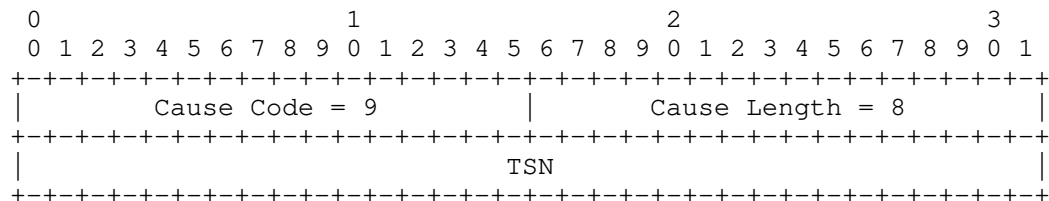


Unrecognized Parameters: variable length

The Unrecognized Parameters field contains the unrecognized parameters copied from the INIT ACK chunk complete with TLV. This error cause is normally contained in an ERROR chunk bundled with the COOKIE ECHO chunk when responding to the INIT ACK chunk, when the sender of the COOKIE ECHO chunk wishes to report unrecognized parameters.

3.3.10.9. No User Data (9)

This error cause is returned to the originator of a DATA chunk if a received DATA chunk has no user data.



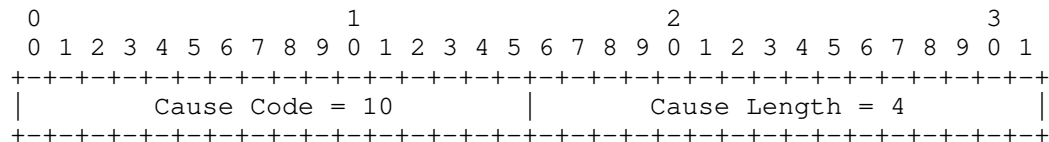
TSN: 32 bits (unsigned integer)

This parameter contains the TSN of the DATA chunk received with no user data field.

This cause code is normally returned in an ABORT chunk (see Section 6.2).

3.3.10.10. Cookie Received While Shutting Down (10)

A COOKIE ECHO chunk was received while the endpoint was in the SHUTDOWN-ACK-SENT state. This error is usually returned in an ERROR chunk bundled with the retransmitted SHUTDOWN ACK chunk.



3.3.10.11. Restart of an Association with New Addresses (11)

An INIT chunk was received on an existing association. But the INIT chunk added addresses to the association that were previously not part of the association. The new addresses are listed in the error cause. This error cause is normally sent as part of an ABORT chunk refusing the INIT chunk (see Section 5.2).

```

      0               1               2               3
    0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-----+-----+-----+-----+-----+-----+-----+-----+
|                                     |                                     |
|          Cause Code = 11          |          Cause Length          |
+-----+-----+-----+-----+-----+-----+-----+-----+
/                                     /
\                                     \
+-----+-----+-----+-----+-----+-----+-----+-----+

```

Note: Each New Address TLV is an exact copy of the TLV that was found in the INIT chunk that was new, including the Parameter Type and the Parameter Length.

3.3.10.12. User-Initiated Abort (12)

This error cause MAY be included in ABORT chunks that are sent because of an upper-layer request. The upper layer can specify an Upper Layer Abort Reason that is transported by SCTP transparently and MAY be delivered to the upper-layer protocol at the peer.

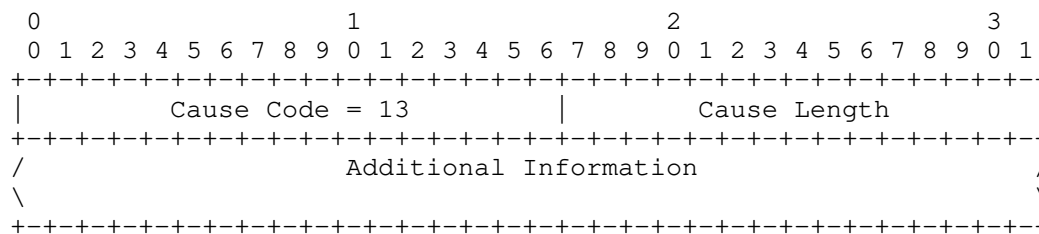
```

      0               1               2               3
    0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-----+-----+-----+-----+-----+-----+-----+-----+
|                                     |                                     |
|          Cause Code = 12          |          Cause Length          |
+-----+-----+-----+-----+-----+-----+-----+-----+
/                                     /
\                                     \
+-----+-----+-----+-----+-----+-----+-----+-----+

```

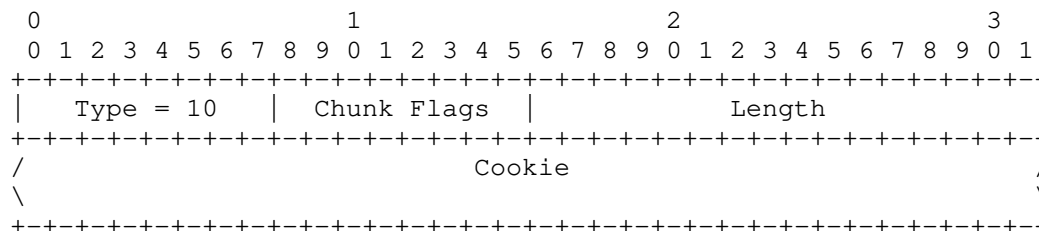
3.3.10.13. Protocol Violation (13)

This error cause MAY be included in ABORT chunks that are sent because an SCTP endpoint detects a protocol violation of the peer that is not covered by the error causes described in Section 3.3.10.1 to Section 3.3.10.12. An implementation MAY provide additional information specifying what kind of protocol violation has been detected.



3.3.11. Cookie Echo (COOKIE ECHO) (10)

This chunk is used only during the initialization of an association. It is sent by the initiator of an association to its peer to complete the initialization process. This chunk **MUST** precede any DATA chunk sent within the association, but **MAY** be bundled with one or more DATA chunks in the same packet.



Chunk Flags: 8 bits

Set to 0 on transmit and ignored on receipt.

Length: 16 bits (unsigned integer)

Set to the size of the chunk in bytes, including the 4 bytes of the chunk header and the size of the cookie.

Cookie: variable size

This field **MUST** contain the exact cookie received in the State Cookie parameter from the previous INIT ACK chunk.

An implementation **SHOULD** make the cookie as small as possible to ensure interoperability.

Note: A Cookie Echo does not contain a State Cookie parameter; instead, the data within the State Cookie's Parameter Value becomes the data within the Cookie Echo's Chunk Value. This allows an implementation to change only the first 2 bytes of the State Cookie parameter to become a COOKIE ECHO chunk.

3.3.12. Cookie Acknowledgement (COOKIE ACK) (11)

This chunk is used only during the initialization of an association. It is used to acknowledge the receipt of a COOKIE ECHO chunk. This chunk MUST precede any DATA or SACK chunk sent within the association, but MAY be bundled with one or more DATA chunks or SACK chunk's in the same SCTP packet.

```

      0               1               2               3
    0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
|  Type = 11  |  Chunk Flags  |                Length = 4          |
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+

```

Chunk Flags: 8 bits

Set to 0 on transmit and ignored on receipt.

3.3.13. Shutdown Complete (SHUTDOWN COMPLETE) (14)

This chunk MUST be used to acknowledge the receipt of the SHUTDOWN ACK chunk at the completion of the shutdown process; see Section 9.2 for details.

The SHUTDOWN COMPLETE chunk has no parameters.

```

      0               1               2               3
    0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
|  Type = 14  |  Reserved  |T|                Length = 4          |
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+

```

Chunk Flags: 8 bits

Reserved: 7 bits

Set to 0 on transmit and ignored on receipt.

T bit: 1 bit

The T bit is set to 0 if the sender filled in the Verification Tag expected by the peer. If the Verification Tag is reflected, the T bit MUST be set to 1. Reflecting means that the sent Verification Tag is the same as the received one.

Note: Special rules apply to this chunk for verification, please see Section 8.5.1 for details.

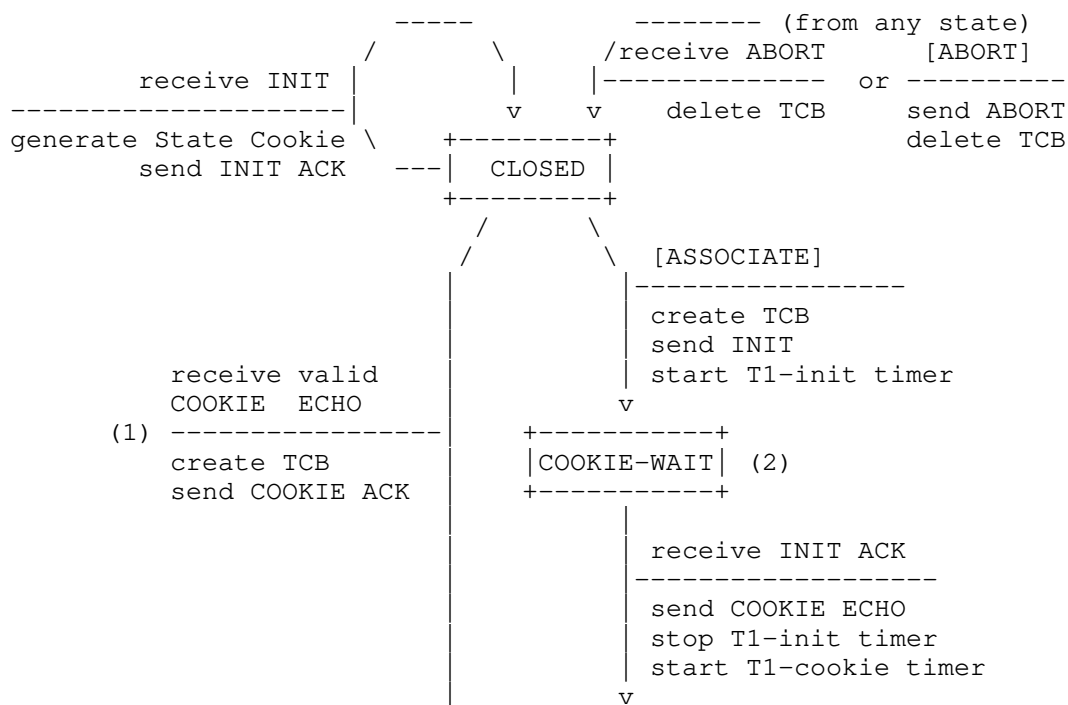
4. SCTP Association State Diagram

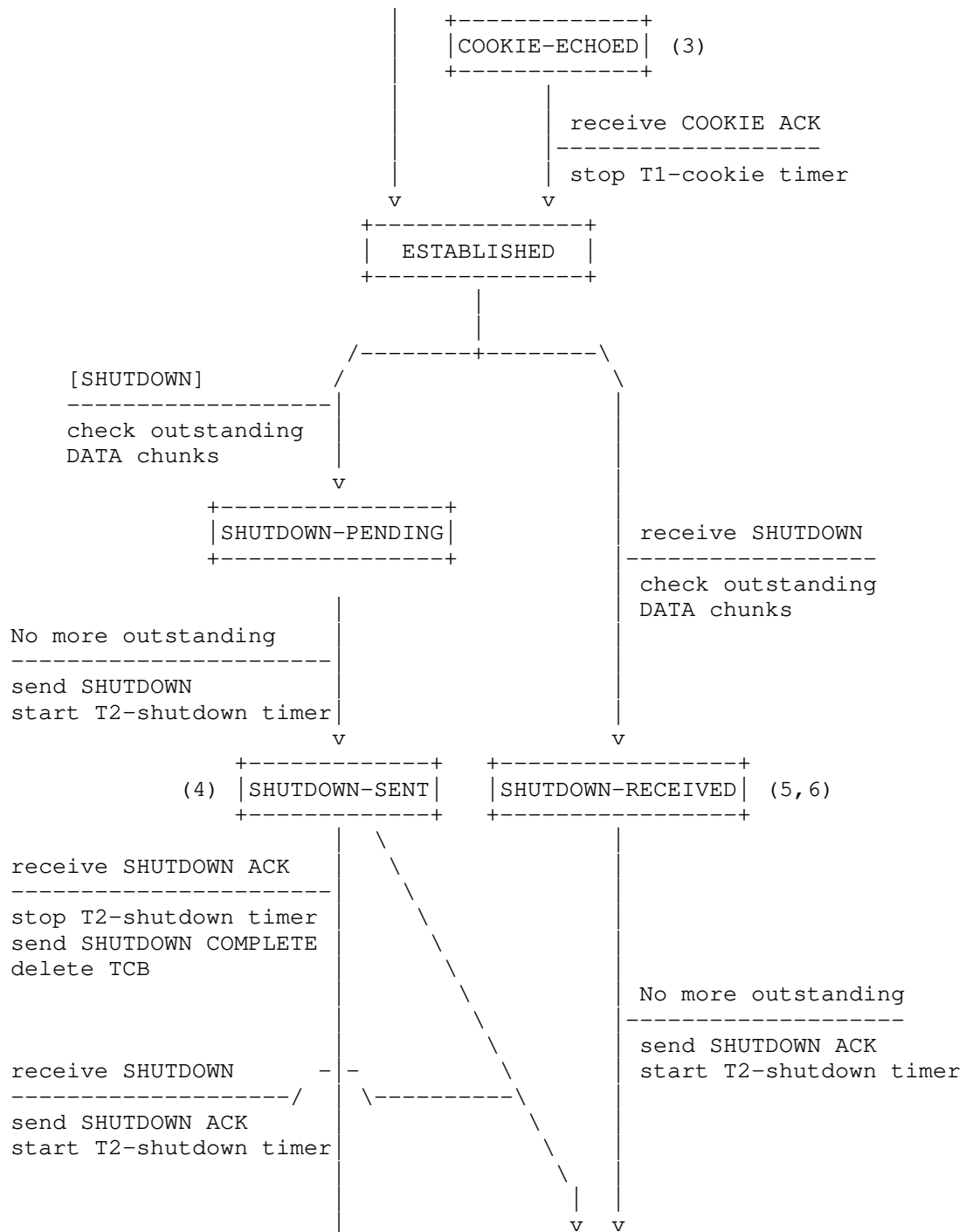
During the life time of an SCTP association, the SCTP endpoint's association progresses from one state to another in response to various events. The events that might potentially advance an association's state include:

- * SCTP user primitive calls, e.g., [ASSOCIATE], [SHUTDOWN], [ABORT],
- * Reception of INIT, COOKIE ECHO, ABORT, SHUTDOWN, etc., control chunks, or
- * Some timeout events.

The state diagram in the figures below illustrates state changes, together with the causing events and resulting actions. Note that some of the error conditions are not shown in the state diagram. Full descriptions of all special cases are found in the text.

Note: Chunk names are given in all capital letters, while parameter names have the first letter capitalized, e.g., COOKIE ECHO chunk type vs. State Cookie parameter. If more than one event/message can occur that causes a state transition, it is labeled (A), (B).





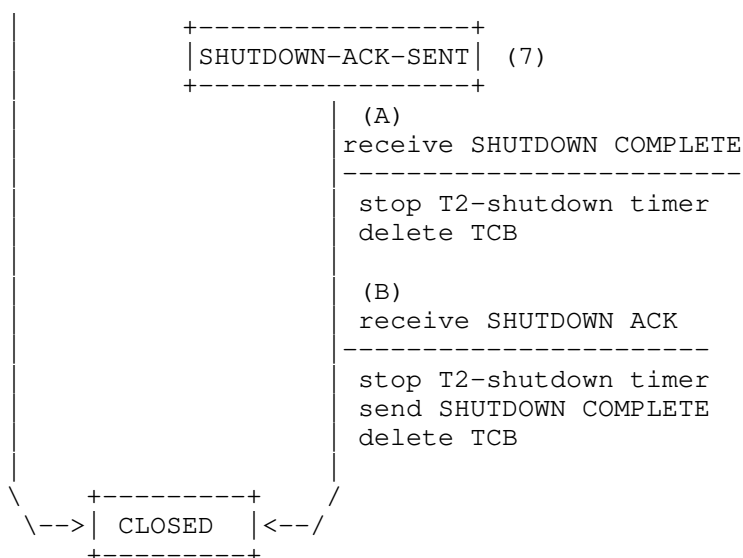


Figure 3: State Transition Diagram of SCTP

The following applies:

- 1) If the State Cookie in the received COOKIE ECHO chunk is invalid (i.e., failed to pass the integrity check), the receiver MUST silently discard the packet. Or, if the received State Cookie is expired (see Section 5.1.5), the receiver MUST send back an ERROR chunk. In either case, the receiver stays in the CLOSED state.
- 2) If the T1-init timer expires, the endpoint MUST retransmit the INIT chunk and restart the T1-init timer without changing state. This MUST be repeated up to 'Max.Init.Retransmits' times. After that, the endpoint MUST abort the initialization process and report the error to the SCTP user.
- 3) If the T1-cookie timer expires, the endpoint MUST retransmit COOKIE ECHO chunk and restart the T1-cookie timer without changing state. This MUST be repeated up to 'Max.Init.Retransmits' times. After that, the endpoint MUST abort the initialization process and report the error to the SCTP user.
- 4) In the SHUTDOWN-SENT state, the endpoint MUST acknowledge any received DATA chunks without delay.
- 5) In the SHUTDOWN-RECEIVED state, the endpoint MUST NOT accept any new send requests from its SCTP user.

- 6) In the SHUTDOWN-RECEIVED state, the endpoint MUST transmit or retransmit data and leave this state when all data in queue is transmitted.
- 7) In the SHUTDOWN-ACK-SENT state, the endpoint MUST NOT accept any new send requests from its SCTP user.

The CLOSED state is used to indicate that an association is not created (i.e., does not exist).

5. Association Initialization

Before the first data transmission can take place from one SCTP endpoint ("A") to another SCTP endpoint ("Z"), the two endpoints MUST complete an initialization process in order to set up an SCTP association between them.

The SCTP user at an endpoint can use the ASSOCIATE primitive to initialize an SCTP association to another SCTP endpoint.

Implementation Note: From an SCTP user's point of view, an association might be implicitly opened, without an ASSOCIATE primitive (see Section 11.1.2) being invoked, by the initiating endpoint's sending of the first user data to the destination endpoint. The initiating SCTP will assume default values for all mandatory and optional parameters for the INIT/INIT ACK chunk.

Once the association is established, unidirectional streams are open for data transfer on both ends (see Section 5.1.1).

5.1. Normal Establishment of an Association

The initialization process consists of the following steps (assuming that SCTP endpoint "A" tries to set up an association with SCTP endpoint "Z" and "Z" accepts the new association):

- A) "A" first builds a TCB and sends an INIT chunk to "Z". In the INIT chunk, "A" MUST provide its Verification Tag (Tag_A) in the Initiate Tag field. Tag_A SHOULD be a random number in the range of 1 to 4294967295 (see Section 5.3.1 for Tag value selection). After sending the INIT chunk, "A" starts the T1-init timer and enters the COOKIE-WAIT state.

- B) "Z" responds immediately with an INIT ACK chunk. The destination IP address of the INIT ACK chunk MUST be set to the source IP address of the INIT chunk to which this INIT ACK chunk is responding. In the response, besides filling in other parameters, "Z" MUST set the Verification Tag field to Tag_A, and also provide its own Verification Tag (Tag_Z) in the Initiate Tag field.

Moreover, "Z" MUST generate and send along with the INIT ACK chunk a State Cookie. See Section 5.1.3 for State Cookie generation.

After sending an INIT ACK chunk with the State Cookie parameter, "Z" MUST NOT allocate any resources or keep any states for the new association. Otherwise, "Z" will be vulnerable to resource attacks.

- C) Upon reception of the INIT ACK chunk from "Z", "A" stops the Tl-init timer and leaves the COOKIE-WAIT state. "A" then sends the State Cookie received in the INIT ACK chunk in a COOKIE ECHO chunk, starts the Tl-cookie timer, and enters the COOKIE-ECHOED state.

The COOKIE ECHO chunk MAY be bundled with any pending outbound DATA chunks, but it MUST be the first chunk in the packet and until the COOKIE ACK chunk is returned the sender MUST NOT send any other packets to the peer.

- D) Upon reception of the COOKIE ECHO chunk, endpoint "Z" replies with a COOKIE ACK chunk after building a TCB and moving to the ESTABLISHED state. A COOKIE ACK chunk MAY be bundled with any pending DATA chunks (and/or SACK chunks), but the COOKIE ACK chunk MUST be the first chunk in the packet.

Implementation Note: An implementation can choose to send the Communication Up notification to the SCTP user upon reception of a valid COOKIE ECHO chunk.

- E) Upon reception of the COOKIE ACK chunk, endpoint "A" moves from the COOKIE-ECHOED state to the ESTABLISHED state, stopping the Tl-cookie timer. It can also notify its ULP about the successful establishment of the association with a Communication Up notification (see Section 11).

An INIT or INIT ACK chunk MUST NOT be bundled with any other chunk. They MUST be the only chunks present in the SCTP packets that carry them.

An endpoint MUST send the INIT ACK chunk to the IP address from which it received the INIT chunk.

Tl-init timer and Tl-cookie timer SHOULD follow the same rules given in Section 6.3. If the application provided multiple IP addresses of the peer, there SHOULD be a Tl-init and Tl-cookie timer for each address of the peer. Retransmissions of INIT chunks and COOKIE ECHO chunks SHOULD use all addresses of the peer similar to retransmissions of DATA chunks.

If an endpoint receives an INIT, INIT ACK, or COOKIE ECHO chunk but decides not to establish the new association due to missing mandatory parameters in the received INIT or INIT ACK chunk, invalid parameter values, or lack of local resources, it SHOULD respond with an ABORT chunk. It SHOULD also specify the cause of abort, such as the type of the missing mandatory parameters, etc., by including the error cause parameters with the ABORT chunk. The Verification Tag field in the common header of the outbound SCTP packet containing the ABORT chunk MUST be set to the Initiate Tag value of the received INIT or INIT ACK chunk this ABORT chunk is responding to.

Note that a COOKIE ECHO chunk that does not pass the integrity check is not considered an 'invalid mandatory parameter' and requires special handling; see Section 5.1.5.

After the reception of the first DATA chunk in an association the endpoint MUST immediately respond with a SACK chunk to acknowledge the DATA chunk. Subsequent acknowledgements SHOULD be done as described in Section 6.2.

When the TCB is created, each endpoint MUST set its internal Cumulative TSN Ack Point to the value of its transmitted Initial TSN minus one.

Implementation Note: The IP addresses and SCTP port are generally used as the key to find the TCB within an SCTP instance.

5.1.1. Handle Stream Parameters

In the INIT and INIT ACK chunks, the sender of the chunk MUST indicate the number of outbound streams (OSs) it wishes to have in the association, as well as the maximum inbound streams (MISs) it will accept from the other endpoint.

After receiving the stream configuration information from the other side, each endpoint MUST perform the following check: If the peer's MIS is less than the endpoint's OS, meaning that the peer is incapable of supporting all the outbound streams the endpoint wants

to configure, the endpoint MUST use MIS outbound streams and MAY report any shortage to the upper layer. The upper layer can then choose to abort the association if the resource shortage is unacceptable.

After the association is initialized, the valid outbound stream identifier range for either endpoint MUST be 0 to min(local OS, remote MIS) - 1.

5.1.2. Handle Address Parameters

During the association initialization, an endpoint uses the following rules to discover and collect the destination transport address(es) of its peer.

- A) If there are no address parameters present in the received INIT or INIT ACK chunk, the endpoint MUST take the source IP address from which the chunk arrives and record it, in combination with the SCTP source port number, as the only destination transport address for this peer.
- B) If there is a Host Name Address parameter present in the received INIT or INIT ACK chunk, the endpoint MUST immediately send an ABORT chunk and MAY include an "Unresolvable Address" error cause to its peer. The ABORT chunk SHOULD be sent to the source IP address from which the last peer packet was received.
- C) If there are only IPv4/IPv6 addresses present in the received INIT or INIT ACK chunk, the receiver MUST derive and record all the transport addresses from the received chunk AND the source IP address that sent the INIT or INIT ACK chunk. The transport addresses are derived by the combination of SCTP source port (from the common header) and the IP Address parameter(s) carried in the INIT or INIT ACK chunk and the source IP address of the IP datagram. The receiver SHOULD use only these transport addresses as destination transport addresses when sending subsequent packets to its peer.
- D) An INIT or INIT ACK chunk MUST be treated as belonging to an already established association (or one in the process of being established) if the use of any of the valid address parameters contained within the chunk would identify an existing TCB.

Implementation Note: In some cases (e.g., when the implementation does not control the source IP address that is used for transmitting), an endpoint might need to include in its INIT or INIT ACK chunk all possible IP addresses from which packets to the peer could be transmitted.

After all transport addresses are derived from the INIT or INIT ACK chunk using the above rules, the endpoint selects one of the transport addresses as the initial primary path.

The packet containing the INIT ACK chunk MUST be sent to the source address of the packet containing the INIT chunk.

The sender of INIT chunks MAY include a 'Supported Address Types' parameter in the INIT chunk to indicate what types of addresses are acceptable.

Implementation Note: In the case that the receiver of an INIT ACK chunk fails to resolve the address parameter due to an unsupported type, it can abort the initiation process and then attempt a reinitiation by using a 'Supported Address Types' parameter in the new INIT chunk to indicate what types of address it prefers.

If an SCTP endpoint that only supports either IPv4 or IPv6 receives IPv4 and IPv6 addresses in an INIT or INIT ACK chunk from its peer, it MUST use all the addresses belonging to the supported address family. The other addresses MAY be ignored. The endpoint SHOULD NOT respond with any kind of error indication.

If an SCTP endpoint lists in the 'Supported Address Types' parameter either IPv4 or IPv6, but uses the other family for sending the packet containing the INIT chunk, or if it also lists addresses of the other family in the INIT chunk, then the address family that is not listed in the 'Supported Address Types' parameter SHOULD also be considered as supported by the receiver of the INIT chunk. The receiver of the INIT chunk SHOULD NOT respond with any kind of error indication.

5.1.3. Generating State Cookie

When sending an INIT ACK chunk as a response to an INIT chunk, the sender of INIT ACK chunk creates a State Cookie and sends it in the State Cookie parameter of the INIT ACK chunk. Inside this State Cookie, the sender MUST include a MAC (see [RFC2104] for an example) to provide integrity protection on the State Cookie. The State Cookie SHOULD also contain a timestamp on when the State Cookie is created, and the lifespan of the State Cookie, along with all the information necessary for it to establish the association including the port numbers and the verification tags.

The method used to generate the MAC is strictly a private matter for the receiver of the INIT chunk. The use of a MAC is mandatory to prevent denial-of-service attacks. MAC algorithms can have different performance depending on the platform. Choosing a high performance MAC algorithm increases the resistance against cookie flooding

attacks. A MAC with acceptable security properties SHOULD be used. The secret key SHOULD be random ([RFC4086] provides some information on randomness guidelines). The secret keys need to have an appropriate size. The secret key SHOULD be changed reasonably frequently (e.g., hourly), and the timestamp in the State Cookie MAY be used to determine which key is used to verify the MAC.

If the State Cookie is not encrypted, it MUST NOT contain information which is not being envisioned to be shared.

An implementation SHOULD make the cookie as small as possible to ensure interoperability.

5.1.4. State Cookie Processing

When an endpoint (in the COOKIE-WAIT state) receives an INIT ACK chunk with a State Cookie parameter, it MUST immediately send a COOKIE ECHO chunk to its peer with the received State Cookie. The sender MAY also add any pending DATA chunks to the packet after the COOKIE ECHO chunk.

The endpoint MUST also start the Tl-cookie timer after sending the COOKIE ECHO chunk. If the timer expires, the endpoint MUST retransmit the COOKIE ECHO chunk and restart the Tl-cookie timer. This is repeated until either a COOKIE ACK chunk is received or 'Max.Init.Retransmits' (see Section 16) is reached causing the peer endpoint to be marked unreachable (and thus the association enters the CLOSED state).

5.1.5. State Cookie Authentication

When an endpoint receives a COOKIE ECHO chunk from another endpoint with which it has no association, it takes the following actions:

- 1) Compute a MAC using the information carried in the State Cookie and the secret key. The timestamp in the State Cookie MAY be used to determine which secret key to use. If secrets are kept only for a limited amount of time and the secret key to use is not available anymore, the packet containing the COOKIE ECHO chunk MUST be silently discarded. [RFC2104] can be used as a guideline for generating the MAC,
- 2) Authenticate the State Cookie as one that it previously generated by comparing the computed MAC against the one carried in the State Cookie. If this comparison fails, the SCTP packet, including the COOKIE ECHO chunk and any DATA chunks, MUST be silently discarded,

- 3) Compare the port numbers and the Verification Tag contained within the COOKIE ECHO chunk to the actual port numbers and the Verification Tag within the SCTP common header of the received packet. If these values do not match, the packet MUST be silently discarded.
- 4) Compare the creation timestamp in the State Cookie to the current local time. If the elapsed time is longer than the lifespan carried in the State Cookie, then the packet, including the COOKIE ECHO chunk and any attached DATA chunks, SHOULD be discarded, and the endpoint MUST transmit an ERROR chunk with a "Stale Cookie" error cause to the peer endpoint.
- 5) If the State Cookie is valid, create an association to the sender of the COOKIE ECHO chunk with the information in the State Cookie carried in the COOKIE ECHO chunk and enter the ESTABLISHED state.
- 6) Send a COOKIE ACK chunk to the peer acknowledging receipt of the COOKIE ECHO chunk. The COOKIE ACK chunk MAY be bundled with an outbound DATA chunk or SACK chunk; however, the COOKIE ACK chunk MUST be the first chunk in the SCTP packet.
- 7) Immediately acknowledge any DATA chunk bundled with the COOKIE ECHO chunk with a SACK chunk (subsequent DATA chunk acknowledgement SHOULD follow the rules defined in Section 6.2). As mentioned in step 6, if the SACK chunk is bundled with the COOKIE ACK chunk, the COOKIE ACK chunk MUST appear first in the SCTP packet.

If a COOKIE ECHO chunk is received from an endpoint with which the receiver of the COOKIE ECHO chunk has an existing association, the procedures in Section 5.2 SHOULD be followed.

5.1.6. An Example of Normal Association Establishment

In the following example, "A" initiates the association and then sends a user message to "Z", then "Z" sends two user messages to "A" later (assuming no bundling or fragmentation occurs):

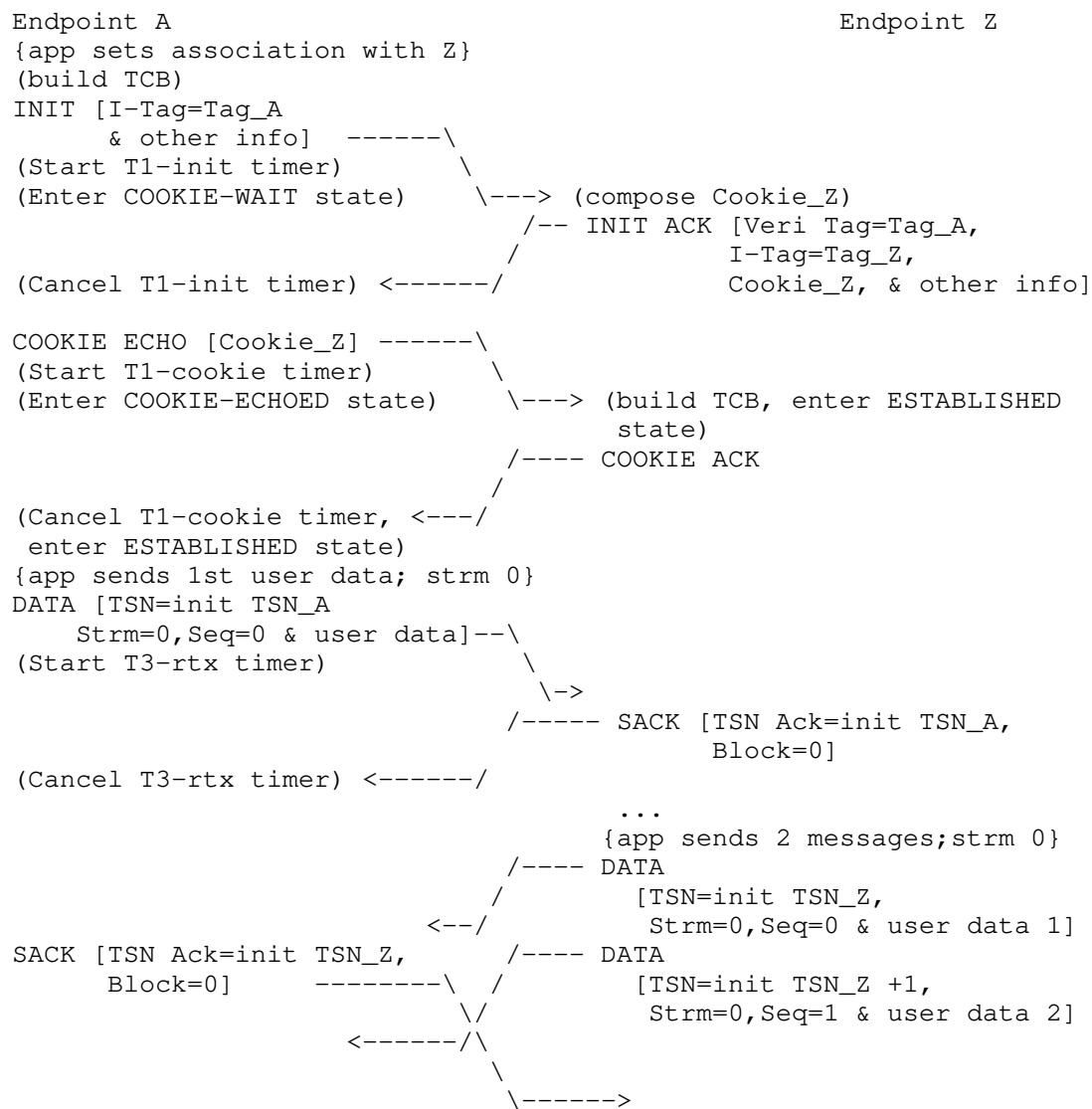


Figure 4: A Setup Example

If the Tl-init timer expires at "A" after the INIT or COOKIE ECHO chunks are sent, the same INIT or COOKIE ECHO chunk with the same Initiate Tag (i.e., Tag_A) or State Cookie is retransmitted and the timer restarted. This is repeated 'Max.Init.Retransmits' times before "A" considers "Z" unreachable and reports the failure to its upper layer (and thus the association enters the CLOSED state).

When retransmitting the INIT chunk, the endpoint MUST follow the rules defined in Section 6.3 to determine the proper timer value.

5.2. Handle Duplicate or Unexpected INIT, INIT ACK, COOKIE ECHO, and COOKIE ACK Chunks

During the life time of an association (in one of the possible states), an endpoint can receive from its peer endpoint one of the setup chunks (INIT, INIT ACK, COOKIE ECHO, and COOKIE ACK). The receiver treats such a setup chunk as a duplicate and process it as described in this section.

Note: An endpoint will not receive the chunk unless the chunk was sent to an SCTP transport address and is from an SCTP transport address associated with this endpoint. Therefore, the endpoint processes such a chunk as part of its current association.

The following scenarios can cause duplicated or unexpected chunks:

- A) The peer has crashed without being detected, restarted itself, and sent a new INIT chunk trying to restore the association,
- B) Both sides are trying to initialize the association at about the same time,
- C) The chunk is from a stale packet that was used to establish the present association or a past association that is no longer in existence,
- D) The chunk is a false packet generated by an attacker, or
- E) The peer never received the COOKIE ACK chunk and is retransmitting its COOKIE ECHO chunk.

The rules in the following sections are applied in order to identify and correctly handle these cases.

5.2.1. INIT Chunk Received in COOKIE-WAIT or COOKIE-ECHOED State (Item B)

This usually indicates an initialization collision, i.e., each endpoint is attempting, at about the same time, to establish an association with the other endpoint.

Upon receipt of an INIT chunk in the COOKIE-WAIT state, an endpoint MUST respond with an INIT ACK chunk using the same parameters it sent in its original INIT chunk (including its Initiate Tag, unchanged). When responding, the following rules MUST be applied:

- 1) The packet containing the INIT ACK chunk MUST only be sent to an address passed by the upper layer in the request to initialize the association.
- 2) The packet containing the INIT ACK chunk MUST only be sent to an address reported in the incoming INIT chunk.
- 3) The packet containing the INIT ACK chunk SHOULD be sent to the source address of the received packet containing the INIT chunk.

Upon receipt of an INIT chunk in the COOKIE-ECHOED state, an endpoint MUST respond with an INIT ACK chunk using the same parameters it sent in its original INIT chunk (including its Initiate Tag, unchanged), provided that no NEW address has been added to the forming association. If the INIT chunk indicates that a new address has been added to the association, then the entire INIT chunk MUST be discarded, and the state of the existing association SHOULD NOT be changed. An ABORT chunk SHOULD be sent in response that MAY include the error 'Restart of an association with new addresses'. The error SHOULD list the addresses that were added to the restarting association.

When responding in either state (COOKIE-WAIT or COOKIE-ECHOED) with an INIT ACK chunk, the original parameters are combined with those from the newly received INIT chunk. The endpoint MUST also generate a State Cookie with the INIT ACK chunk. The endpoint uses the parameters sent in its INIT chunk to calculate the State Cookie.

After that, the endpoint MUST NOT change its state, the Tl-init timer MUST be left running, and the corresponding TCB MUST NOT be destroyed. The normal procedures for handling State Cookies when a TCB exists will resolve the duplicate INIT chunks to a single association.

For an endpoint that is in the COOKIE-ECHOED state, it MUST populate its Tie-Tags within both the association TCB and inside the State Cookie (see Section 5.2.2 for a description of the Tie-Tags).

5.2.2. Unexpected INIT Chunk in States Other than CLOSED, COOKIE-ECHOED, COOKIE-WAIT, and SHUTDOWN-ACK-SENT

Unless otherwise stated, upon receipt of an unexpected INIT chunk for this association, the endpoint MUST generate an INIT ACK chunk with a State Cookie. Before responding, the endpoint MUST check to see if the unexpected INIT chunk adds new addresses to the association. If new addresses are added to the association, the endpoint MUST respond with an ABORT chunk, copying the 'Initiate Tag' of the unexpected INIT chunk into the 'Verification Tag' of the outbound packet

carrying the ABORT chunk. In the ABORT chunk, the error cause MAY be set to 'restart of an association with new addresses'. The error SHOULD list the addresses that were added to the restarting association. If no new addresses are added, when responding to the INIT chunk in the outbound INIT ACK chunk, the endpoint MUST copy its current Tie-Tags to a reserved place within the State Cookie and the association's TCB. We refer to these locations inside the cookie as the Peer's-Tie-Tag and the Local-Tie-Tag. We will refer to the copy within an association's TCB as the Local Tag and Peer's Tag. The outbound SCTP packet containing this INIT ACK chunk MUST carry a Verification Tag value equal to the Initiate Tag found in the unexpected INIT chunk. And the INIT ACK chunk MUST contain a new Initiate Tag (randomly generated; see Section 5.3.1). Other parameters for the endpoint SHOULD be copied from the existing parameters of the association (e.g., number of outbound streams) into the INIT ACK chunk and cookie.

After sending the INIT ACK or ABORT chunk, the endpoint MUST take no further actions; i.e., the existing association, including its current state, and the corresponding TCB MUST NOT be changed.

Only when a TCB exists and the association is not in a COOKIE-WAIT or SHUTDOWN-ACK-SENT state are the Tie-Tags populated with a random value other than 0. For a normal association INIT chunk (i.e., the endpoint is in the CLOSED state), the Tie-Tags MUST be set to 0 (indicating that no previous TCB existed).

5.2.3. Unexpected INIT ACK Chunk

If an INIT ACK chunk is received by an endpoint in any state other than the COOKIE-WAIT or CLOSED state, the endpoint SHOULD discard the INIT ACK chunk. An unexpected INIT ACK chunk usually indicates the processing of an old or duplicated INIT chunk.

5.2.4. Handle a COOKIE ECHO Chunk when a TCB Exists

When a COOKIE ECHO chunk is received by an endpoint in any state for an existing association (i.e., not in the CLOSED state) the following rules are applied:

- 1) Compute a MAC as described in step 1 of Section 5.1.5,
- 2) Authenticate the State Cookie as described in step 2 of Section 5.1.5 (this is case C or D above).
- 3) Compare the timestamp in the State Cookie to the current time. If the State Cookie is older than the lifespan carried in the State Cookie and the Verification Tags contained in the State

Cookie do not match the current association's Verification Tags, the packet, including the COOKIE ECHO chunk and any DATA chunks, SHOULD be discarded. The endpoint also MUST transmit an ERROR chunk with a "Stale Cookie" error cause to the peer endpoint (this is case C or D in Section 5.2).

If both Verification Tags in the State Cookie match the Verification Tags of the current association, consider the State Cookie valid (this is case E in Section 5.2) even if the lifespan is exceeded.

- 4) If the State Cookie proves to be valid, unpack the TCB into a temporary TCB.
- 5) Refer to Table 12 to determine the correct action to be taken.

Local Tag	Peer's Tag	Local-Tie-Tag	Peer's-Tie-Tag	Action
X	X	M	M	(A)
M	X	A	A	(B)
M	0	A	A	(B)
X	M	0	0	(C)
M	M	A	A	(D)

Table 12: Handling of a COOKIE ECHO Chunk when a TCB Exists

Legend:

- X - Tag does not match the existing TCB.
- M - Tag matches the existing TCB.
- 0 - Tag unknown (Peer's Tag not known yet / No tie-tag in cookie).
- A - All cases, i.e., M, X, or 0.

For any case not shown in Table 12, the cookie SHOULD be silently discarded.

Action

- A) In this case, the peer might have restarted. When the endpoint recognizes this potential 'restart', the existing session is treated the same as if it received an ABORT chunk followed by a new COOKIE ECHO chunk with the following exceptions:

- * Any SCTP DATA chunks MAY be retained (this is an implementation-specific option).
- * A notification of RESTART SHOULD be sent to the ULP instead of a "COMMUNICATION LOST" notification.

All the congestion control parameters (e.g., cwnd, ssthresh) related to this peer MUST be reset to their initial values (see Section 6.2.1).

After this, the endpoint enters the ESTABLISHED state.

If the endpoint is in the SHUTDOWN-ACK-SENT state and recognizes that the peer has restarted (Action A), it MUST NOT set up a new association but instead resend the SHUTDOWN ACK chunk and send an ERROR chunk with a "Cookie Received While Shutting Down" error cause to its peer.

- B) In this case, both sides might be attempting to start an association at about the same time, but the peer endpoint sent its INIT chunk after responding to the local endpoint's INIT chunk. Thus, it might have picked a new Verification Tag, not being aware of the previous tag it had sent this endpoint. The endpoint SHOULD stay in or enter the ESTABLISHED state, but it MUST update its peer's Verification Tag from the State Cookie, stop any T1-init or T1-cookie timers that might be running, and send a COOKIE ACK chunk.
- C) In this case, the local endpoint's cookie has arrived late. Before it arrived, the local endpoint sent an INIT chunk and received an INIT ACK chunk and finally sent a COOKIE ECHO chunk with the peer's same tag but a new tag of its own. The cookie SHOULD be silently discarded. The endpoint SHOULD NOT change states and SHOULD leave any timers running.
- D) When both local and remote tags match, the endpoint SHOULD enter the ESTABLISHED state, if it is in the COOKIE-ECHOED state. It SHOULD stop any T1-cookie timer that is running and send a COOKIE ACK chunk.

Note: The "peer's Verification Tag" is the tag received in the Initiate Tag field of the INIT or INIT ACK chunk.

5.2.4.1. An Example of a Association Restart

In the following example, "A" initiates the association after a restart has occurred. Endpoint "Z" had no knowledge of the restart until the exchange (i.e., Heartbeats had not yet detected the failure of "A") (assuming no bundling or fragmentation occurs):

```

Endpoint A                                     Endpoint Z
<----- Association is established----->
Tag=Tag_A                                     Tag=Tag_Z
<----->
{A crashes and restarts}
{app sets up a association with Z}
(build TCB)
INIT [I-Tag=Tag_A'
      & other info] -----\
(Start T1-init timer)       \
(Enter COOKIE-WAIT state)   \----> (find an existing TCB,
                                   populate TieTags if needed,
                                   compose Cookie_Z with Tie-Tags
                                   and other info)
                                   /--- INIT ACK [Veri Tag=Tag_A',
                                   I-Tag=Tag_Z',
                                   Cookie_Z]
                                   (leave original TCB in place)
(Cancel T1-init timer) <-----/
COOKIE ECHO [Veri=Tag_Z',
             Cookie_Z]-----\
(Start T1-init timer)       \
(Enter COOKIE-ECHOED state)  \----> (Find existing association,
                                   Tie-Tags in Cookie_Z match
                                   Tie-Tags in TCB,
                                   Tags do not match, i.e.,
                                   case X X M M above,
                                   Announce Restart to ULP
                                   and reset association).
                                   /----- COOKIE ACK
(Cancel T1-init timer, <-----/
  Enter ESTABLISHED state)
{app sends 1st user data; strm 0}
DATA [TSN=initial TSN_A
      Strm=0,Seq=0 & user data]--\
(Start T3-rtx timer)           \
                                   \->
                                   /--- SACK [TSN Ack=init TSN_A,Block=0]
(Cancel T3-rtx timer) <-----/

```

Figure 5: A Restart Example

5.2.5. Handle Duplicate COOKIE ACK Chunk

At any state other than COOKIE-ECHOED, an endpoint SHOULD silently discard a received COOKIE ACK chunk.

5.2.6. Handle Stale Cookie Error

Receipt of an ERROR chunk with a "Stale Cookie" error cause indicates one of a number of possible events:

- A) The association failed to completely setup before the State Cookie issued by the sender was processed.
- B) An old State Cookie was processed after setup completed.
- C) An old State Cookie is received from someone that the receiver is not interested in having an association with and the ABORT chunk was lost.

When processing an ERROR chunk with a "Stale Cookie" error cause an endpoint SHOULD first examine if an association is in the process of being set up, i.e., the association is in the COOKIE-ECHOED state. In all cases, if the association is not in the COOKIE-ECHOED state, the ERROR chunk SHOULD be silently discarded.

If the association is in the COOKIE-ECHOED state, the endpoint MAY elect one of the following three alternatives.

- 1) Send a new INIT chunk to the endpoint to generate a new State Cookie and reattempt the setup procedure.
- 2) Discard the TCB and report to the upper layer the inability to set up the association.
- 3) Send a new INIT chunk to the endpoint, adding a Cookie Preservative parameter requesting an extension to the life time of the State Cookie. When calculating the time extension, an implementation SHOULD use the RTT information measured based on the previous COOKIE ECHO / ERROR chunk exchange, and SHOULD add no more than 1 second beyond the measured RTT, due to long State Cookie life times making the endpoint more subject to a replay attack.

5.3. Other Initialization Issues

5.3.1. Selection of Tag Value

Initiate Tag values SHOULD be selected from the range of 1 to $2^{32} - 1$. It is very important that the Initiate Tag value be randomized to help protect against "man in the middle" and "sequence number" attacks. The methods described in [RFC4086] can be used for the Initiate Tag randomization. Careful selection of Initiate Tags is also necessary to prevent old duplicate packets from previous associations being mistakenly processed as belonging to the current association.

Moreover, the Verification Tag value used by either endpoint in a given association MUST NOT change during the life time of an association. A new Verification Tag value MUST be used each time the endpoint tears down and then reestablishes an association to the same peer.

5.4. Path Verification

During association establishment, the two peers exchange a list of addresses. In the predominant case, these lists accurately represent the addresses owned by each peer. However, a misbehaving peer might supply addresses that it does not own. To prevent this, the following rules are applied to all addresses of the new association:

- 1) Any addresses passed to the sender of the INIT chunk by its upper layer in the request to initialize an association are automatically considered to be CONFIRMED.
- 2) For the receiver of the COOKIE ECHO chunk, the only CONFIRMED address is the address to which the packet containing the INIT ACK chunk was sent.
- 3) All other addresses not covered by rules 1 and 2 are considered UNCONFIRMED and are subject to probing for verification.

To probe an address for verification, an endpoint will send HEARTBEAT chunks including a 64-bit random nonce and a path indicator (to identify the address that the HEARTBEAT chunk is sent to) within the Heartbeat Info parameter.

Upon receipt of the HEARTBEAT ACK chunk, a verification is made that the nonce included in the Heartbeat Info parameter is the one sent to the address indicated inside the Heartbeat Info parameter. When this match occurs, the address that the original HEARTBEAT was sent to is now considered CONFIRMED and available for normal data transfer.

These probing procedures are started when an association moves to the ESTABLISHED state and are ended when all paths are confirmed.

In each RTO, a probe MAY be sent on an active UNCONFIRMED path in an attempt to move it to the CONFIRMED state. If during this probing the path becomes inactive, this rate is lowered to the normal HEARTBEAT rate. At the expiration of the RTO timer, the error counter of any path that was probed but not CONFIRMED is incremented by one and subjected to path failure detection, as defined in Section 8.2. When probing UNCONFIRMED addresses, however, the association overall error count is not incremented.

The number of packets containing HEARTBEAT chunks sent at each RTO SHOULD be limited by the 'HB.Max.Burst' parameter. It is an implementation decision as to how to distribute packets containing HEARTBEAT chunks to the peer's addresses for path verification.

Whenever a path is confirmed, an indication MAY be given to the upper layer.

An endpoint MUST NOT send any chunks to an UNCONFIRMED address, with the following exceptions:

- * A HEARTBEAT chunk including a nonce MAY be sent to an UNCONFIRMED address.
- * A HEARTBEAT ACK chunk MAY be sent to an UNCONFIRMED address.
- * A COOKIE ACK chunk MAY be sent to an UNCONFIRMED address, but it MUST be bundled with a HEARTBEAT chunk including a nonce. An implementation that does not support bundling MUST NOT send a COOKIE ACK chunk to an UNCONFIRMED address.
- * A COOKIE ECHO chunk MAY be sent to an UNCONFIRMED address, but it MUST be bundled with a HEARTBEAT chunk including a nonce, and the size of the SCTP packet MUST NOT exceed the PMTU. If the implementation does not support bundling or if the bundled COOKIE ECHO chunk plus HEARTBEAT chunk (including nonce) would result in an SCTP packet larger than the PMTU, then the implementation MUST NOT send a COOKIE ECHO chunk to an UNCONFIRMED address.

6. User Data Transfer

Data transmission MUST only happen in the ESTABLISHED, SHUTDOWN-PENDING, and SHUTDOWN-RECEIVED states. The only exception to this is that DATA chunks are allowed to be bundled with an outbound COOKIE ECHO chunk when in the COOKIE-WAIT state.

DATA chunks MUST only be received according to the rules below in ESTABLISHED, SHUTDOWN-PENDING, and SHUTDOWN-SENT states. A DATA chunk received in CLOSED is out of the blue and SHOULD be handled per Section 8.4. A DATA chunk received in any other state SHOULD be discarded.

A SACK chunk MUST be processed in ESTABLISHED, SHUTDOWN-PENDING, and SHUTDOWN-RECEIVED states. An incoming SACK chunk MAY be processed in COOKIE-ECHOED. A SACK chunk in the CLOSED state is out of the blue and SHOULD be processed according to the rules in Section 8.4. A SACK chunk received in any other state SHOULD be discarded.

For transmission efficiency, SCTP defines mechanisms for bundling of small user messages and fragmentation of large user messages. The following diagram depicts the flow of user messages through SCTP.

In this section, the term "data sender" refers to the endpoint that transmits a DATA chunk and the term "data receiver" refers to the endpoint that receives a DATA chunk. A data receiver will transmit SACK chunks.

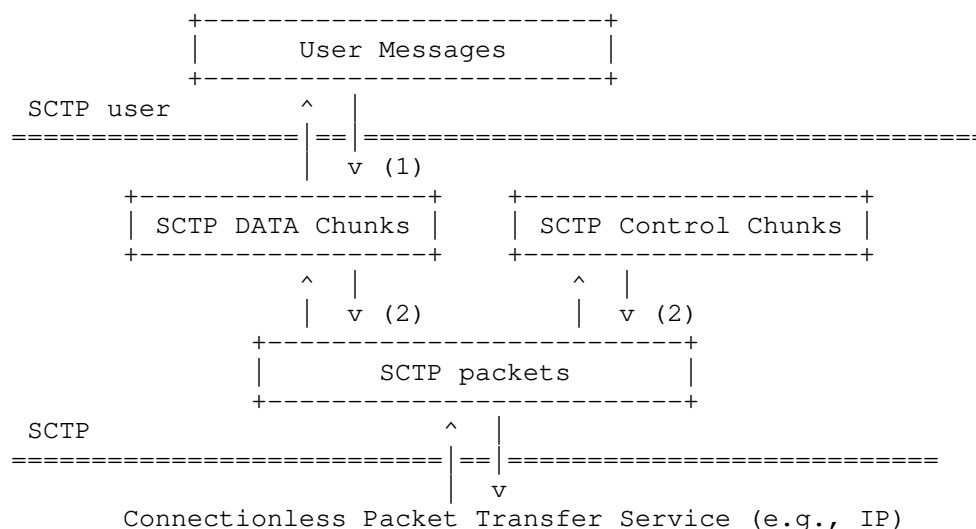


Figure 6: Illustration of User Data Transfer

The following applies:

- 1) When converting user messages into DATA chunks, an endpoint **MUST** fragment large user messages into multiple DATA chunks. The size of each DATA chunk **SHOULD** be smaller than or equal to the Association Maximum DATA Chunk Size (AMDCS). The data receiver will normally reassemble the fragmented message from DATA chunks before delivery to the user (see Section 6.9 for details).
- 2) Multiple DATA and control chunks **MAY** be bundled by the sender into a single SCTP packet for transmission, as long as the final size of the SCTP packet does not exceed the current PMTU. The receiver will unbundle the packet back into the original chunks. Control chunks **MUST** come before DATA chunks in the packet.

The fragmentation and bundling mechanisms, as detailed in Section 6.9 and Section 6.10, are **OPTIONAL** to implement by the data sender, but they **MUST** be implemented by the data receiver, i.e., an endpoint **MUST** properly receive and process bundled or fragmented data.

6.1. Transmission of DATA Chunks

This section specifies the rules for sending DATA chunks. In particular, it defines zero window probing, which is required to avoid the indefinite stalling of an association in case of a loss of packets containing SACK chunks performing window updates.

This document is specified as if there is a single retransmission timer per destination transport address, but implementations **MAY** have a retransmission timer for each DATA chunk.

The following general rules **MUST** be applied by the data sender for transmission and/or retransmission of outbound DATA chunks:

- A) At any given time, the data sender **MUST NOT** transmit new data to any destination transport address if its peer's `rwnd` indicates that the peer has no buffer space (i.e., `rwnd` is smaller than the size of the next DATA chunk; see Section 6.2.1), except for zero window probes.

A zero window probe is a DATA chunk sent when the receiver has no buffer space. This rule allows the sender to probe for a change in `rwnd` that the sender missed due to the SACK chunks having been lost in transit from the data receiver to the data sender. A zero window probe **MUST** only be sent when the `cwnd` allows (see Rule B below). A zero window probe **SHOULD** only be sent when all outstanding DATA chunks have been cumulatively acknowledged and no DATA chunks are in flight. Senders **MUST** support zero window probing.

If the sender continues to receive SACK chunks from the peer while doing zero window probing, the unacknowledged window probes SHOULD NOT increment the error counter for the association or any destination transport address. This is because the receiver could keep its window closed for an indefinite time. Section 6.2 describes the receiver behavior when it advertises a zero window. The sender SHOULD send the first zero window probe after 1 RTO when it detects that the receiver has closed its window and SHOULD increase the probe interval exponentially afterwards. Also note that the cwnd SHOULD be adjusted according to Section 7.2.1. Zero window probing does not affect the calculation of cwnd.

The sender MUST also have an algorithm for sending new DATA chunks to avoid silly window syndrome (SWS) as described in [RFC1122]. The algorithm can be similar to the one described in Section 4.2.3.4 of [RFC1122].

- B) At any given time, the sender MUST NOT transmit new data to a given transport address if it has $cwnd + (PMDCS - 1)$ or more bytes of data outstanding to that transport address. If data is available, the sender SHOULD exceed cwnd by up to $(PMDCS - 1)$ bytes on a new data transmission if the flightsize does not currently reach cwnd. The breach of cwnd MUST constitute one packet only.
- C) When the time comes for the sender to transmit, before sending new DATA chunks, the sender MUST first transmit any DATA chunks that are marked for retransmission (limited by the current cwnd).
- D) When the time comes for the sender to transmit new DATA chunks, the protocol parameter 'Max.Burst' SHOULD be used to limit the number of packets sent. The limit MAY be applied by adjusting cwnd temporarily, as follows:

```
if ((flightsize + Max.Burst * PMDCS) < cwnd)
    cwnd = flightsize + Max.Burst * PMDCS;
```

Or, it MAY be applied by strictly limiting the number of packets emitted by the output routine. When calculating the number of packets to transmit, and particularly when using the formula above, cwnd SHOULD NOT be changed permanently.

- E) Then, the sender can send as many new DATA chunks as rule A and rule B allow.

Multiple DATA chunks committed for transmission MAY be bundled in a single packet. Furthermore, DATA chunks being retransmitted MAY be bundled with new DATA chunks, as long as the resulting SCTP packet size does not exceed the PMTU. A ULP can request that no bundling is performed, but this only turns off any delays that an SCTP implementation might be using to increase bundling efficiency. It does not in itself stop all bundling from occurring (i.e., in case of congestion or retransmission).

Before an endpoint transmits a DATA chunk, if any received DATA chunks have not been acknowledged (e.g., due to delayed ack), the sender SHOULD create a SACK chunk and bundle it with the outbound DATA chunk, as long as the size of the final SCTP packet does not exceed the current PMTU. See Section 6.2.

When the window is full (i.e., transmission is disallowed by rule A and/or rule B), the sender MAY still accept send requests from its upper layer, but MUST transmit no more DATA chunks until some or all of the outstanding DATA chunks are acknowledged and transmission is allowed by rule A and rule B again.

Whenever a transmission or retransmission is made to any address, if the T3-rtx timer of that address is not currently running, the sender MUST start that timer. If the timer for that address is already running, the sender MUST restart the timer if the earliest (i.e., lowest TSN) outstanding DATA chunk sent to that address is being retransmitted. Otherwise, the data sender MUST NOT restart the timer.

When starting or restarting the T3-rtx timer, the timer value SHOULD be adjusted according to the timer rules defined in Section 6.3.2 and Section 6.3.3.

The data sender MUST NOT use a TSN that is more than $2^{31} - 1$ above the beginning TSN of the current send window.

For each stream, the data sender MUST NOT have more than $2^{16} - 1$ ordered user messages in the current send window.

Whenever the sender of a DATA chunk can benefit from the corresponding SACK chunk being sent back without delay, the sender MAY set the I bit in the DATA chunk header. Please note that why the sender has set the I bit is irrelevant to the receiver.

Reasons for setting the I bit include, but are not limited to, the following (see Section 4 of [RFC7053] for a discussion of the benefits):

- * The application requests that the I bit of the last DATA chunk of a user message be set when providing the user message to the SCTP implementation (see Section 11.1).
- * The sender is in the SHUTDOWN-PENDING state.
- * The sending of a DATA chunk fills the congestion or receiver window.

6.2. Acknowledgement on Reception of DATA Chunks

The SCTP endpoint MUST always acknowledge the reception of each valid DATA chunk when the DATA chunk received is inside its receive window.

When the receiver's advertised window is 0, the receiver MUST drop any new incoming DATA chunk with a TSN larger than the largest TSN received so far. Also, if the new incoming DATA chunk holds a TSN value less than the largest TSN received so far, then the receiver SHOULD drop the largest TSN held for reordering and accept the new incoming DATA chunk. In either case, if such a DATA chunk is dropped, the receiver MUST immediately send back a SACK chunk with the current receive window showing only DATA chunks received and accepted so far. The dropped DATA chunk(s) MUST NOT be included in the SACK chunk, as they were not accepted. The receiver MUST also have an algorithm for advertising its receive window to avoid receiver silly window syndrome (SWS), as described in [RFC1122]. The algorithm can be similar to the one described in Section 4.2.3.3 of [RFC1122].

The guidelines on delayed acknowledgement algorithm specified in Section 4.2 of [RFC5681] SHOULD be followed. Specifically, an acknowledgement SHOULD be generated for at least every second packet (not every second DATA chunk) received, and SHOULD be generated within 200 ms of the arrival of any unacknowledged DATA chunk. In some situations, it might be beneficial for an SCTP transmitter to be more conservative than the algorithms detailed in this document allow. However, an SCTP transmitter MUST NOT be more aggressive in sending SACK chunks than the following algorithms allow.

An SCTP receiver MUST NOT generate more than one SACK chunk for every incoming packet, other than to update the offered window as the receiving application consumes new data. When the window opens up, an SCTP receiver SHOULD send additional SACK chunks to update the window even if no new data is received. The receiver MUST avoid sending a large number of window updates -- in particular, large bursts of them. One way to achieve this is to send a window update only if the window can be increased by at least a quarter of the receive buffer size of the association.

Implementation Note: The maximum delay for generating an acknowledgement MAY be configured by the SCTP administrator, either statically or dynamically, in order to meet the specific timing requirement of the protocol being carried.

An implementation MUST NOT allow the maximum delay (protocol parameter 'SACK.Delay') to be configured to be more than 500 ms. In other words, an implementation MAY lower the value of 'SACK.Delay' below 500 ms but MUST NOT raise it above 500 ms.

Acknowledgements MUST be sent in SACK chunks unless shutdown was requested by the ULP, in which case an endpoint MAY send an acknowledgement in the SHUTDOWN chunk. A SACK chunk can acknowledge the reception of multiple DATA chunks. See Section 3.3.4 for SACK chunk format. In particular, the SCTP endpoint MUST fill in the Cumulative TSN Ack field to indicate the latest sequential TSN (of a valid DATA chunk) it has received. Any received DATA chunks with TSN greater than the value in the Cumulative TSN Ack field are reported in the Gap Ack Block fields. The SCTP endpoint MUST report as many Gap Ack Blocks as can fit in a single SACK chunk such that the size of the SCTP packet does not exceed the current PMTU.

The SHUTDOWN chunk does not contain Gap Ack Block fields. Therefore, the endpoint SHOULD use a SACK chunk instead of the SHUTDOWN chunk to acknowledge DATA chunks received out of order.

Upon receipt of an SCTP packet containing a DATA chunk with the I bit set, the receiver SHOULD NOT delay the sending of the corresponding SACK chunk, i.e., the receiver SHOULD immediately respond with the corresponding SACK chunk.

When a packet arrives with duplicate DATA chunk(s) and with no new DATA chunk(s), the endpoint MUST immediately send a SACK chunk with no delay. If a packet arrives with duplicate DATA chunk(s) bundled with new DATA chunks, the endpoint MAY immediately send a SACK chunk. Normally, receipt of duplicate DATA chunks will occur when the original SACK chunk was lost and the peer's RTO has expired. The duplicate TSN number(s) SHOULD be reported in the SACK chunk as duplicate.

When an endpoint receives a SACK chunk, it MAY use the duplicate TSN information to determine if SACK chunk loss is occurring. Further use of this data is for future study.

The data receiver is responsible for maintaining its receive buffers. The data receiver SHOULD notify the data sender in a timely manner of changes in its ability to receive data. How an implementation manages its receive buffers is dependent on many factors (e.g.,

operating system, memory management system, amount of memory, etc.). However, the data sender strategy defined in Section 6.2.1 is based on the assumption of receiver operation similar to the following:

- A) At initialization of the association, the endpoint tells the peer how much receive buffer space it has allocated to the association in the INIT or INIT ACK chunk. The endpoint sets `a_rwnd` to this value.
- B) As DATA chunks are received and buffered, decrement `a_rwnd` by the number of bytes received and buffered. This is, in effect, closing `rwnd` at the data sender and restricting the amount of data it can transmit.
- C) As DATA chunks are delivered to the ULP and released from the receive buffers, increment `a_rwnd` by the number of bytes delivered to the upper layer. This is, in effect, opening up `rwnd` on the data sender and allowing it to send more data. The data receiver SHOULD NOT increment `a_rwnd` unless it has released bytes from its receive buffer. For example, if the receiver is holding fragmented DATA chunks in a reassembly queue, it SHOULD NOT increment `a_rwnd`.
- D) When sending a SACK chunk, the data receiver SHOULD place the current value of `a_rwnd` into the `a_rwnd` field. The data receiver SHOULD take into account that the data sender will not retransmit DATA chunks that are acked via the Cumulative TSN Ack (i.e., will drop from its retransmit queue).

Under certain circumstances, the data receiver MAY drop DATA chunks that it has received but has not released from its receive buffers (i.e., delivered to the ULP). These DATA chunks might have been acked in Gap Ack Blocks. For example, the data receiver might be holding data in its receive buffers while reassembling a fragmented user message from its peer when it runs out of receive buffer space. It MAY drop these DATA chunks even though it has acknowledged them in Gap Ack Blocks. If a data receiver drops DATA chunks, it MUST NOT include them in Gap Ack Blocks in subsequent SACK chunks until they are received again via retransmission. In addition, the endpoint SHOULD take into account the dropped data when calculating its `a_rwnd`.

An endpoint SHOULD NOT revoke a SACK chunk and discard data. Only in extreme circumstances might an endpoint use this procedure (such as out of buffer space). The data receiver SHOULD take into account that dropping data that has been acked in Gap Ack Blocks can result in suboptimal retransmission strategies in the data sender and thus in suboptimal performance.

The following example illustrates the use of delayed acknowledgements:

Endpoint A	Endpoint Z
{App sends 3 messages; strm 0}	
DATA [TSN=7,Strm=0,Seq=3] ----->	(ack delayed)
(Start T3-rtx timer)	
DATA [TSN=8,Strm=0,Seq=4] ----->	(send ack)
	/----- SACK [TSN Ack=8,block=0]
(cancel T3-rtx timer) <-----/	
DATA [TSN=9,Strm=0,Seq=5] ----->	(ack delayed)
(Start T3-rtx timer)	
	...
	{App sends 1 message; strm 1}
	(bundle SACK with DATA)
	/----- SACK [TSN Ack=9,block=0] \
	DATA [TSN=6,Strm=1,Seq=2]
(cancel T3-rtx timer) <-----/	(Start T3-rtx timer)
(ack delayed)	
(send ack)	
SACK [TSN Ack=6,block=0] ----->	(cancel T3-rtx timer)

Figure 7: Delayed Acknowledgement Example

If an endpoint receives a DATA chunk with no user data (i.e., the Length field is set to 16), it SHOULD send an ABORT chunk with a "No User Data" error cause.

An endpoint SHOULD NOT send a DATA chunk with no user data part. This avoids the need to be able to return a zero-length user message in the API, especially in the socket API as specified in [RFC6458] for details.

6.2.1. Processing a Received SACK Chunk

Each SACK chunk an endpoint receives contains an `a_rwnd` value. This value represents the amount of buffer space the data receiver, at the time of transmitting the SACK chunk, has left of its total receive buffer space (as specified in the INIT/INIT ACK chunk). Using `a_rwnd`, Cumulative TSN Ack, and Gap Ack Blocks, the data sender can develop a representation of the peer's receive buffer space.

One of the problems the data sender takes into account when processing a SACK chunk is that a SACK chunk can be received out of order. That is, a SACK chunk sent by the data receiver can pass an earlier SACK chunk and be received first by the data sender. If a SACK chunk is received out of order, the data sender can develop an incorrect view of the peer's receive buffer space.

Since there is no explicit identifier that can be used to detect out-of-order SACK chunks, the data sender uses heuristics to determine if a SACK chunk is new.

An endpoint SHOULD use the following rules to calculate the `rwnd`, using the `a_rwnd` value, the Cumulative TSN Ack, and Gap Ack Blocks in a received SACK chunk.

- A) At the establishment of the association, the endpoint initializes the `rwnd` to the Advertised Receiver Window Credit (`a_rwnd`) the peer specified in the INIT or INIT ACK chunk.
- B) Any time a DATA chunk is transmitted (or retransmitted) to a peer, the endpoint subtracts the data size of the chunk from the `rwnd` of that peer.
- C) Any time a DATA chunk is marked for retransmission, either via T3-rtx timer expiration (Section 6.3.3) or via Fast Retransmit (Section 7.2.4), add the data size of those chunks to the `rwnd`.
- D) Any time a SACK chunk arrives, the endpoint performs the following:
 - i) If Cumulative TSN Ack is less than the Cumulative TSN Ack Point, then drop the SACK chunk. Since Cumulative TSN Ack is monotonically increasing, a SACK chunk whose Cumulative TSN Ack is less than the Cumulative TSN Ack Point indicates an out-of-order SACK chunk.
 - ii) Set `rwnd` equal to the newly received `a_rwnd` minus the number of bytes still outstanding after processing the Cumulative TSN Ack and the Gap Ack Blocks.
 - iii) If the SACK chunk is missing a TSN that was previously acknowledged via a Gap Ack Block (e.g., the data receiver reneged on the data), then consider the corresponding DATA that might be possibly missing: Count one miss indication towards Fast Retransmit as described in Section 7.2.4, and if no retransmit timer is running for the destination address to which the DATA chunk was originally transmitted, then T3-rtx is started for that destination address.

- iv) If the Cumulative TSN Ack matches or exceeds the Fast Recovery exitpoint (Section 7.2.4), Fast Recovery is exited.

6.3. Management of Retransmission Timer

An SCTP endpoint uses a retransmission timer T3-rtx to ensure data delivery in the absence of any feedback from its peer. The duration of this timer is referred to as RTO (retransmission timeout).

When an endpoint's peer is multi-homed, the endpoint will calculate a separate RTO for each different destination transport address of its peer endpoint.

The computation and management of RTO in SCTP follow closely how TCP manages its retransmission timer. To compute the current RTO, an endpoint maintains two state variables per destination transport address: SRTT (smoothed round-trip time) and RTTVAR (round-trip time variation).

6.3.1. RTO Calculation

The rules governing the computation of SRTT, RTTVAR, and RTO are as follows:

- C1) Until an RTT measurement has been made for a packet sent to the given destination transport address, set RTO to the protocol parameter 'RTO.Initial'.

- C2) When the first RTT measurement R is made, perform

```
SRTT = R;  
RTTVAR = R/2;  
RTO = SRTT + 4 * RTTVAR;
```

- C3) When a new RTT measurement R' is made, perform:

```
RTTVAR = (1 - RTO.Beta) * RTTVAR + RTO.Beta * |SRTT - R'|;  
SRTT = (1 - RTO.Alpha) * SRTT + RTO.Alpha * R';
```

Note: The value of SRTT used in the update to RTTVAR is its value before updating SRTT itself using the second assignment.

After the computation, update

```
RTO = SRTT + 4 * RTTVAR;
```

C4) When data is in flight and when allowed by rule C5 below, a new RTT measurement MUST be made each round trip. Furthermore, new RTT measurements SHOULD be made no more than once per round trip for a given destination transport address. There are two reasons for this recommendation: First, it appears that measuring more frequently often does not in practice yield any significant benefit [ALLMAN99]; second, if measurements are made more often, then the values of 'RTO.Alpha' and 'RTO.Beta' in rule C3 above SHOULD be adjusted so that SRTT and RTTVAR still adjust to changes at roughly the same rate (in terms of how many round trips it takes them to reflect new values) as they would if making only one measurement per round-trip and using 'RTO.Alpha' and 'RTO.Beta' as given in rule C3. However, the exact nature of these adjustments remains a research issue.

C5) Karn's algorithm: RTT measurements MUST NOT be made using chunks that were retransmitted (and thus for which it is ambiguous whether the reply was for the first instance of the chunk or for a later instance).

RTT measurements SHOULD only be made using a DATA chunk with TSN r , if no DATA chunk with TSN less than or equal to r was retransmitted since the DATA chunk with TSN r was sent first.

C6) Whenever RTO is computed, if it is less than 'RTO.Min' seconds then it is rounded up to 'RTO.Min' seconds. The reason for this rule is that RTOs that do not have a high minimum value are susceptible to unnecessary timeouts [ALLMAN99].

C7) A maximum value MAY be placed on RTO provided it is at least 'RTO.max' seconds.

There is no requirement for the clock granularity G used for computing RTT measurements and the different state variables, other than:

G1) Whenever RTTVAR is computed, if $RTTVAR == 0$, then adjust $RTTVAR = G$.

Experience [ALLMAN99] has shown that finer clock granularities (less than 100 msec) perform somewhat better than more coarse granularities.

See Section 16 for suggested parameter values.

6.3.2. Retransmission Timer Rules

The rules for managing the retransmission timer are as follows:

- R1) Every time a DATA chunk is sent to any address (including a retransmission), if the T3-rtx timer of that address is not running, start it running so that it will expire after the RTO of that address. The RTO used here is that obtained after any doubling due to previous T3-rtx timer expirations on the corresponding destination address as discussed in rule E2 below.
- R2) Whenever all outstanding data sent to an address have been acknowledged, turn off the T3-rtx timer of that address.
- R3) Whenever a SACK chunk is received that acknowledges the DATA chunk with the earliest outstanding TSN for that address, restart the T3-rtx timer for that address with its current RTO (if there is still outstanding data on that address).
- R4) Whenever a SACK chunk is received missing a TSN that was previously acknowledged via a Gap Ack Block, start the T3-rtx for the destination address to which the DATA chunk was originally transmitted if it is not already running.

The following example shows the use of various timer rules (assuming that the receiver uses delayed acks).

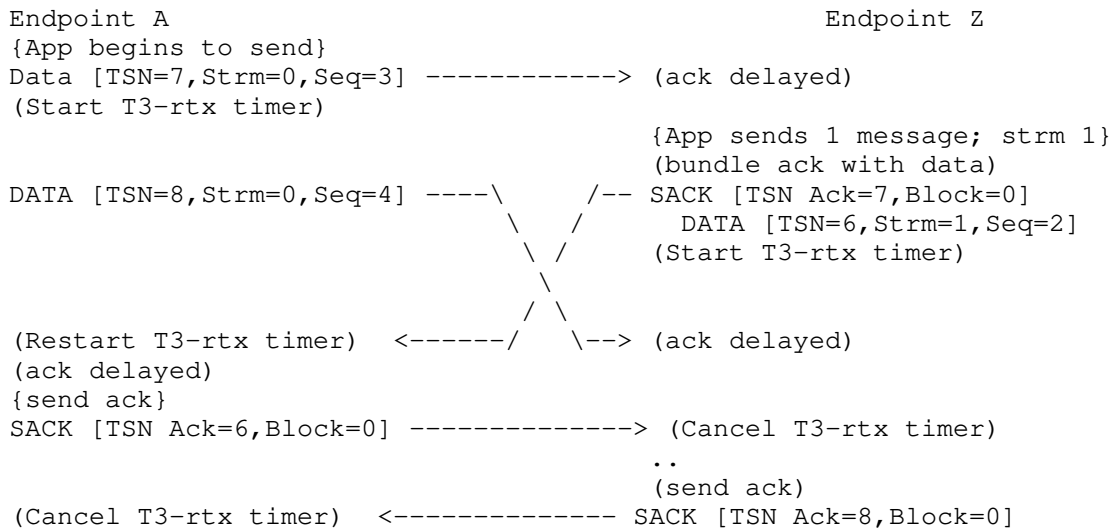


Figure 8: Timer Rule Examples

6.3.3. Handle T3-rtx Expiration

Whenever the retransmission timer T3-rtx expires for a destination address, do the following:

- E1) For the destination address for which the timer expires, adjust its ssthresh with rules defined in Section 7.2.3 and set the cwnd = PMDCS.
- E2) For the destination address for which the timer expires, set $RTO = RTO * 2$ ("back off the timer"). The maximum value discussed in rule C7 above ('RTO.max') MAY be used to provide an upper bound to this doubling operation.
- E3) Determine how many of the earliest (i.e., lowest TSN) outstanding DATA chunks for the address for which the T3-rtx has expired will fit into a single SCTP packet, subject to the PMTU corresponding to the destination transport address to which the retransmission is being sent (this might be different from the address for which the timer expires; see Section 6.4). Call this value K. Bundle and retransmit those K DATA chunks in a single packet to the destination endpoint.
- E4) Start the retransmission timer T3-rtx on the destination address to which the retransmission is sent, if rule R1 above indicates to do so. The RTO to be used for starting T3-rtx SHOULD be the one for the destination address to which the retransmission is sent, which, when the receiver is multi-homed, might be different from the destination address for which the timer expired (see Section 6.4 below).

After retransmitting, once a new RTT measurement is obtained (which can happen only when new data has been sent and acknowledged, per rule C5, or for a measurement made from a HEARTBEAT chunk; see Section 8.3), the computation in rule C3 is performed, including the computation of RTO, which might result in "collapsing" RTO back down after it has been subject to doubling (rule E2).

Any DATA chunks that were sent to the address for which the T3-rtx timer expired but did not fit in an SCTP packet of size smaller than or equal to the PMTU (rule E3 above) SHOULD be marked for retransmission and sent as soon as cwnd allows (normally, when a SACK chunk arrives).

The final rule for managing the retransmission timer concerns failover (see Section 6.4.1):

F1) Whenever an endpoint switches from the current destination transport address to a different one, the current retransmission timers are left running. As soon as the endpoint transmits a packet containing DATA chunk(s) to the new transport address, start the timer on that transport address, using the RTO value of the destination address to which the data is being sent, if rule R1 indicates to do so.

6.4. Multi-Homed SCTP Endpoints

An SCTP endpoint is considered multi-homed if there is more than one transport address that can be used as a destination address to reach that endpoint.

Moreover, the ULP of an endpoint selects one of the multiple destination addresses of a multi-homed peer endpoint as the primary path (see Section 5.1.2 and Section 11.1 for details).

By default, an endpoint SHOULD always transmit to the primary path, unless the SCTP user explicitly specifies the destination transport address (and possibly source transport address) to use.

An endpoint SHOULD transmit reply chunks (e.g., INIT ACK, COOKIE ACK, HEARTBEAT ACK) in response to control chunks to the same destination transport address from which it received the control chunk to which it is replying.

The selection of the destination transport address for packets containing SACK chunks is implementation dependent. However, an endpoint SHOULD NOT vary the destination transport address of a SACK chunk when it receives DATA chunks coming from the same source address.

When acknowledging multiple DATA chunks received in packets from different source addresses in a single SACK chunk, the SACK chunk MAY be transmitted to one of the destination transport addresses from which the DATA or control chunks being acknowledged were received.

When a receiver of a duplicate DATA chunk sends a SACK chunk to a multi-homed endpoint, it MAY be beneficial to vary the destination address and not use the source address of the DATA chunk. The reason is that receiving a duplicate from a multi-homed endpoint might indicate that the return path (as specified in the source address of the DATA chunk) for the SACK chunk is broken.

Furthermore, when its peer is multi-homed, an endpoint SHOULD try to retransmit a chunk that timed out to an active destination transport address that is different from the last destination address to which the chunk was sent.

When its peer is multi-homed, an endpoint SHOULD send fast retransmissions to the same destination transport address to which the original data was sent. If the primary path has been changed and the original data was sent to the old primary path before the Fast Retransmit, the implementation MAY send it to the new primary path.

Retransmissions do not affect the total outstanding data count. However, if the DATA chunk is retransmitted onto a different destination address, both the outstanding data counts on the new destination address and the old destination address to which the data chunk was last sent is adjusted accordingly.

6.4.1. Failover from an Inactive Destination Address

Some of the transport addresses of a multi-homed SCTP endpoint might become inactive due to either the occurrence of certain error conditions (see Section 8.2) or adjustments from the SCTP user.

When there is outbound data to send and the primary path becomes inactive (e.g., due to failures), or where the SCTP user explicitly requests to send data to an inactive destination transport address, before reporting an error to its ULP, the SCTP endpoint SHOULD try to send the data to an alternate active destination transport address if one exists.

When retransmitting data that timed out, if the endpoint is multi-homed, it needs to consider each source-destination address pair in its retransmission selection policy. When retransmitting timed-out data, the endpoint SHOULD attempt to pick the most divergent source-destination pair from the original source-destination pair to which the packet was transmitted.

Note: Rules for picking the most divergent source-destination pair are an implementation decision and are not specified within this document.

6.5. Stream Identifier and Stream Sequence Number

Every DATA chunk MUST carry a valid stream identifier. If an endpoint receives a DATA chunk with an invalid stream identifier, it SHOULD acknowledge the reception of the DATA chunk following the normal procedure, immediately send an ERROR chunk with cause set to "Invalid Stream Identifier" (see Section 3.3.10), and discard the DATA chunk. The endpoint MAY bundle the ERROR chunk and the SACK chunk in the same packet.

The Stream Sequence Number in all the outgoing streams MUST start from 0 when the association is established. The Stream Sequence Number of an outgoing stream MUST be incremented by 1 for each ordered user message sent on that outgoing stream. In particular, when the Stream Sequence Number reaches the value 65535 the next Stream Sequence Number MUST be set to 0. For unordered user messages the Stream Sequence Number MUST NOT be changed.

6.6. Ordered and Unordered Delivery

Within a stream, an endpoint MUST deliver DATA chunks received with the U flag set to 0 to the upper layer according to the order of their Stream Sequence Number. If DATA chunks arrive out of order of their Stream Sequence Number, the endpoint MUST hold the received DATA chunks from delivery to the ULP until they are reordered.

However, an SCTP endpoint can indicate that no ordered delivery is required for a particular DATA chunk transmitted within the stream by setting the U flag of the DATA chunk to 1.

When an endpoint receives a DATA chunk with the U flag set to 1, it bypasses the ordering mechanism and immediately deliver the data to the upper layer (after reassembly if the user data is fragmented by the data sender).

This provides an effective way of transmitting "out-of-band" data in a given stream. Also, a stream can be used as an "unordered" stream by simply setting the U flag to 1 in all DATA chunks sent through that stream.

Implementation Note: When sending an unordered DATA chunk, an implementation MAY choose to place the DATA chunk in an outbound packet that is at the head of the outbound transmission queue if possible.

The 'Stream Sequence Number' field in a DATA chunk with U flag set to 1 has no significance. The sender can fill the 'Stream Sequence Number' with arbitrary value, but the receiver MUST ignore the field.

Note: When transmitting ordered and unordered data, an endpoint does not increment its Stream Sequence Number when transmitting a DATA chunk with U flag set to 1.

6.7. Report Gaps in Received DATA TSNs

Upon the reception of a new DATA chunk, an endpoint examines the continuity of the TSNs received. If the endpoint detects a gap in the received DATA chunk sequence, it SHOULD send a SACK chunk with Gap Ack Blocks immediately. The data receiver continues sending a SACK chunk after receipt of each SCTP packet that does not fill the gap.

Based on the Gap Ack Block from the received SACK chunk, the endpoint can calculate the missing DATA chunks and make decisions on whether to retransmit them (see Section 6.2.1 for details).

Multiple gaps can be reported in one single SACK chunk (see Section 3.3.4).

When its peer is multi-homed, the SCTP endpoint SHOULD always try to send the SACK chunk to the same destination address from which the last DATA chunk was received.

Upon the reception of a SACK chunk, the endpoint MUST remove all DATA chunks that have been acknowledged by the SACK chunk's Cumulative TSN Ack from its transmit queue. All DATA chunks with TSNs not included in the Gap Ack Blocks that are smaller than the highest acknowledged TSN reported in the SACK chunk MUST be treated as "missing" by the sending endpoint. The number of "missing" reports for each outstanding DATA chunk MUST be recorded by the data sender to make retransmission decisions. See Section 7.2.4 for details.

The following example shows the use of SACK chunk to report a gap.

```

Endpoint A                                Endpoint Z
{App sends 3 messages; strm 0}
DATA [TSN=6,Strm=0,Seq=2] -----> (ack delayed)
(Start T3-rtx timer)

DATA [TSN=7,Strm=0,Seq=3] -----> X (lost)

DATA [TSN=8,Strm=0,Seq=4] -----> (gap detected,
                                   immediately send ack)
                                   /----- SACK [TSN Ack=6,Block=1,
                                   /          Start=2,End=2]
                                   <-----/
(remove 6 from out-queue,
 and mark 7 as "1" missing report)

```

Figure 9: Reporting a Gap using SACK Chunk

The maximum number of Gap Ack Blocks that can be reported within a single SACK chunk is limited by the current PMTU. When a single SACK chunk cannot cover all the Gap Ack Blocks needed to be reported due to the PMTU limitation, the endpoint **MUST** send only one SACK chunk. This single SACK chunk **MUST** report the Gap Ack Blocks from the lowest to highest TSNs, within the size limit set by the PMTU, and leave the remaining highest TSN numbers unacknowledged.

6.8. CRC32c Checksum Calculation

When sending an SCTP packet, the endpoint **MUST** strengthen the data integrity of the transmission by including the CRC32c checksum value calculated on the packet, as described below.

After the packet is constructed (containing the SCTP common header and one or more control or DATA chunks), the transmitter **MUST**

- 1) fill in the proper Verification Tag in the SCTP common header and initialize the checksum field to 0,
- 2) calculate the CRC32c checksum of the whole packet, including the SCTP common header and all the chunks (refer to Appendix A for details of the CRC32c algorithm); and
- 3) put the resultant value into the checksum field in the common header, and leave the rest of the bits unchanged.

When an SCTP packet is received, the receiver **MUST** first check the CRC32c checksum as follows:

- 1) Store the received CRC32c checksum value aside.

- 2) Replace the 32 bits of the checksum field in the received SCTP packet with 0 and calculate a CRC32c checksum value of the whole received packet.
- 3) Verify that the calculated CRC32c checksum is the same as the received CRC32c checksum. If it is not, the receiver MUST treat the packet as an invalid SCTP packet.

The default procedure for handling invalid SCTP packets is to silently discard them.

Any hardware implementation SHOULD permit alternative verification of the CRC in software.

6.9. Fragmentation and Reassembly

An endpoint MAY support fragmentation when sending DATA chunks, but it MUST support reassembly when receiving DATA chunks. If an endpoint supports fragmentation, it MUST fragment a user message if the size of the user message to be sent causes the outbound SCTP packet size to exceed the current PMTU. An endpoint that does not support fragmentation and is requested to send a user message such that the outbound SCTP packet size would exceed the current PMTU MUST return an error to its upper layer and MUST NOT attempt to send the user message.

An SCTP implementation MAY provide a mechanism to the upper layer that disables fragmentation when sending DATA chunks. When fragmentation of DATA chunks is disabled, the SCTP implementation MUST behave in the same way an implementation that does not support fragmentation, i.e., it rejects calls that would result in sending SCTP packets that exceed the current PMTU.

Implementation Note: In this error case, the SEND primitive discussed in Section 11.1 would need to return an error to the upper layer.

If its peer is multi-homed, the endpoint SHOULD choose a DATA chunk size smaller than or equal to the AMDCS.

Once a user message is fragmented, it cannot be re-fragmented. Instead, if the PMTU has been reduced, then IP fragmentation MUST be used. Therefore, an SCTP association can fail if IP fragmentation is not working on any path. Please see Section 7.3 for details of PMTU discovery.

When determining when to fragment, the SCTP implementation MUST take into account the SCTP packet header as well as the DATA chunk header(s). The implementation MUST also take into account the space required for a SACK chunk if bundling a SACK chunk with the DATA chunk.

Fragmentation takes the following steps:

- 1) The data sender MUST break the user message into a series of DATA chunks. The sender SHOULD choose a size of DATA chunks that is smaller than or equal to the AMDCS.
- 2) The transmitter MUST then assign, in sequence, a separate TSN to each of the DATA chunks in the series. The transmitter assigns the same Stream Sequence Number to each of the DATA chunks. If the user indicates that the user message is to be delivered using unordered delivery, then the U flag of each DATA chunk of the user message MUST be set to 1.
- 3) The transmitter MUST also set the B/E bits of the first DATA chunk in the series to '10', the B/E bits of the last DATA chunk in the series to '01', and the B/E bits of all other DATA chunks in the series to '00'.

An endpoint MUST recognize fragmented DATA chunks by examining the B/E bits in each of the received DATA chunks, and queue the fragmented DATA chunks for reassembly. Once the user message is reassembled, SCTP passes the reassembled user message to the specific stream for possible reordering and final dispatching.

If the data receiver runs out of buffer space while still waiting for more fragments to complete the reassembly of the message, it SHOULD dispatch part of its inbound message through a partial delivery API (see Section 11), freeing some of its receive buffer space so that the rest of the message can be received.

6.10. Bundling

An endpoint bundles chunks by simply including multiple chunks in one outbound SCTP packet. The total size of the resultant SCTP packet MUST be less than or equal to the current PMTU.

If its peer endpoint is multi-homed, the sending endpoint SHOULD choose a size no larger than the PMTU of the current primary path.

When bundling control chunks with DATA chunks, an endpoint MUST place control chunks first in the outbound SCTP packet. The transmitter MUST transmit DATA chunks within an SCTP packet in increasing order of TSN.

Note: Since control chunks are placed first in a packet and since DATA chunks are transmitted before SHUTDOWN or SHUTDOWN ACK chunks, DATA chunks cannot be bundled with SHUTDOWN or SHUTDOWN ACK chunks.

Partial chunks MUST NOT be placed in an SCTP packet. A partial chunk is a chunk that is not completely contained in the SCTP packet; i.e., the SCTP packet is too short to contain all the bytes of the chunk as indicated by the chunk length.

An endpoint MUST process received chunks in their order in the packet. The receiver uses the Chunk Length field to determine the end of a chunk and beginning of the next chunk taking account of the fact that all chunks end on a 4-byte boundary. If the receiver detects a partial chunk, it MUST drop the chunk.

An endpoint MUST NOT bundle INIT, INIT ACK, or SHUTDOWN COMPLETE chunks with any other chunks.

7. Congestion Control

Congestion control is one of the basic functions in SCTP. To manage congestion, the mechanisms and algorithms in this section are to be employed.

Implementation Note: As far as its specific performance requirements are met, an implementation is always allowed to adopt a more conservative congestion control algorithm than the one defined below.

The congestion control algorithms used by SCTP are based on [RFC5681]. This section describes how the algorithms defined in [RFC5681] are adapted for use in SCTP. We first list differences in protocol designs between TCP and SCTP, and then describe SCTP's congestion control scheme. The description will use the same terminology as in TCP congestion control whenever appropriate.

SCTP congestion control is always applied to the entire association, and not to individual streams.

7.1. SCTP Differences from TCP Congestion Control

Gap Ack Blocks in the SCTP SACK chunk carry the same semantic meaning as the TCP SACK. TCP considers the information carried in the SACK as advisory information only. SCTP considers the information carried in the Gap Ack Blocks in the SACK chunk as advisory. In SCTP, any DATA chunk that has been acknowledged by a SACK chunk, including DATA that arrived at the receiving end out of order, is not considered fully delivered until the Cumulative TSN Ack Point passes the TSN of the DATA chunk (i.e., the DATA chunk has been acknowledged by the Cumulative TSN Ack field in the SACK chunk). Consequently, the value of cwnd controls the amount of outstanding data, rather than (as in the case of non-SACK TCP) the upper bound between the highest acknowledged sequence number and the latest DATA chunk that can be sent within the congestion window. SCTP SACK leads to different implementations of Fast Retransmit and Fast Recovery than non-SACK TCP. As an example, see [FALL96].

The biggest difference between SCTP and TCP, however, is multi-homing. SCTP is designed to establish robust communication associations between two endpoints each of which might be reachable by more than one transport address. Potentially different addresses might lead to different data paths between the two endpoints; thus, ideally one needs a separate set of congestion control parameters for each of the paths. The treatment here of congestion control for multi-homed receivers is new with SCTP and might require refinement in the future. The current algorithms make the following assumptions:

- * The sender usually uses the same destination address until being instructed by the upper layer to do otherwise; however, SCTP MAY change to an alternate destination in the event an address is marked inactive (see Section 8.2). Also, SCTP MAY retransmit to a different transport address than the original transmission.
- * The sender keeps a separate congestion control parameter set for each of the destination addresses it can send to (not each source-destination pair but for each destination). The parameters SHOULD decay if the address is not used for a long enough time period. [RFC5681] specifies this period of time as a retransmission timeout.
- * For each of the destination addresses, an endpoint does slow start upon the first transmission to that address.

Note: TCP guarantees in-sequence delivery of data to its upper-layer protocol within a single TCP session. This means that when TCP notices a gap in the received sequence number, it waits until the gap

is filled before delivering the data that was received with sequence numbers higher than that of the missing data. On the other hand, SCTP can deliver data to its upper-layer protocol even if there is a gap in TSN if the Stream Sequence Numbers are in sequence for a particular stream (i.e., the missing DATA chunks are for a different stream) or if unordered delivery is indicated. Although this does not affect cwnd, it might affect rwnd calculation.

7.2. SCTP Slow-Start and Congestion Avoidance

The slow-start and congestion avoidance algorithms MUST be used by an endpoint to control the amount of data being injected into the network. The congestion control in SCTP is employed in regard to the association, not to an individual stream. In some situations, it might be beneficial for an SCTP sender to be more conservative than the algorithms allow; however, an SCTP sender MUST NOT be more aggressive than the following algorithms allow.

Like TCP, an SCTP endpoint uses the following three control variables to regulate its transmission rate.

- * Receiver advertised window size (rwnd, in bytes), which is set by the receiver based on its available buffer space for incoming packets.

Note: This variable is kept on the entire association.

- * Congestion control window (cwnd, in bytes), which is adjusted by the sender based on observed network conditions.

Note: This variable is maintained on a per-destination-address basis.

- * Slow-start threshold (ssthresh, in bytes), which is used by the sender to distinguish slow-start and congestion avoidance phases.

Note: This variable is maintained on a per-destination-address basis.

SCTP also requires one additional control variable, `partial_bytes_acked`, which is used during congestion avoidance phase to facilitate cwnd adjustment.

Unlike TCP, an SCTP sender MUST keep a set of these control variables cwnd, ssthresh, and `partial_bytes_acked` for EACH destination address of its peer (when its peer is multi-homed). When calculating one of these variables, the length of the DATA chunk including the padding SHOULD be used.

Only one `rwnd` is kept for the whole association (no matter if the peer is multi-homed or has a single address).

7.2.1. Slow-Start

Beginning data transmission into a network with unknown conditions or after a sufficiently long idle period requires SCTP to probe the network to determine the available capacity. The slow-start algorithm is used for this purpose at the beginning of a transfer, or after repairing loss detected by the retransmission timer.

- * The initial `cwnd` before data transmission MUST be set to $\min(4 * PMDCS, \max(2 * PMDCS, 4404))$ bytes if the peer address is an IPv4 address and to $\min(4 * PMDCS, \max(2 * PMDCS, 4344))$ bytes if the peer address is an IPv6 address.
- * The initial `cwnd` after a retransmission timeout MUST be no more than `PMDCS`, and only one packet is allowed to be in flight until successful acknowledgement.
- * The initial value of `ssthresh` SHOULD be arbitrarily high (e.g., the size of the largest possible advertised window).
- * Whenever `cwnd` is greater than zero, the endpoint is allowed to have `cwnd` bytes of data outstanding on that transport address. A limited overbooking as described in Section 6.1 B) SHOULD be supported.
- * When `cwnd` is less than or equal to `ssthresh`, an SCTP endpoint MUST use the slow-start algorithm to increase `cwnd` only if the current congestion window is being fully utilized, and the data sender is not in Fast Recovery. Only when these two conditions are met can the `cwnd` be increased; otherwise, the `cwnd` MUST NOT be increased. If these conditions are met, then `cwnd` MUST be increased by, at most, the lesser of
 1. the total size of the previously outstanding DATA chunk(s) acknowledged, and
 2. L times the destination's `PMDCS`.

The first upper bound protects against the ACK-Splitting attack outlined in [SAVAGE99]. The positive integer L SHOULD be 1, and MAY be larger than 1. See [RFC3465] for details of choosing L .

In instances where its peer endpoint is multi-homed, if an endpoint receives a SACK chunk that results in updating the cwnd, then it SHOULD update its cwnd (or cwnds) apportioned to the destination addresses to which it transmitted the acknowledged data.

- * While the endpoint does not transmit data on a given transport address, the cwnd of the transport address SHOULD be adjusted to $\max(\text{cwnd} / 2, 4 * \text{PMDCS})$ once per RTT. Before the first cwnd adjustment, the ssthresh of the transport address SHOULD be set to the cwnd.

7.2.2. Congestion Avoidance

When cwnd is greater than ssthresh, cwnd SHOULD be incremented by PMDCS per RTT if the sender has cwnd or more bytes of data outstanding for the corresponding transport address. The basic recommendations for incrementing cwnd during congestion avoidance are as follows:

- * SCTP MAY increment cwnd by PMDCS.
- * SCTP SHOULD increment cwnd by PMDCS once per RTT when the sender has cwnd or more bytes of data outstanding for the corresponding transport address.
- * SCTP MUST NOT increment cwnd by more than PMDCS per RTT.

In practice, an implementation can achieve this goal in the following way:

- * `partial_bytes_acked` is initialized to 0.
- * Whenever cwnd is greater than ssthresh, upon each SACK chunk arrival, increase `partial_bytes_acked` by the total number of bytes (including the chunk header and the padding) of all new DATA chunks acknowledged in that SACK chunk, including chunks acknowledged by the new Cumulative TSN Ack, by Gap Ack Blocks, and by the number of bytes of duplicated chunks reported in Duplicate TSNs.
- * When (1) `partial_bytes_acked` is greater than cwnd and (2) before the arrival of the SACK chunk the sender had less than cwnd bytes of data outstanding (i.e., before the arrival of the SACK chunk, `flightsize` was less than cwnd), reset `partial_bytes_acked` to cwnd.

- * When (1) `partial_bytes_acked` is equal to or greater than `cwnd` and (2) before the arrival of the SACK chunk the sender had `cwnd` or more bytes of data outstanding (i.e., before the arrival of the SACK chunk, `flightsize` was greater than or equal to `cwnd`), `partial_bytes_acked` is reset to `(partial_bytes_acked - cwnd)`. Next, `cwnd` is increased by `PMDCS`.
- * Same as in the slow start, when the sender does not transmit DATA chunks on a given transport address, the `cwnd` of the transport address SHOULD be adjusted to `max(cwnd / 2, 4 * PMDCS)` per RTO.
- * When all of the data transmitted by the sender has been acknowledged by the receiver, `partial_bytes_acked` is initialized to 0.

7.2.3. Congestion Control

Upon detection of packet losses from SACK chunks (see Section 7.2.4), an endpoint SHOULD do the following:

```
ssthresh = max(cwnd / 2, 4 * PMDCS)
cwnd = ssthresh
partial_bytes_acked = 0
```

Basically, a packet loss causes `cwnd` to be cut in half.

When the T3-rtx timer expires on an address, SCTP SHOULD perform slow start by:

```
ssthresh = max(cwnd / 2, 4 * PMDCS)
cwnd = PMDCS
partial_bytes_acked = 0
```

and ensure that no more than one SCTP packet will be in flight for that address until the endpoint receives acknowledgement for successful delivery of data to that address.

7.2.4. Fast Retransmit on Gap Reports

In the absence of data loss, an endpoint performs delayed acknowledgement. However, whenever an endpoint notices a hole in the arriving TSN sequence, it SHOULD start sending a SACK chunk back every time a packet arrives carrying data until the hole is filled.

Whenever an endpoint receives a SACK chunk that indicates that some TSNs are missing, it SHOULD wait for two further miss indications (via subsequent SACK chunks for a total of three missing reports) on the same TSNs before taking action with regard to Fast Retransmit.

Miss indications SHOULD follow the HTNA (Highest TSN Newly Acknowledged) algorithm. For each incoming SACK chunk, miss indications are incremented only for missing TSNs prior to the highest TSN newly acknowledged in the SACK chunk. A newly acknowledged DATA chunk is one not previously acknowledged in a SACK chunk. If an endpoint is in Fast Recovery and a SACK chunk arrives that advances the Cumulative TSN Ack Point, the miss indications are incremented for all TSNs reported missing in the SACK chunk.

When the third consecutive miss indication is received for a TSN(s), the data sender does the following:

- 1) Mark the DATA chunk(s) with three miss indications for retransmission.
- 2) If not in Fast Recovery, adjust the ssthresh and cwnd of the destination address(es) to which the missing DATA chunks were last sent, according to the formula described in Section 7.2.3.
- 3) If not in Fast Recovery, determine how many of the earliest (i.e., lowest TSN) DATA chunks marked for retransmission will fit into a single packet, subject to constraint of the PMTU of the destination transport address to which the packet is being sent. Call this value K. Retransmit those K DATA chunks in a single packet. When a Fast Retransmit is being performed, the sender SHOULD ignore the value of cwnd and SHOULD NOT delay retransmission for this single packet.
- 4) Restart the T3-rtx timer only if the last SACK chunk acknowledged the lowest outstanding TSN number sent to that address, or the endpoint is retransmitting the first outstanding DATA chunk sent to that address.
- 5) Mark the DATA chunk(s) as being fast retransmitted and thus ineligible for a subsequent Fast Retransmit. Those TSNs marked for retransmission due to the Fast-Retransmit algorithm that did not fit in the sent datagram carrying K other TSNs are also marked as ineligible for a subsequent Fast Retransmit. However, as they are marked for retransmission they will be retransmitted later on as soon as cwnd allows.
- 6) If not in Fast Recovery, enter Fast Recovery and mark the highest outstanding TSN as the Fast Recovery exit point. When a SACK chunk acknowledges all TSNs up to and including this exit point, Fast Recovery is exited. While in Fast Recovery, the ssthresh and cwnd SHOULD NOT change for any destinations due to a subsequent Fast Recovery event (i.e., one SHOULD NOT reduce the cwnd further due to a subsequent Fast Retransmit).

Note: Before the above adjustments, if the received SACK chunk also acknowledges new DATA chunks and advances the Cumulative TSN Ack Point, the cwnd adjustment rules defined in Section 7.2.1 and Section 7.2.2 MUST be applied first.

7.2.5. Reinitialization

During the lifetime of an SCTP association events can happen, which result in using the network under unknown new conditions. When detected by an SCTP implementation, the congestion control MUST be reinitialized.

7.2.5.1. Change of Differentiated Services Code Points

SCTP implementations MAY allow an application to configure the Differentiated Services Code Point (DSCP) used for sending packets. If a DSCP change might result in outgoing packets being queued in different queues, the congestion control parameters for all affected destination addresses MUST be reset to their initial values.

7.2.5.2. Change of Routes

SCTP implementations MAY be aware of routing changes affecting packets sent to a destination address. In particular, this includes the selection of a different source address used for sending packets to a destination address. If such a routing change happens, the congestion control parameters for the affected destination addresses MUST be reset to their initial values.

7.3. PMTU Discovery

[RFC8899], [RFC8201], and [RFC1191] specify "Packetization Layer Path MTU Discovery", whereby an endpoint maintains an estimate of PMTU along a given Internet path and refrains from sending packets along that path that exceed the PMTU, other than occasional attempts to probe for a change in the PMTU. [RFC8899] is thorough in its discussion of the PMTU discovery mechanism and strategies for determining the current end-to-end PMTU setting as well as detecting changes in this value.

An endpoint SHOULD apply these techniques, and SHOULD do so on a per-destination-address basis.

There are two important SCTP-specific points regarding PMTU discovery:

- 1) SCTP associations can span multiple addresses. An endpoint **MUST** maintain separate PMTU estimates for each destination address of its peer.
- 2) The sender **SHOULD** track an AMDCS that will be the smallest PMDCS discovered for all of the peer's destination addresses. When fragmenting messages into multiple parts this AMDCS **SHOULD** be used to calculate the size of each DATA chunk. This will allow retransmissions to be seamlessly sent to an alternate address without encountering IP fragmentation.

8. Fault Management

8.1. Endpoint Failure Detection

An endpoint **SHOULD** keep a counter on the total number of consecutive retransmissions to its peer (this includes data retransmissions to all the destination transport addresses of the peer if it is multi-homed), including the number of unacknowledged HEARTBEAT chunks observed on the path that is currently used for data transfer. Unacknowledged HEARTBEAT chunks observed on paths different from the path currently used for data transfer **SHOULD NOT** increment the association error counter, as this could lead to association closure even if the path that is currently used for data transfer is available (but idle). If the value of this counter exceeds the limit indicated in the protocol parameter 'Association.Max.Retrans', the endpoint **SHOULD** consider the peer endpoint unreachable and **SHALL** stop transmitting any more data to it (and thus the association enters the CLOSED state). In addition, the endpoint **SHOULD** report the failure to the upper layer and optionally report back all outstanding user data remaining in its outbound queue. The association is automatically closed when the peer endpoint becomes unreachable.

The counter used for endpoint failure detection **MUST** be reset each time a DATA chunk sent to that peer endpoint is acknowledged (by the reception of a SACK chunk). When a HEARTBEAT ACK chunk is received from the peer endpoint, the counter **SHOULD** also be reset. The receiver of the HEARTBEAT ACK chunk **MAY** choose not to clear the counter if there is outstanding data on the association. This allows for handling the possible difference in reachability based on DATA chunks and HEARTBEAT chunks.

8.2. Path Failure Detection

When its peer endpoint is multi-homed, an endpoint **SHOULD** keep an error counter for each of the destination transport addresses of the peer endpoint.

Each time the T3-rtx timer expires on any address, or when a HEARTBEAT chunk sent to an idle address is not acknowledged within an RTO, the error counter of that destination address will be incremented. When the value in the error counter exceeds the protocol parameter 'Path.Max.Retrans' of that destination address, the endpoint SHOULD mark the destination transport address as inactive, and a notification SHOULD be sent to the upper layer.

When an outstanding TSN is acknowledged or a HEARTBEAT chunk sent to that address is acknowledged with a HEARTBEAT ACK chunk, the endpoint SHOULD clear the error counter of the destination transport address to which the DATA chunk was last sent (or HEARTBEAT chunk was sent) and SHOULD also report to the upper layer when an inactive destination address is marked as active. When the peer endpoint is multi-homed and the last chunk sent to it was a retransmission to an alternate address, there exists an ambiguity as to whether or not the acknowledgement could be credited to the address of the last chunk sent. However, this ambiguity does not seem to have significant consequences for SCTP behavior. If this ambiguity is undesirable, the transmitter MAY choose not to clear the error counter if the last chunk sent was a retransmission.

Note: When configuring the SCTP endpoint, the user ought to avoid having the value of 'Association.Max.Retrans' larger than the summation of the 'Path.Max.Retrans' of all the destination addresses for the remote endpoint. Otherwise, all the destination addresses might become inactive while the endpoint still considers the peer endpoint reachable. When this condition occurs, how SCTP chooses to function is implementation specific.

When the primary path is marked inactive (due to excessive retransmissions, for instance), the sender MAY automatically transmit new packets to an alternate destination address if one exists and is active. If more than one alternate address is active when the primary path is marked inactive, only ONE transport address SHOULD be chosen and used as the new destination transport address.

8.3. Path Heartbeat

By default, an SCTP endpoint SHOULD monitor the reachability of the idle destination transport address(es) of its peer by sending a HEARTBEAT chunk periodically to the destination transport address(es). The sending of HEARTBEAT chunks MAY begin upon reaching the ESTABLISHED state and is discontinued after sending either a SHUTDOWN chunk or SHUTDOWN ACK chunk. A receiver of a HEARTBEAT chunks MUST respond to a HEARTBEAT chunk with a HEARTBEAT ACK chunk after entering the COOKIE-ECHOED state (sender of the INIT chunk) or the ESTABLISHED state (receiver of the INIT chunk), up until reaching

the SHUTDOWN-SENT state (sender of the SHUTDOWN chunk) or the SHUTDOWN-ACK-SENT state (receiver of the SHUTDOWN chunk).

A destination transport address is considered "idle" if no new chunk that can be used for updating path RTT (usually including first transmission DATA, INIT, COOKIE ECHO, or HEARTBEAT chunks, etc.) and no HEARTBEAT chunk has been sent to it within the current heartbeat period of that address. This applies to both active and inactive destination addresses.

The upper layer can optionally initiate the following functions:

- A) Disable heartbeat on a specific destination transport address of a given association,
- B) Change the 'HB.interval',
- C) Re-enable heartbeat on a specific destination transport address of a given association, and
- D) Request the sending of an on-demand HEARTBEAT chunk on a specific destination transport address of a given association.

The endpoint SHOULD increment the respective error counter of the destination transport address each time a HEARTBEAT chunk is sent to that address and not acknowledged within one RTO.

When the value of this counter exceeds the protocol parameter 'Path.Max.Retrans', the endpoint SHOULD mark the corresponding destination address as inactive if it is not so marked and SHOULD also report to the upper layer the change in reachability of this destination address. After this, the endpoint SHOULD continue sending HEARTBEAT chunks on this destination address but SHOULD stop increasing the counter.

The sender of the HEARTBEAT chunk SHOULD include in the Heartbeat Information field of the chunk the current time when the packet is sent and the destination address to which the packet is sent.

Implementation Note: An alternative implementation of the heartbeat mechanism that can be used is to increment the error counter variable every time a HEARTBEAT chunk is sent to a destination. Whenever a HEARTBEAT ACK chunk arrives, the sender SHOULD clear the error counter of the destination that the HEARTBEAT chunk was sent to. This in effect would clear the previously stroked error (and any other error counts as well).

The receiver of the HEARTBEAT chunk SHOULD immediately respond with a HEARTBEAT ACK chunk that contains the Heartbeat Information TLV, together with any other received TLVs, copied unchanged from the received HEARTBEAT chunk.

Upon the receipt of the HEARTBEAT ACK chunk, the sender of the HEARTBEAT chunk SHOULD clear the error counter of the destination transport address to which the HEARTBEAT chunk was sent and mark the destination transport address as active if it is not so marked. The endpoint SHOULD report to the upper layer when an inactive destination address is marked as active due to the reception of the latest HEARTBEAT ACK chunk. The receiver of the HEARTBEAT ACK chunk SHOULD also clear the association overall error count (as defined in Section 8.1).

The receiver of the HEARTBEAT ACK chunk SHOULD also perform an RTT measurement for that destination transport address using the time value carried in the HEARTBEAT ACK chunk.

On an idle destination address that is allowed to heartbeat, it is RECOMMENDED that a HEARTBEAT chunk is sent once per RTO of that destination address plus the protocol parameter 'HB.interval', with jittering of +/- 50% of the RTO value, and exponential backoff of the RTO if the previous HEARTBEAT chunk is unanswered.

A primitive is provided for the SCTP user to change the 'HB.interval' and turn on or off the heartbeat on a given destination address. The 'HB.interval' set by the SCTP user is added to the RTO of that destination (including any exponential backoff). Only one heartbeat SHOULD be sent each time the heartbeat timer expires (if multiple destinations are idle). It is an implementation decision on how to choose which of the candidate idle destinations to heartbeat to (if more than one destination is idle).

When tuning the 'HB.interval', there is a side effect that SHOULD be taken into account. When this value is increased, i.e., the time between the sending of HEARTBEAT chunks is longer, the detection of lost ABORT chunks takes longer as well. If a peer endpoint sends an ABORT chunk for any reason and the ABORT chunk is lost, the local endpoint will only discover the lost ABORT chunk by sending a DATA chunk or HEARTBEAT chunk (thus causing the peer to send another ABORT chunk). This is to be considered when tuning the heartbeat timer. If the sending of HEARTBEAT chunks is disabled, only sending DATA chunks to the association will discover a lost ABORT chunk from the peer.

8.4. Handle "Out of the Blue" Packets

An SCTP packet is called an "out of the blue" (OOTB) packet if it is correctly formed (i.e., passed the receiver's CRC32c check; see Section 6.8), but the receiver is not able to identify the association to which this packet belongs.

The receiver of an OOTB packet does the following:

- 1) If the OOTB packet is to or from a non-unicast address, a receiver SHOULD silently discard the packet. Otherwise,
- 2) If the OOTB packet contains an ABORT chunk, the receiver MUST silently discard the OOTB packet and take no further action. Otherwise,
- 3) If the packet contains an INIT chunk with a Verification Tag set to 0, it SHOULD be processed as described in Section 5.1. If, for whatever reason, the INIT chunk cannot be processed normally and an ABORT chunk has to be sent in response, the Verification Tag of the packet containing the ABORT chunk MUST be the Initiate Tag of the received INIT chunk, and the T bit of the ABORT chunk has to be set to 0, indicating that the Verification Tag is not reflected. Otherwise,
- 4) If the packet contains a COOKIE ECHO chunk as the first chunk, it MUST be processed as described in Section 5.1. Otherwise,
- 5) If the packet contains a SHUTDOWN ACK chunk, the receiver SHOULD respond to the sender of the OOTB packet with a SHUTDOWN COMPLETE chunk. When sending the SHUTDOWN COMPLETE chunk, the receiver of the OOTB packet MUST fill in the Verification Tag field of the outbound packet with the Verification Tag received in the SHUTDOWN ACK chunk and set the T bit in the Chunk Flags to indicate that the Verification Tag is reflected. Otherwise,
- 6) If the packet contains a SHUTDOWN COMPLETE chunk, the receiver SHOULD silently discard the packet and take no further action. Otherwise,
- 7) If the packet contains a ERROR chunk with the "Stale Cookie" error cause or a COOKIE ACK chunk, the SCTP packet SHOULD be silently discarded. Otherwise,
- 8) The receiver SHOULD respond to the sender of the OOTB packet with an ABORT chunk. When sending the ABORT chunk, the receiver of the OOTB packet MUST fill in the Verification Tag field of the outbound packet with the value found in the Verification Tag

field of the OOTB packet and set the T bit in the Chunk Flags to indicate that the Verification Tag is reflected. After sending this ABORT chunk, the receiver of the OOTB packet MUST discard the OOTB packet and MUST NOT take any further action.

8.5. Verification Tag

The Verification Tag rules defined in this section apply when sending or receiving SCTP packets that do not contain an INIT, SHUTDOWN COMPLETE, COOKIE ECHO (see Section 5.1), ABORT, or SHUTDOWN ACK chunk. The rules for sending and receiving SCTP packets containing one of these chunk types are discussed separately in Section 8.5.1.

When sending an SCTP packet, the endpoint MUST fill in the Verification Tag field of the outbound packet with the tag value in the Initiate Tag parameter of the INIT or INIT ACK chunk received from its peer.

When receiving an SCTP packet, the endpoint MUST ensure that the value in the Verification Tag field of the received SCTP packet matches its own tag. If the received Verification Tag value does not match the receiver's own tag value, the receiver MUST silently discard the packet and MUST NOT process it any further except for those cases listed in Section 8.5.1 below.

8.5.1. Exceptions in Verification Tag Rules

A) Rules for packets carrying an INIT chunk:

- * The sender MUST set the Verification Tag of the packet to 0.
- * When an endpoint receives an SCTP packet with the Verification Tag set to 0, it SHOULD verify that the packet contains only an INIT chunk. Otherwise, the receiver MUST silently discard the packet.

B) Rules for packets carrying an ABORT chunk:

- * The endpoint MUST always fill in the Verification Tag field of the outbound packet with the destination endpoint's tag value, if it is known.
- * If the ABORT chunk is sent in response to an OOTB packet, the endpoint MUST follow the procedure described in Section 8.4.
- * The receiver of an ABORT chunk MUST accept the packet if the Verification Tag field of the packet matches its own tag and the T bit is not set OR if it is set to its peer's tag and the T bit is set in the Chunk Flags. Otherwise, the receiver MUST silently discard the packet and take no further action.

C) Rules for packets carrying a SHUTDOWN COMPLETE chunk:

- * When sending a SHUTDOWN COMPLETE chunk, if the receiver of the SHUTDOWN ACK chunk has a TCB, then the destination endpoint's tag MUST be used, and the T bit MUST NOT be set. Only where no TCB exists SHOULD the sender use the Verification Tag from the SHUTDOWN ACK chunk, and MUST set the T bit.
- * The receiver of a SHUTDOWN COMPLETE chunk accepts the packet if the Verification Tag field of the packet matches its own tag and the T bit is not set OR if it is set to its peer's tag and the T bit is set in the Chunk Flags. Otherwise, the receiver MUST silently discard the packet and take no further action. An endpoint MUST ignore the SHUTDOWN COMPLETE chunk if it is not in the SHUTDOWN-ACK-SENT state.

D) Rules for packets carrying a COOKIE ECHO chunk:

- * When sending a COOKIE ECHO chunk, the endpoint MUST use the value of the Initiate Tag received in the INIT ACK chunk.
- * The receiver of a COOKIE ECHO chunk follows the procedures in Section 5.

E) Rules for packets carrying a SHUTDOWN ACK chunk:

- * If the receiver is in COOKIE-ECHOED or COOKIE-WAIT state the procedures in Section 8.4 SHOULD be followed; in other words, it is treated as an OOTB packet.

9. Termination of Association

An endpoint SHOULD terminate its association when it exits from service. An association can be terminated by either abort or shutdown. An abort of an association is abortive by definition in that any data pending on either end of the association is discarded and not delivered to the peer. A shutdown of an association is considered a graceful close where all data in queue by either endpoint is delivered to the respective peers. However, in the case of a shutdown, SCTP does not support a half-open state (like TCP) wherein one side might continue sending data while the other end is closed. When either endpoint performs a shutdown, the association on each peer will stop accepting new data from its user and only deliver data in queue at the time of sending or receiving the SHUTDOWN chunk.

9.1. Abort of an Association

When an endpoint decides to abort an existing association, it MUST send an ABORT chunk to its peer endpoint. The sender MUST fill in the peer's Verification Tag in the outbound packet and MUST NOT bundle any DATA chunk with the ABORT chunk. If the association is aborted on request of the upper layer, a "User-Initiated Abort" error cause (see Section 3.3.10.12) SHOULD be present in the ABORT chunk.

An endpoint MUST NOT respond to any received packet that contains an ABORT chunk (also see Section 8.4).

An endpoint receiving an ABORT chunk MUST apply the special Verification Tag check rules described in Section 8.5.1.

After checking the Verification Tag, the receiving endpoint MUST remove the association from its record and SHOULD report the termination to its upper layer. If a "User-Initiated Abort" error cause is present in the ABORT chunk, the Upper Layer Abort Reason SHOULD be made available to the upper layer.

9.2. Shutdown of an Association

Using the SHUTDOWN primitive (see Section 11.1), the upper layer of an endpoint in an association can gracefully close the association. This will allow all outstanding DATA chunks from the peer of the shutdown initiator to be delivered before the association terminates.

Upon receipt of the SHUTDOWN primitive from its upper layer, the endpoint enters the SHUTDOWN-PENDING state and remains there until all outstanding data has been acknowledged by its peer. The endpoint accepts no new data from its upper layer, but retransmits data to the peer endpoint if necessary to fill gaps.

Once all its outstanding data has been acknowledged, the endpoint sends a SHUTDOWN chunk to its peer including in the Cumulative TSN Ack field the last sequential TSN it has received from the peer. It SHOULD then start the T2-shutdown timer and enter the SHUTDOWN-SENT state. If the timer expires, the endpoint MUST resend the SHUTDOWN chunk with the updated last sequential TSN received from its peer.

The rules in Section 6.3 MUST be followed to determine the proper timer value for T2-shutdown. To indicate any gaps in TSN, the endpoint MAY also bundle a SACK chunk with the SHUTDOWN chunk in the same SCTP packet.

An endpoint SHOULD limit the number of retransmissions of the SHUTDOWN chunk to the protocol parameter 'Association.Max.Retrans'. If this threshold is exceeded, the endpoint SHOULD destroy the TCB and SHOULD report the peer endpoint unreachable to the upper layer (and thus the association enters the CLOSED state). The reception of any packet from its peer (i.e., as the peer sends all of its queued DATA chunks) SHOULD clear the endpoint's retransmission count and restart the T2-shutdown timer, giving its peer ample opportunity to transmit all of its queued DATA chunks that have not yet been sent.

Upon reception of the SHUTDOWN chunk, the peer endpoint does the following:

- * enter the SHUTDOWN-RECEIVED state,
- * stop accepting new data from its SCTP user, and
- * verify, by checking the Cumulative TSN Ack field of the chunk, that all its outstanding DATA chunks have been received by the SHUTDOWN chunk sender.

Once an endpoint has reached the SHUTDOWN-RECEIVED state, it MUST ignore ULP shutdown requests but MUST continue responding to SHUTDOWN chunks from its peer.

If there are still outstanding DATA chunks left, the SHUTDOWN chunk receiver MUST continue to follow normal data transmission procedures defined in Section 6, until all outstanding DATA chunks are acknowledged; however, the SHUTDOWN chunk receiver MUST NOT accept new data from its SCTP user.

While in the SHUTDOWN-SENT state, the SHUTDOWN chunk sender MUST immediately respond to each received packet containing one or more DATA chunks with a SHUTDOWN chunk and restart the T2-shutdown timer. If a SHUTDOWN chunk by itself cannot acknowledge all of the received DATA chunks (i.e., there are TSNs that can be acknowledged that are larger than the cumulative TSN, and thus gaps exist in the TSN sequence), or if duplicate TSNs have been received, then a SACK chunk MUST also be sent.

The sender of the SHUTDOWN chunk MAY also start an overall guard timer T5-shutdown-guard to bound the overall time for the shutdown sequence. At the expiration of this timer, the sender SHOULD abort the association by sending an ABORT chunk. If the T5-shutdown-guard timer is used, it SHOULD be set to the RECOMMENDED value of 5 times 'RTO.Max'.

If the receiver of the SHUTDOWN chunk has no more outstanding DATA chunks, the SHUTDOWN chunk receiver MUST send a SHUTDOWN ACK chunk and start a T2-shutdown timer of its own, entering the SHUTDOWN-ACK-SENT state. If the timer expires, the endpoint MUST resend the SHUTDOWN ACK chunk.

The sender of the SHUTDOWN ACK chunk SHOULD limit the number of retransmissions of the SHUTDOWN ACK chunk to the protocol parameter 'Association.Max.Retrans'. If this threshold is exceeded, the endpoint SHOULD destroy the TCB and SHOULD report the peer endpoint unreachable to the upper layer (and thus the association enters the CLOSED state).

Upon the receipt of the SHUTDOWN ACK chunk, the sender of the SHUTDOWN chunk MUST stop the T2-shutdown timer, send a SHUTDOWN COMPLETE chunk to its peer, and remove all record of the association.

Upon reception of the SHUTDOWN COMPLETE chunk, the endpoint verifies that it is in the SHUTDOWN-ACK-SENT state; if it is not, the chunk SHOULD be discarded. If the endpoint is in the SHUTDOWN-ACK-SENT state, the endpoint SHOULD stop the T2-shutdown timer and remove all knowledge of the association (and thus the association enters the CLOSED state).

An endpoint SHOULD ensure that all its outstanding DATA chunks have been acknowledged before initiating the shutdown procedure.

An endpoint SHOULD reject any new data request from its upper layer if it is in the SHUTDOWN-PENDING, SHUTDOWN-SENT, SHUTDOWN-RECEIVED, or SHUTDOWN-ACK-SENT state.

If an endpoint is in the SHUTDOWN-ACK-SENT state and receives an INIT chunk (e.g., if the SHUTDOWN COMPLETE chunk was lost) with source and destination transport addresses (either in the IP addresses or in the INIT chunk) that belong to this association, it SHOULD discard the INIT chunk and retransmit the SHUTDOWN ACK chunk.

Note: Receipt of a packet containing an INIT chunk with the same source and destination IP addresses as used in transport addresses assigned to an endpoint but with a different port number indicates the initialization of a separate association.

The sender of the INIT or COOKIE ECHO chunk SHOULD respond to the receipt of a SHUTDOWN ACK chunk with a stand-alone SHUTDOWN COMPLETE chunk in an SCTP packet with the Verification Tag field of its common header set to the same tag that was received in the packet containing the SHUTDOWN ACK chunk. This is considered an OOTB packet as defined in Section 8.4. The sender of the INIT chunk lets T1-init continue

running and remains in the COOKIE-WAIT or COOKIE-ECHOED state. Normal T1-init timer expiration will cause the INIT or COOKIE chunk to be retransmitted and thus start a new association.

If a SHUTDOWN chunk is received in the COOKIE-WAIT or COOKIE ECHOED state, the SHUTDOWN chunk SHOULD be silently discarded.

If an endpoint is in the SHUTDOWN-SENT state and receives a SHUTDOWN chunk from its peer, the endpoint SHOULD respond immediately with a SHUTDOWN ACK chunk to its peer, and move into the SHUTDOWN-ACK-SENT state restarting its T2-shutdown timer.

If an endpoint is in the SHUTDOWN-ACK-SENT state and receives a SHUTDOWN ACK, it MUST stop the T2-shutdown timer, send a SHUTDOWN COMPLETE chunk to its peer, and remove all record of the association.

10. ICMP Handling

Whenever an ICMP message is received by an SCTP endpoint, the following procedures MUST be followed to ensure proper utilization of the information being provided by layer 3.

- ICMP1) An implementation MAY ignore all ICMPv4 messages where the type field is not set to "Destination Unreachable".
- ICMP2) An implementation MAY ignore all ICMPv6 messages where the type field is not "Destination Unreachable", "Parameter Problem", or "Packet Too Big".
- ICMP3) An implementation SHOULD ignore any ICMP messages where the code indicates "Port Unreachable".
- ICMP4) An implementation MAY ignore all ICMPv6 messages of type "Parameter Problem" if the code is not "Unrecognized Next Header Type Encountered".
- ICMP5) An implementation MUST use the payload of the ICMP message (v4 or v6) to locate the association that sent the message to which ICMP is responding. If the association cannot be found, an implementation SHOULD ignore the ICMP message.

- ICMP6) An implementation MUST validate that the Verification Tag contained in the ICMP message matches the Verification Tag of the peer. If the Verification Tag is not 0 and does not match, discard the ICMP message. If it is 0 and the ICMP message contains enough bytes to verify that the chunk type is an INIT chunk and that the Initiate Tag matches the tag of the peer, continue with ICMP7. If the ICMP message is too short or the chunk type or the Initiate Tag does not match, silently discard the packet.
- ICMP7) If the ICMP message is either an ICMPv6 message of type "Packet Too Big" or an ICMPv4 message of type "Destination Unreachable" and code "Fragmentation Needed", an implementation SHOULD process this information as defined for PMTU discovery.
- ICMP8) If the ICMP code is an "Unrecognized Next Header Type Encountered" or a "Protocol Unreachable", an implementation MUST treat this message as an abort with the T bit set if it does not contain an INIT chunk. If it does contain an INIT chunk and the association is in the COOKIE-WAIT state, handle the ICMP message like an ABORT chunk.
- ICMP9) If the ICMP type is "Destination Unreachable", the implementation MAY move the destination to the unreachable state or, alternatively, increment the path error counter. SCTP MAY provide information to the upper layer indicating the reception of ICMP messages when reporting a network status change.

These procedures differ from [RFC1122] and from its requirements for processing of port-unreachable messages and the requirements that an implementation MUST abort associations in response to a "protocol unreachable" message. Port-unreachable messages are not processed, since an implementation will send an ABORT chunk, not a port unreachable. The stricter handling of the "protocol unreachable" message is due to security concerns for hosts that do not support SCTP.

11. Interface with Upper Layer

The Upper Layer Protocols (ULPs) request services by passing primitives to SCTP and receive notifications from SCTP for various events.

The primitives and notifications described in this section can be used as a guideline for implementing SCTP. The following functional description of ULP interface primitives is shown for illustrative

purposes. Different SCTP implementations can have different ULP interfaces. However, all SCTP implementations are expected to provide a certain minimum set of services to guarantee that all SCTP implementations can support the same protocol hierarchy.

Please note that this section is informational only.

[RFC6458] and the Socket API Considerations section of [RFC7053] define an extension of the socket API for SCTP as described in this document.

11.1. ULP-to-SCTP

The following sections functionally characterize a ULP/SCTP interface. The notation used is similar to most procedure or function calls in high-level languages.

The ULP primitives described below specify the basic functions that SCTP performs to support inter-process communication. Individual implementations define their own exact format, and provide combinations or subsets of the basic functions in single calls.

11.1.1. Initialize

```
INITIALIZE ([local port],[local eligible address list])  
-> local SCTP instance name
```

This primitive allows SCTP to initialize its internal data structures and allocate necessary resources for setting up its operation environment. Once SCTP is initialized, ULP can communicate directly with other endpoints without re-invoking this primitive.

SCTP will return a local SCTP instance name to the ULP.

Mandatory attributes:
None.

Optional attributes:
local port: SCTP port number, if ULP wants it to be specified.
local eligible address list: an address list that the local SCTP endpoint binds. By default, if an address list is not included, all IP addresses assigned to the host are used by the local endpoint.

Implementation Note: If this optional attribute is supported by an implementation, it will be the responsibility of the implementation to enforce that the IP source address field of any SCTP packets sent by this endpoint contains one of the IP addresses indicated in the local eligible address list.

11.1.2. Associate

```
ASSOCIATE(local SCTP instance name,  
initial destination transport addr list, outbound stream count)  
-> association id [,destination transport addr list]  
[,outbound stream count]
```

This primitive allows the upper layer to initiate an association to a specific peer endpoint.

The peer endpoint is specified by one or more of the transport addresses that defines the endpoint (see Section 2.3). If the local SCTP instance has not been initialized, the ASSOCIATE is considered an error.

An association id, which is a local handle to the SCTP association, will be returned on successful establishment of the association. If SCTP is not able to open an SCTP association with the peer endpoint, an error is returned.

Other association parameters can be returned, including the complete destination transport addresses of the peer as well as the outbound stream count of the local endpoint. One of the transport addresses from the returned destination addresses will be selected by the local endpoint as default primary path for sending SCTP packets to this peer. The returned "destination transport addr list" can be used by the ULP to change the default primary path or to force sending a packet to a specific transport address.

Implementation Note: If ASSOCIATE primitive is implemented as a blocking function call, the ASSOCIATE primitive can return association parameters in addition to the association id upon successful establishment. If ASSOCIATE primitive is implemented as a non-blocking call, only the association id is returned and association parameters are passed using the COMMUNICATION UP notification.

Mandatory attributes:

local SCTP instance name: obtained from the INITIALIZE operation.

initial destination transport addr list: a non-empty list of

transport addresses of the peer endpoint with which the association is to be established.

outbound stream count: the number of outbound streams the ULP would like to open towards this peer endpoint.

Optional attributes:
None.

11.1.3. Shutdown

SHUTDOWN(association id) -> result

Gracefully closes an association. Any locally queued user data will be delivered to the peer. The association will be terminated only after the peer acknowledges all the SCTP packets sent. A success code will be returned on successful termination of the association. If attempting to terminate the association results in a failure, an error code is returned.

Mandatory attributes:
association id: local handle to the SCTP association.

Optional attributes:
None.

11.1.4. Abort

ABORT(association id [, Upper Layer Abort Reason]) -> result

Ungracefully closes an association. Any locally queued user data will be discarded, and an ABORT chunk is sent to the peer. A success code will be returned on successful abort of the association. If attempting to abort the association results in a failure, an error code is returned.

Mandatory attributes:
association id: local handle to the SCTP association.

Optional attributes:
Upper Layer Abort Reason: reason of the abort to be passed to the peer.

11.1.5. Send

```
SEND(association id, buffer address, byte count [,context]  
[,stream id] [,life time] [,destination transport address]  
[,unordered flag] [,no-bundle flag] [,payload protocol-id]  
[,sack-immediately flag]) -> result
```

This is the main method to send user data via SCTP.

Mandatory attributes:

association id: local handle to the SCTP association.

buffer address: the location where the user message to be transmitted is stored.

byte count: the size of the user data in number of bytes.

Optional attributes:

context: an optional information provided that will be carried in the sending failure notification to the ULP if the transportation of this user message fails.

stream id: to indicate which stream to send the data on. If not specified, stream 0 will be used.

life time: specifies the life time of the user data. The user data will not be sent by SCTP after the life time expires. This parameter can be used to avoid efforts to transmit stale user messages. SCTP notifies the ULP if the data cannot be initiated to transport (i.e., sent to the destination via SCTP's SEND primitive) within the life time variable. However, the user data will be transmitted if SCTP has attempted to transmit a chunk before the life time expired.

Implementation Note: In order to better support the data life time option, the transmitter can hold back the assigning of the TSN number to an outbound DATA chunk to the last moment. And, for implementation simplicity, once a TSN number has been assigned the sender considers the send of this DATA chunk as committed, overriding any life time option attached to the DATA chunk.

destination transport address: specified as one of the destination transport addresses of the peer endpoint to which this packet is sent. Whenever possible, SCTP uses this destination transport address for sending the packets, instead of the current primary path.

unordered flag: this flag, if present, indicates that the user

would like the data delivered in an unordered fashion to the peer (i.e., the U flag is set to 1 on all DATA chunks carrying this message).

no-bundle flag: instructs SCTP not to delay the sending of DATA chunks for this user data just to allow it to be bundled with other outbound DATA chunks. When faced with network congestion, SCTP might still bundle the data, even when this flag is present.

payload protocol-id: a 32-bit unsigned integer that is to be passed to the peer indicating the type of payload protocol data being transmitted. Note that the upper layer is responsible for the host to network byte order conversion of this field, which is passed by SCTP as 4 bytes of opaque data.

sack-immediately flag: set the I bit on the last DATA chunk used for the user message to be transmitted.

11.1.6. Set Primary

SETPRIMARY(association id, destination transport address,
[source transport address]) -> result

Instructs the local SCTP to use the specified destination transport address as the primary path for sending packets.

The result of attempting this operation is returned. If the specified destination transport address is not present in the "destination transport address list" returned earlier in an associate command or communication up notification, an error is returned.

Mandatory attributes:

association id: local handle to the SCTP association.

destination transport address: specified as one of the transport addresses of the peer endpoint, which is used as the primary address for sending packets. This overrides the current primary address information maintained by the local SCTP endpoint.

Optional attributes:

source transport address: optionally, some implementations can allow you to set the default source address placed in all outgoing IP datagrams.

11.1.7. Receive


```
RECEIVE(association id, buffer address, buffer size [,stream id])  
-> byte count [,transport address] [,stream id]  
[,stream sequence number] [,partial flag] [,payload protocol-id]
```

This primitive reads the first user message in the SCTP in-queue into the buffer specified by ULP, if there is one available. The size of the message read, in bytes, will be returned. It might, depending on the specific implementation, also return other information such as the sender's address, the stream id on which it is received, whether there are more messages available for retrieval, etc. For ordered messages, their Stream Sequence Number might also be returned.

Depending upon the implementation, if this primitive is invoked when no message is available the implementation returns an indication of this condition or blocks the invoking process until data does become available.

Mandatory attributes:

association id: local handle to the SCTP association

buffer address: the memory location indicated by the ULP to store the received message.

buffer size: the maximum size of data to be received, in bytes.

Optional attributes:

stream id: to indicate which stream to receive the data on.

stream sequence number: the Stream Sequence Number assigned by the sending SCTP peer.

partial flag: if this returned flag is set to 1, then this primitive contains a partial delivery of the whole message. When this flag is set, the stream id and stream sequence number accompanies this primitive. When this flag is set to 0, it indicates that no more deliveries will be received for this stream sequence number.

payload protocol-id: a 32-bit unsigned integer that is received from the peer indicating the type of payload protocol of the received data. Note that the upper layer is responsible for the host to network byte order conversion of this field, which is passed by SCTP as 4 bytes of opaque data.

11.1.8. Status

```
STATUS(association id) -> status data
```

This primitive returns a data block containing the following information:

- * association connection state,
- * destination transport address list,
- * destination transport address reachability states,
- * current receiver window size,
- * current congestion window sizes,
- * number of unacknowledged DATA chunks,
- * number of DATA chunks pending receipt,
- * primary path,
- * most recent SRTT on primary path,
- * RTO on primary path,
- * SRTT and RTO on other destination addresses, etc.

Mandatory attributes:

association id: local handle to the SCTP association.

Optional attributes:

None.

11.1.9. Change Heartbeat

CHANGE HEARTBEAT(association id, destination transport address, new state [,interval]) -> result

Instructs the local endpoint to enable or disable heartbeat on the specified destination transport address.

The result of attempting this operation is returned.

Note: Even when enabled, heartbeat will not take place if the destination transport address is not idle.

Mandatory attributes:

association id: local handle to the SCTP association.

destination transport address: specified as one of the transport

addresses of the peer endpoint.

new state: the new state of heartbeat for this destination transport address (either enabled or disabled).

Optional attributes:

interval: if present, indicates the frequency of the heartbeat if this is to enable heartbeat on a destination transport address. This value is added to the RTO of the destination transport address. This value, if present, affects all destinations.

11.1.10. Request Heartbeat

REQUESTHEARTBEAT(association id, destination transport address)
-> result

Instructs the local endpoint to perform a heartbeat on the specified destination transport address of the given association. The returned result indicates whether the transmission of the HEARTBEAT chunk to the destination address is successful.

Mandatory attributes:

association id: local handle to the SCTP association.

destination transport address: the transport address of the association on which a heartbeat is issued.

Optional attributes:

None.

11.1.11. Get SRTT Report

GETSRTTREPORT(association id, destination transport address)
-> srtt result

Instructs the local SCTP to report the current SRTT measurement on the specified destination transport address of the given association. The returned result can be an integer containing the most recent SRTT in milliseconds.

Mandatory attributes:

association id: local handle to the SCTP association.

destination transport address: the transport address of the association on which the SRTT measurement is to be reported.

Optional attributes:

None.

11.1.12. Set Failure Threshold

```
SETFAILURETHRESHOLD(association id, destination transport address,  
failure threshold) -> result
```

This primitive allows the local SCTP to customize the reachability failure detection threshold 'Path.Max.Retrans' for the specified destination address. Note that this can also be done using the SETPROTOCOLPARAMETERS primitive (Section 11.1.13).

Mandatory attributes:

association id: local handle to the SCTP association.

destination transport address: the transport address of the association on which the failure detection threshold is to be set.

failure threshold: the new value of 'Path.Max.Retrans' for the destination address.

Optional attributes:

None.

11.1.13. Set Protocol Parameters

```
SETPROTOCOLPARAMETERS(association id,  
[destination transport address,] protocol parameter list)  
-> result
```

This primitive allows the local SCTP to customize the protocol parameters.

Mandatory attributes:

association id: local handle to the SCTP association.

protocol parameter list: the specific names and values of the protocol parameters (e.g., 'Association.Max.Retrans' (see Section 16), or other parameters like the DSCP) that the SCTP user wishes to customize.

Optional attributes:

destination transport address: some of the protocol parameters might be set on a per destination transport address basis.

11.1.14. Receive Unsent Message

```
RECEIVE_UNSENT(data retrieval id, buffer address, buffer size  
[,stream id] [, stream sequence number] [,partial flag]  
[,payload protocol-id])
```

This primitive reads a user message, which has never been sent, into the buffer specified by ULP.

Mandatory attributes:

data retrieval id: the identification passed to the ULP in the failure notification.

buffer address: the memory location indicated by the ULP to store the received message.

buffer size: the maximum size of data to be received, in bytes.

Optional attributes:

stream id: this is a return value that is set to indicate which stream the data was sent to.

stream sequence number: this value is returned indicating the Stream Sequence Number that was associated with the message.

partial flag: if this returned flag is set to 1, then this message is a partial delivery of the whole message. When this flag is set, the stream id and stream sequence number accompanies this primitive. When this flag is set to 0, it indicates that no more deliveries will be received for this stream sequence number.

payload protocol-id: The 32 bit unsigned integer that was set to be sent to the peer indicating the type of payload protocol of the received data.

11.1.15. Receive Unacknowledged Message

```
RECEIVE_UNACKED(data retrieval id, buffer address, buffer size,  
[,stream id] [,stream sequence number] [,partial flag]  
[,payload protocol-id])
```

This primitive reads a user message, which has been sent and has not been acknowledged by the peer, into the buffer specified by ULP.

Mandatory attributes:

data retrieval id: the identification passed to the ULP in the failure notification.

buffer address: the memory location indicated by the ULP to store

the received message.

buffer size: the maximum size of data to be received, in bytes.

Optional attributes:

stream id: this is a return value that is set to indicate which stream the data was sent to.

stream sequence number: this value is returned indicating the Stream Sequence Number that was associated with the message.

partial flag: if this returned flag is set to 1, then this message is a partial delivery of the whole message. When this flag is set, the stream id and stream sequence number accompanies this primitive. When this flag is set to 0, it indicates that no more deliveries will be received for this stream sequence number.

payload protocol-id: the 32-bit unsigned integer that was sent to the peer indicating the type of payload protocol of the received data.

11.1.16. Destroy SCTP Instance

DESTROY(local SCTP instance name)

Mandatory attributes:

local SCTP instance name: this is the value that was passed to the application in the initialize primitive and it indicates which SCTP instance is to be destroyed.

Optional attributes:

None.

11.2. SCTP-to-ULP

It is assumed that the operating system or application environment provides a means for the SCTP to asynchronously signal the ULP process. When SCTP does signal a ULP process, certain information is passed to the ULP.

Implementation Note: In some cases, this might be done through a separate socket or error channel.

11.2.1. DATA ARRIVE Notification

SCTP invokes this notification on the ULP when a user message is successfully received and ready for retrieval.

The following might optionally be passed with the notification:

association id: local handle to the SCTP association.

stream id: to indicate which stream the data is received on.

11.2.2. SEND FAILURE Notification

If a message cannot be delivered, SCTP invokes this notification on the ULP.

The following might optionally be passed with the notification:

association id: local handle to the SCTP association.

data retrieval id: an identification used to retrieve unsent and unacknowledged data.

mode: Indicate whether no part of the message never has been sent or if at least part of it has been sent but it is not completely acknowledged.

cause code: indicating the reason of the failure, e.g., size too large, message life time expiration, etc.

context: optional information associated with this message (see Section 11.1.5).

11.2.3. NETWORK STATUS CHANGE Notification

When a destination transport address is marked inactive (e.g., when SCTP detects a failure) or marked active (e.g., when SCTP detects a recovery), SCTP invokes this notification on the ULP.

The following is passed with the notification:

association id: local handle to the SCTP association.

destination transport address: this indicates the destination transport address of the peer endpoint affected by the change.

new-status: this indicates the new status.

11.2.4. COMMUNICATION UP Notification

This notification is used when SCTP becomes ready to send or receive user messages, or when a lost communication to an endpoint is restored.

Implementation Note: If the ASSOCIATE primitive is implemented as a blocking function call, the association parameters are returned as a result of the ASSOCIATE primitive itself. In that case, COMMUNICATION UP notification is optional at the association initiator's side.

The following is passed with the notification:

association id: local handle to the SCTP association.

status: This indicates what type of event has occurred.

destination transport address list: the complete set of transport addresses of the peer.

outbound stream count: the maximum number of streams allowed to be used in this association by the ULP.

inbound stream count: the number of streams the peer endpoint has requested with this association (this might not be the same number as 'outbound stream count').

11.2.5. COMMUNICATION LOST Notification

When SCTP loses communication to an endpoint completely (e.g., via Heartbeats) or detects that the endpoint has performed an abort operation, it invokes this notification on the ULP.

The following is passed with the notification:

association id: local handle to the SCTP association.

status: this indicates what type of event has occurred; the status might indicate that a failure OR a normal termination event occurred in response to a shutdown or abort request.

The following might be passed with the notification:

last-acked: the TSN last acked by that peer endpoint.

last-sent: the TSN last sent to that peer endpoint.

Upper Layer Abort Reason: the abort reason specified in case of a user-initiated abort.

11.2.6. COMMUNICATION ERROR Notification

When SCTP receives an ERROR chunk from its peer and decides to notify its ULP, it can invoke this notification on the ULP.

The following can be passed with the notification:

association id: local handle to the SCTP association.

error info: this indicates the type of error and optionally some additional information received through the ERROR chunk.

11.2.7. RESTART Notification

When SCTP detects that the peer has restarted, it might send this notification to its ULP.

The following can be passed with the notification:

association id: local handle to the SCTP association.

11.2.8. SHUTDOWN COMPLETE Notification

When SCTP completes the shutdown procedures (Section 9.2), this notification is passed to the upper layer.

The following can be passed with the notification:

association id: local handle to the SCTP association.

12. Security Considerations

12.1. Security Objectives

As a common transport protocol designed to reliably carry time-sensitive user messages, such as billing or signaling messages for telephony services, between two networked endpoints, SCTP has the following security objectives.

- * availability of reliable and timely data transport services
- * integrity of the user-to-user information carried by SCTP

12.2. SCTP Responses to Potential Threats

SCTP could potentially be used in a wide variety of risk situations. It is important for operators of systems running SCTP to analyze their particular situations and decide on the appropriate counter-measures.

Operators of systems running SCTP might consult [RFC2196] for guidance in securing their site.

12.2.1. Countering Insider Attacks

The principles of [RFC2196] might be applied to minimize the risk of theft of information or sabotage by insiders. Such procedures include publication of security policies, control of access at the physical, software, and network levels, and separation of services.

12.2.2. Protecting against Data Corruption in the Network

Where the risk of undetected errors in datagrams delivered by the lower-layer transport services is considered to be too great, additional integrity protection is required. If this additional protection were provided in the application layer, the SCTP header would remain vulnerable to deliberate integrity attacks. While the existing SCTP mechanisms for detection of packet replays are considered sufficient for normal operation, stronger protections are needed to protect SCTP when the operating environment contains significant risk of deliberate attacks from a sophisticated adversary.

The SCTP Authentication extension SCTP-AUTH [RFC4895] MAY be used when the threat environment requires stronger integrity protections, but does not require confidentiality.

12.2.3. Protecting Confidentiality

In most cases, the risk of breach of confidentiality applies to the signaling data payload, not to the SCTP or lower-layer protocol overheads. If that is true, encryption of the SCTP user data only might be considered. As with the supplementary checksum service, user data encryption MAY be performed by the SCTP user application. [RFC6083] MAY be used for this. Alternately, the user application MAY use an implementation-specific API to request that the IP Encapsulating Security Payload (ESP) [RFC4303] be used to provide confidentiality and integrity.

Particularly for mobile users, the requirement for confidentiality might include the masking of IP addresses and ports. In this case, ESP SHOULD be used instead of application-level confidentiality. If ESP is used to protect confidentiality of SCTP traffic, an ESP cryptographic transform that includes cryptographic integrity protection MUST be used, because if there is a confidentiality threat there will also be a strong integrity threat.

Regardless of where confidentiality is provided, the Internet Key Exchange Protocol version 2 (IKEv2) [RFC7296] SHOULD be used for key management of ESP.

Operators might consult [RFC4301] for more information on the security services available at and immediately above the Internet Protocol layer.

12.2.4. Protecting against Blind Denial-of-Service Attacks

A blind attack is one where the attacker is unable to intercept or otherwise see the content of data flows passing to and from the target SCTP node. Blind denial-of-service attacks can take the form of flooding, masquerade, or improper monopolization of services.

12.2.4.1. Flooding

The objective of flooding is to cause loss of service and incorrect behavior at target systems through resource exhaustion, interference with legitimate transactions, and exploitation of buffer-related software bugs. Flooding can be directed either at the SCTP node or at resources in the intervening IP Access Links or the Internet. Where the latter entities are the target, flooding will manifest itself as loss of network services, including potentially the breach of any firewalls in place.

In general, protection against flooding begins at the equipment design level, where it includes measures such as:

- * avoiding commitment of limited resources before determining that the request for service is legitimate.
- * giving priority to completion of processing in progress over the acceptance of new work.
- * identification and removal of duplicate or stale queued requests for service.
- * not responding to unexpected packets sent to non-unicast addresses.

Network equipment is expected to be capable of generating an alarm and log if a suspicious increase in traffic occurs. The log provides information such as the identity of the incoming link and source address(es) used, which will help the network or SCTP system operator to take protective measures. Procedures are expected to be in place for the operator to act on such alarms if a clear pattern of abuse emerges.

The design of SCTP is resistant to flooding attacks, particularly in its use of a four-way startup handshake, its use of a cookie to defer commitment of resources at the responding SCTP node until the handshake is completed, and its use of a Verification Tag to prevent insertion of extraneous packets into the flow of an established association.

ESP might be useful in reducing the risk of certain kinds of denial-of-service attacks.

Support for the Host Name Address parameter has been removed from the protocol. Endpoints receiving INIT or INIT ACK chunks containing the Host Name Address parameter MUST send an ABORT chunk in response and MAY include an "Unresolvable Address" error cause.

12.2.4.2. Blind Masquerade

Masquerade can be used to deny service in several ways:

- * by tying up resources at the target SCTP node to which the impersonated node has limited access. For example, the target node can by policy permit a maximum of one SCTP association with the impersonated SCTP node. The masquerading attacker can attempt to establish an association purporting to come from the impersonated node so that the latter cannot do so when it requires it.
- * by deliberately allowing the impersonation to be detected, thereby provoking counter-measures that cause the impersonated node to be locked out of the target SCTP node.
- * by interfering with an established association by inserting extraneous content such as a SHUTDOWN chunk.

SCTP reduces the risk of blind masquerade attacks through IP spoofing by use of the four-way startup handshake. Because the initial exchange is memory-less, no lockout mechanism is triggered by blind masquerade attacks. In addition, the packet containing the INIT ACK chunk with the State Cookie is transmitted back to the IP address from which it received the packet containing the INIT chunk. Thus,

the attacker would not receive the INIT ACK chunk containing the State Cookie. SCTP protects against insertion of extraneous packets into the flow of an established association by use of the Verification Tag.

Logging of received INIT chunks and abnormalities such as unexpected INIT ACK chunks might be considered as a way to detect patterns of hostile activity. However, the potential usefulness of such logging has to be weighed against the increased SCTP startup processing it implies, rendering the SCTP node more vulnerable to flooding attacks. Logging is pointless without the establishment of operating procedures to review and analyze the logs on a routine basis.

12.2.4.3. Improper Monopolization of Services

Attacks under this heading are performed openly and legitimately by the attacker. They are directed against fellow users of the target SCTP node or of the shared resources between the attacker and the target node. Possible attacks include the opening of a large number of associations between the attacker's node and the target, or transfer of large volumes of information within a legitimately established association.

Policy limits are expected to be placed on the number of associations per adjoining SCTP node. SCTP user applications are expected to be capable of detecting large volumes of illegitimate or "no-op" messages within a given association and either logging or terminating the association as a result, based on local policy.

12.3. SCTP Interactions with Firewalls

It is helpful for some firewalls if they can inspect just the first fragment of a fragmented SCTP packet and unambiguously determine whether it corresponds to an INIT chunk (for further information, please refer to [RFC1858]). Accordingly, we stress the requirements, as stated in Section 3.1, that (1) an INIT chunk MUST NOT be bundled with any other chunk in a packet and (2) a packet containing an INIT chunk MUST have a zero Verification Tag. The receiver of an INIT chunk MUST silently discard the INIT chunk and all further chunks if the INIT chunk is bundled with other chunks or the packet has a non-zero Verification Tag.

12.4. Protection of Non-SCTP-Capable Hosts

To provide a non-SCTP-capable host with the same level of protection against attacks as for SCTP-capable ones, all SCTP implementations MUST implement the ICMP handling described in Section 10.

When an SCTP implementation receives a packet containing multiple control or DATA chunks and the processing of the packet would result in sending multiple chunks in response, the sender of the response chunk(s) MUST NOT send more than one packet containing chunks other than DATA chunks. This requirement protects the network for triggering a packet burst in response to a single packet. If bundling is supported, multiple response chunks that fit into a single packet MAY be bundled together into one single response packet. If bundling is not supported, then the sender MUST NOT send more than one response chunk and MUST discard all other responses. Note that this rule does not apply to a SACK chunk, since a SACK chunk is, in itself, a response to DATA chunks and a SACK chunk does not require a response of more DATA chunks.

An SCTP implementation MUST abort the association if it receives a SACK chunk acknowledging a TSN that has not been sent.

An SCTP implementation that receives an INIT chunk that would require a large packet in response, due to the inclusion of multiple "Unrecognized Parameter" parameters, MAY (at its discretion) elect to omit some or all of the "Unrecognized Parameter" parameters to reduce the size of the INIT ACK chunk. Due to a combination of the size of the State Cookie parameter and the number of addresses a receiver of an INIT chunk indicates to a peer, it is always possible that the INIT ACK chunk will be larger than the original INIT chunk. An SCTP implementation SHOULD attempt to make the INIT ACK chunk as small as possible to reduce the possibility of byte amplification attacks.

13. Network Management Considerations

The MIB module for SCTP defined in [RFC3873] applies for the version of the protocol specified in this document.

14. Recommended Transmission Control Block (TCB) Parameters

This section details a set of parameters that are expected to be contained within the TCB for an implementation. This section is for illustrative purposes and is not considered to be requirements on an implementation or as an exhaustive list of all parameters inside an SCTP TCB. Each implementation might need its own additional parameters for optimization.

14.1. Parameters Necessary for the SCTP Instance

Associations: A list of current associations and mappings to the

data consumers for each association. This might be in the form of a hash table or other implementation-dependent structure. The data consumers might be process identification information such as file descriptors, named pipe pointer, or table pointers dependent on how SCTP is implemented.

Secret Key: A secret key used by this endpoint to compute the MAC. This SHOULD be a cryptographic quality random number with a sufficient length. Discussion in [RFC4086] can be helpful in selection of the key.

Address List: The list of IP addresses that this instance has bound. This information is passed to one's peer(s) in INIT and INIT ACK chunks.

SCTP Port: The local SCTP port number to which the endpoint is bound.

14.2. Parameters Necessary per Association (i.e., the TCB)

Peer Verification Tag: Tag value to be sent in every packet and is received in the INIT or INIT ACK chunk.

My Verification Tag: Tag expected in every inbound packet and sent in the INIT or INIT ACK chunk.

State: COOKIE-WAIT, COOKIE-ECHOED, ESTABLISHED, SHUTDOWN-PENDING, SHUTDOWN-SENT, SHUTDOWN-RECEIVED, SHUTDOWN-ACK-SENT.

Note: No "CLOSED" state is illustrated since if a association is "CLOSED" its TCB SHOULD be removed.

Peer Transport Address List: A list of SCTP transport addresses to which the peer is bound. This information is derived from the INIT or INIT ACK chunk and is used to associate an inbound packet with a given association. Normally, this information is hashed or keyed for quick lookup and access of the TCB.

Primary Path: This is the current primary destination transport address of the peer endpoint. It might also specify a source transport address on this endpoint.

Overall Error Count: The overall association error count.

Overall Error Threshold: The threshold for this association that if the Overall Error Count reaches will cause this association to be torn down.

Peer Rwnd: Current calculated value of the peer's rwnd.

Next TSN: The next TSN number to be assigned to a new DATA chunk. This is sent in the INIT or INIT ACK chunk to the peer and incremented each time a DATA chunk is assigned a TSN (normally just prior to transmit or during fragmentation).

Last Rcvd TSN: This is the last TSN received in sequence. This value is set initially by taking the peer's initial TSN, received in the INIT or INIT ACK chunk, and subtracting one from it.

Mapping Array: An array of bits or bytes indicating which out-of-order TSNs have been received (relative to the Last Rcvd TSN). If no gaps exist, i.e., no out-of-order packets have been received, this array will be set to all zero. This structure might be in the form of a circular buffer or bit array.

Ack State: This flag indicates if the next received packet is to be responded to with a SACK chunk. This is initialized to 0. When a packet is received it is incremented. If this value reaches 2 or more, a SACK chunk is sent and the value is reset to 0. Note: This is used only when no DATA chunks are received out of order. When DATA chunks are out of order, SACK chunks are not delayed (see Section 6).

Inbound Streams: An array of structures to track the inbound streams, normally including the next sequence number expected and possibly the stream number.

Outbound Streams: An array of structures to track the outbound streams, normally including the next sequence number to be sent on the stream.

Reasm Queue: A reassembly queue.

Receive Buffer: A buffer to store received user data which has not been delivered to the upper layer.

Local Transport Address List: The list of local IP addresses bound in to this association.

Association Maximum DATA Chunk Size: The smallest Path Maximum DATA Chunk Size of all destination addresses.

14.3. Per Transport Address Data

For each destination transport address in the peer's address list derived from the INIT or INIT ACK chunk, a number of data elements need to be maintained including:

Error Count: The current error count for this destination.

Error Threshold: Current error threshold for this destination, i.e., what value marks the destination down if error count reaches this value.

cwnd: The current congestion window.

ssthresh: The current ssthresh value.

RTO: The current retransmission timeout value.

SRTT: The current smoothed round-trip time.

RTTVAR: The current RTT variation.

partial bytes acked: The tracking method for increase of cwnd when in congestion avoidance mode (see Section 7.2.2).

state: The current state of this destination, i.e., DOWN, UP, ALLOW-HEARTBEAT, NO-HEARTBEAT, etc.

PMTU: The current known PMTU.

PMDCS: The current known PMDCS.

Per Destination Timer: A timer used by each destination.

RTO-Pending: A flag used to track if one of the DATA chunks sent to this address is currently being used to compute an RTT. If this flag is 0, the next DATA chunk sent to this destination is expected to be used to compute an RTT and this flag is expected to be set. Every time the RTT calculation completes (i.e., the DATA chunk is acknowledged), clear this flag.

last-time: The time to which this destination was last sent. This can be used to determine if the sending of a HEARTBEAT chunk is needed.

14.4. General Parameters Needed

Out Queue: A queue of outbound DATA chunks.

In Queue: A queue of inbound DATA chunks.

15. IANA Considerations

This document defines five registries that IANA maintains:

- * through definition of additional chunk types,
- * through definition of additional chunk flags,
- * through definition of additional parameter types,
- * through definition of additional cause codes within ERROR chunks,
or
- * through definition of additional payload protocol identifiers.

IANA is requested to perform the following updates for the above five registries:

- * In the Chunk Types Registry replace in the Reference section the reference to [RFC4960] and [RFC6096] by a reference to this document.

Replace in the Notes section the reference to Section 3.2 of [RFC6096] by a reference to Section 15.2 of this document.

Finally replace each reference to [RFC4960] by a reference to this document for the following chunk types:

- Payload Data (DATA)
- Initiation (INIT)
- Initiation Acknowledgement (INIT ACK)
- Selective Acknowledgement (SACK)
- Heartbeat Request (HEARTBEAT)
- Heartbeat Acknowledgement (HEARTBEAT ACK)
- Abort (ABORT)

- Shutdown (SHUTDOWN)
- Shutdown Acknowledgement (SHUTDOWN ACK)
- Operation Error (ERROR)
- State Cookie (COOKIE ECHO)
- Cookie Acknowledgement (COOKIE ACK)
- Reserved for Explicit Congestion Notification Echo (ECNE)
- Reserved for Congestion Window Reduced (CWR)
- Shutdown Complete (SHUTDOWN COMPLETE)
- Reserved for IETF-defined Chunk Extensions

- * In the Chunk Parameter Types Registry replace in the Reference section the reference to [RFC4960] by a reference to this document.

Replace each reference to [RFC4960] by a reference to this document for the following chunk parameter types:

- Heartbeat Info
- IPv4 Address
- IPv6 Address
- State Cookie
- Unrecognized Parameters
- Cookie Preservative
- Host Name Address
- Supported Address Types

Add a reference to this document for the following chunk parameter type:

- Reserved for ECN Capable (0x8000)

- * In the Chunk Flags Registry replace in the Reference section the reference to [RFC6096] by a reference to this document.

Replace each reference to [RFC4960] by a reference to this document for the following DATA chunk flags:

- E bit
- B bit
- U bit

Replace each reference to [RFC4960] by a reference to this document for the following ABORT chunk flags:

- T bit

Replace each reference to [RFC4960] by a reference to this document for the following SHUTDOWN COMPLETE chunk flags:

- T bit

- * In the Error Cause Codes Registry replace in the Reference section the reference to [RFC6096] by a reference to this document.

Replace each reference to [RFC4960] by a reference to this document for the following cause codes:

- Invalid Stream Identifier
- Missing Mandatory Parameter
- Stale Cookie Error
- Out of Resource
- Unresolvable Address
- Unrecognized Chunk Type
- Invalid Mandatory Parameter
- Unrecognized Parameters
- No User Data
- Cookie Received While Shutting Down
- Restart of an Association with New Addressess

Replace each reference to [RFC4460] by a reference to this document for the following cause codes:

- User Initiated Abort
- Protocol Violation
- * In the SCTP Payload Protocol Identifiers Registry replace in the Reference section the reference to [RFC6096] by a reference to this document.

Replace each reference to [RFC4960] by a reference to this document for the following SCTP payload protocol identifiers:

- Reserved by SCTP

SCTP requires that the IANA Port Numbers registry be opened for SCTP port registrations, Section 15.6 describes how. An IESG-appointed Expert Reviewer supports IANA in evaluating SCTP port allocation requests.

IANA is requested to perform the following update for the Port Number registry. Replace each reference to [RFC4960] by a reference to this document for the following SCTP port numbers:

- * 9 (discard)
- * 20 (ftp-data)
- * 21 (ftp)
- * 22 (ssh)
- * 80 (http)
- * 179 (bgp)
- * 443 (https)

Furthermore, IANA is requested to replace in the HTTP Digest Algorithm Values registry the reference to Appendix B of [RFC4960] to Appendix A of this document.

IANA is also requested to replace in the ONC RPC Netids registry, each of the reference to [RFC4960] by a reference to this document for the following netids:

- * sctp

- * sctp6

IANA is finally requested to replace in the IPFIX Information Elements registry, each of the reference to [RFC4960] by a reference to this document for the following elements with the name:

- * sourceTransportPort
- * destinationTransportPort
- * collectorTransportPort
- * exporterTransportPort
- * postNAPTSourceTransportPort
- * postNAPTDestinationTransportPort

15.1. IETF-Defined Chunk Extension

The assignment of new chunk type codes is done through an IETF Review action, as defined in [RFC8126]. Documentation for a new chunk MUST contain the following information:

- a) A long and short name for the new chunk type.
- b) A detailed description of the structure of the chunk, which MUST conform to the basic structure defined in Section 3.2.
- c) A detailed definition and description of intended use of each field within the chunk, including the chunk flags if any. Defined chunk flags will be used as initial entries in the chunk flags table for the new chunk type.
- d) A detailed procedural description of the use of the new chunk type within the operation of the protocol.

The last chunk type (255) is reserved for future extension if necessary.

For each new chunk type, IANA creates a registration table for the chunk flags of that type. The procedure for registering particular chunk flags is described in Section 15.2.

15.2. IETF Chunk Flags Registration

The assignment of new chunk flags is done through an RFC Required action, as defined in [RFC8126]. Documentation for the chunk flags MUST contain the following information:

- a) A name for the new chunk flag.
- b) A detailed procedural description of the use of the new chunk flag within the operation of the protocol. It MUST be considered that implementations not supporting the flag will send 0 on transmit and just ignore it on receipt.

IANA selects a chunk flags value. This MUST be one of 0x01, 0x02, 0x04, 0x08, 0x10, 0x20, 0x40, or 0x80, which MUST be unique within the chunk flag values for the specific chunk type.

15.3. IETF-Defined Chunk Parameter Extension

The assignment of new chunk parameter type codes is done through an IETF Review action as defined in [RFC8126]. Documentation of the chunk parameter MUST contain the following information:

- a) Name of the parameter type.
- b) Detailed description of the structure of the parameter field. This structure MUST conform to the general Type-Length-Value format described in Section 3.2.1.
- c) Detailed definition of each component of the parameter value.
- d) Detailed description of the intended use of this parameter type, and an indication of whether and under what circumstances multiple instances of this parameter type can be found within the same chunk.
- e) Each parameter type MUST be unique across all chunks.

15.4. IETF-Defined Additional Error Causes

Additional cause codes can be allocated in the range 11 to 65535 through a Specification Required action as defined in [RFC8126]. Provided documentation MUST include the following information:

- a) Name of the error condition.
- b) Detailed description of the conditions under which an SCTP endpoint issues an ERROR (or ABORT) chunk with this cause code.

- c) Expected action by the SCTP endpoint that receives an ERROR (or ABORT) chunk containing this cause code.
- d) Detailed description of the structure and content of data fields that accompany this cause code.

The initial word (32 bits) of a cause code parameter MUST conform to the format shown in Section 3.3.10, i.e.:

- * first 2 bytes contain the cause code value
- * last 2 bytes contain the length of the cause parameter.

15.5. Payload Protocol Identifiers

The assignment of payload protocol identifier is done using the First Come First Served policy as defined in [RFC8126].

Except for value 0, which is reserved to indicate an unspecified payload protocol identifier in a DATA chunk, an SCTP implementation will not be responsible for standardizing or verifying any payload protocol identifiers; An SCTP implementation simply receives the identifier from the upper layer and carries it with the corresponding payload data.

The upper layer, i.e., the SCTP user, SHOULD standardize any specific protocol identifier with IANA if it is so desired. The use of any specific payload protocol identifier is out of the scope of this specification.

15.6. Port Numbers Registry

SCTP services can use contact port numbers to provide service to unknown callers, as in TCP and UDP. An IESG-appointed expert reviewer supports IANA in evaluating SCTP port allocation requests, according to the procedure defined in [RFC8126]. The details of this process are defined in [RFC6335].

16. Suggested SCTP Protocol Parameter Values

The following protocol parameters are RECOMMENDED:

RTO.Initial: 1 second

RTO.Min: 1 second

RTO.Max: 60 seconds

Max.Burst: 4

RTO.Alpha: 1/8

RTO.Beta: 1/4

Valid.Cookie.Life: 60 seconds

Association.Max.Retrans: 10 attempts

Path.Max.Retrans: 5 attempts (per destination address)

Max.Init.Retransmits: 8 attempts

HB.interval: 30 seconds

HB.Max.Burst: 1

SACK.Delay: 200 milliseconds

Implementation Note: The SCTP implementation can allow ULP to customize some of these protocol parameters (see Section 11).

'RTO.Min' SHOULD be set as described above in this section.

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Appendix A. CRC32c Checksum Calculation

We define a 'reflected value' as one that is the opposite of the normal bit order of the machine. The 32-bit CRC (Cyclic Redundancy Check) is calculated as described for CRC32c and uses the polynomial code 0x11EDC6F41 (Castagnoli93) or $x^{32}+x^{28}+x^{27}+x^{26}+x^{25}+x^{23}+x^{22}+x^{20}+x^{19}+x^{18}+x^{14}+x^{13}+x^{11}+x^{10}+x^9+x^8+x^6+x^0$. The CRC is computed using a procedure similar to ETHERNET CRC [ITU.V42.1994], modified to reflect transport-level usage.

CRC computation uses polynomial division. A message bit-string M is transformed to a polynomial, $M(X)$, and the CRC is calculated from $M(X)$ using polynomial arithmetic.

When CRCs are used at the link layer, the polynomial is derived from on-the-wire bit ordering: the first bit 'on the wire' is the high-order coefficient. Since SCTP is a transport-level protocol, it cannot know the actual serial-media bit ordering. Moreover, different links in the path between SCTP endpoints can use different link-level bit orders.

A convention therefore is established for mapping SCTP transport messages to polynomials for purposes of CRC computation. The bit-ordering for mapping SCTP messages to polynomials is that bytes are taken most-significant first, but within each byte, bits are taken least-significant first. The first byte of the message provides the eight highest coefficients. Within each byte, the least-significant SCTP bit gives the most-significant polynomial coefficient within that byte, and the most-significant SCTP bit is the least-significant polynomial coefficient in that byte. (This bit ordering is sometimes called 'mirrored' or 'reflected' [WILLIAMS93].) CRC polynomials are to be transformed back into SCTP transport-level byte values, using a consistent mapping.

The SCTP transport-level CRC value can be calculated as follows:

- * CRC input data are assigned to a byte stream, numbered from 0 to N-1.
- * The transport-level byte stream is mapped to a polynomial value. An N-byte PDU with j bytes numbered 0 to N-1 is considered as coefficients of a polynomial $M(x)$ of order $8*N-1$, with bit 0 of byte j being coefficient $x^{(8*(N-j)-8)}$, and bit 7 of byte j being coefficient $x^{(8*(N-j)-1)}$.
- * The CRC remainder register is initialized with all 1s and the CRC is computed with an algorithm that simultaneously multiplies by x^{32} and divides by the CRC polynomial.
- * The polynomial is multiplied by x^{32} and divided by $G(x)$, the generator polynomial, producing a remainder $R(x)$ of degree less than or equal to 31.
- * The coefficients of $R(x)$ are considered a 32-bit sequence.
- * The bit sequence is complemented. The result is the CRC polynomial.
- * The CRC polynomial is mapped back into SCTP transport-level bytes. The coefficient of x^{31} gives the value of bit 7 of SCTP byte 0, and the coefficient of x^{24} gives the value of bit 0 of byte 0. The coefficient of x^7 gives bit 7 of byte 3, and the coefficient of x^0 gives bit 0 of byte 3. The resulting 4-byte transport-level sequence is the 32-bit SCTP checksum value.

Implementation Note: Standards documents, textbooks, and vendor literature on CRCs often follow an alternative formulation, in which the register used to hold the remainder of the long-division algorithm is initialized to zero rather than all ones, and instead

the first 32 bits of the message are complemented. The long-division algorithm used in our formulation is specified such that the initial multiplication by 2^{32} and the long-division are combined into one simultaneous operation. For such algorithms, and for messages longer than 64 bits, the two specifications are precisely equivalent. That equivalence is the intent of this document.

Implementors of SCTP are warned that both specifications are to be found in the literature, sometimes with no restriction on the long-division algorithm. The choice of formulation in this document is to permit non-SCTP usage, where the same CRC algorithm can be used to protect messages shorter than 64 bits.

There can be a computational advantage in validating the association against the Verification Tag, prior to performing a checksum, as invalid tags will result in the same action as a bad checksum in most cases. The exceptions for this technique would be packets containing INIT chunks and some SHUTDOWN-COMplete chunks, as well as a stale COOKIE ECHO chunks. These special-case exchanges represent small packets and will minimize the effect of the checksum calculation.

The following non-normative sample code is taken from an open-source CRC generator [WILLIAMS93], using the "mirroring" technique and yielding a lookup table for SCTP CRC32c with 256 entries, each 32 bits wide. While neither especially slow nor especially fast, as software table-lookup CRCs go, it has the advantage of working on both big-endian and little-endian CPUs, using the same (host-order) lookup tables, and using only the predefined `ntohl()` and `htonl()` operations. The code is somewhat modified from [WILLIAMS93], to ensure portability between big-endian and little-endian architectures, use fixed sized types to allow portability between 32-bit and 64-bit platforms, and general C code improvements. (Note that if the byte endian-ness of the target architecture is known to be little-endian, the final bit-reversal and byte-reversal steps can be folded into a single operation.)

```
<CODE BEGINS>
/*****/
/* Note: The definitions for Ross Williams's table generator */
/* would be TB_WIDTH=4, TB_POLY=0x1EDC6F41, TB_REVER=TRUE. */
/* For Mr. Williams's direct calculation code, use the settings */
/* cm_width=32, cm_poly=0x1EDC6F41, cm_init=0xFFFFFFFF, */
/* cm_refin=TRUE, cm_refot=TRUE, cm_xorot=0x00000000. */
/*****/

/* Example of the crc table file */
#ifndef __crc32cr_h__
#define __crc32cr_h__
```

```

#define CRC32C_POLY 0x1EDC6F41UL
#define CRC32C(c,d) (c=(c>>8)^crc_c[(c^(d))&0xFF])

uint32_t crc_c[256] = {
    0x00000000UL, 0xF26B8303UL, 0xE13B70F7UL, 0x1350F3F4UL,
    0xC79A971FUL, 0x35F1141CUL, 0x26A1E7E8UL, 0xD4CA64EBUL,
    0x8AD958CFUL, 0x78B2DBCCUL, 0x6BE22838UL, 0x9989AB3BUL,
    0x4D43CFD0UL, 0xBF284CD3UL, 0xAC78BF27UL, 0x5E133C24UL,
    0x105EC76FUL, 0xE235446CUL, 0xF165B798UL, 0x030E349BUL,
    0xD7C45070UL, 0x25AFD373UL, 0x36FF2087UL, 0xC494A384UL,
    0x9A879FA0UL, 0x68EC1CA3UL, 0x7BBCEF57UL, 0x89D76C54UL,
    0x5D1D08BFUL, 0xAF768BBCUL, 0xBC267848UL, 0x4E4DFB4BUL,
    0x20BD8EDEUL, 0xD2D60DDFUL, 0xC186FE29UL, 0x33ED7D2AUL,
    0xE72719C1UL, 0x154C9AC2UL, 0x061C6936UL, 0xF477EA35UL,
    0xAA64D611UL, 0x580F5512UL, 0x4B5FA6E6UL, 0xB93425E5UL,
    0x6DFE410EUL, 0x9F95C20DUL, 0x8CC531F9UL, 0x7EAE2FAUL,
    0x30E349B1UL, 0xC288CAB2UL, 0xD1D83946UL, 0x23B3BA45UL,
    0xF779DEAEUL, 0x05125DADUL, 0x1642AE59UL, 0xE4292D5AUL,
    0xBA3A117EUL, 0x4851927DUL, 0x5B016189UL, 0xA96AE28AUL,
    0x7DA08661UL, 0x8FCB0562UL, 0x9C9BF696UL, 0x6EF07595UL,
    0x417B1DBCUL, 0xB3109EBFUL, 0xA0406D4BUL, 0x522BEE48UL,
    0x86E18AA3UL, 0x748A09A0UL, 0x67DAFA54UL, 0x95B17957UL,
    0xCBA24573UL, 0x39C9C670UL, 0x2A993584UL, 0xD8F2B687UL,
    0x0C38D26CUL, 0xFE53516FUL, 0xED03A29BUL, 0x1F682198UL,
    0x5125DAD3UL, 0xA34E59D0UL, 0xB01EAA24UL, 0x42752927UL,
    0x96BF4DCCUL, 0x64D4CECFUL, 0x77843D3BUL, 0x85EFBE38UL,
    0xDBFC821CUL, 0x2997011FUL, 0x3AC7F2EBUL, 0xC8AC71E8UL,
    0x1C661503UL, 0xEE0D9600UL, 0xFD5D65F4UL, 0x0F36E6F7UL,
    0x61C69362UL, 0x93AD1061UL, 0x80FDE395UL, 0x72966096UL,
    0xA65C047DUL, 0x5437877EUL, 0x4767748AUL, 0xB50CF789UL,
    0xEB1FCBADUL, 0x197448AEUL, 0x0A24BB5AUL, 0xF84F3859UL,
    0x2C855CB2UL, 0xDEEEDFB1UL, 0xCDBE2C45UL, 0x3FD5AF46UL,
    0x7198540DUL, 0x83F3D70EUL, 0x90A324FAUL, 0x62C8A7F9UL,
    0xB602C312UL, 0x44694011UL, 0x5739B3E5UL, 0xA55230E6UL,
    0xFB410CC2UL, 0x092A8FC1UL, 0x1A7A7C35UL, 0xE811FF36UL,
    0x3CDB9BDDUL, 0xCEB018DEUL, 0xDDE0EB2AUL, 0x2F8B6829UL,
    0x82F63B78UL, 0x709DB87BUL, 0x63CD4B8FUL, 0x91A6C88CUL,
    0x456CAC67UL, 0xB7072F64UL, 0xA457DC90UL, 0x563C5F93UL,
    0x082F63B7UL, 0xFA44E0B4UL, 0xE9141340UL, 0x1B7F9043UL,
    0xCFB5F4A8UL, 0x3DDE77ABUL, 0x2E8E845FUL, 0xDCE5075CUL,
    0x92A8FC17UL, 0x60C37F14UL, 0x73938CE0UL, 0x81F80FE3UL,
    0x55326B08UL, 0xA759E80BUL, 0xB4091BFFUL, 0x466298FCUL,
    0x1871A4D8UL, 0xEA1A27DBUL, 0xF94AD42FUL, 0x0B21572CUL,
    0xDFEB33C7UL, 0x2D80B0C4UL, 0x3ED04330UL, 0xCCBBC033UL,
    0xA24BB5A6UL, 0x502036A5UL, 0x4370C551UL, 0xB11B4652UL,
    0x65D122B9UL, 0x97BAA1BAUL, 0x84EA524EUL, 0x7681D14DUL,
    0x2892ED69UL, 0xDAF96E6AUL, 0xC9A99D9EUL, 0x3BC21E9DUL,
    0xEF087A76UL, 0x1D63F975UL, 0x0E330A81UL, 0xFC588982UL,

```

```
0xB21572C9UL, 0x407EF1CAUL, 0x532E023EUL, 0xA145813DUL,
0x758FE5D6UL, 0x87E466D5UL, 0x94B49521UL, 0x66DF1622UL,
0x38CC2A06UL, 0xCA7A905UL, 0xD9F75AF1UL, 0x2B9CD9F2UL,
0xFF56BD19UL, 0x0D3D3E1AUL, 0x1E6DCDEEUL, 0xEC064EEDUL,
0xC38D26C4UL, 0x31E6A5C7UL, 0x22B65633UL, 0xD0DDD530UL,
0x0417B1DBUL, 0xF67C32D8UL, 0xE52CC12CUL, 0x1747422FUL,
0x49547E0BUL, 0xBB3FFD08UL, 0xA86F0EFCUL, 0x5A048DFFUL,
0x8ECEE914UL, 0x7CA56A17UL, 0x6FF599E3UL, 0x9D9E1AE0UL,
0xD3D3E1ABUL, 0x21B862A8UL, 0x32E8915CUL, 0xC083125FUL,
0x144976B4UL, 0xE622F5B7UL, 0xF5720643UL, 0x07198540UL,
0x590AB964UL, 0xAB613A67UL, 0xB831C993UL, 0x4A5A4A90UL,
0x9E902E7BUL, 0x6CFBAD78UL, 0x7FAB5E8CUL, 0x8DC0DD8FUL,
0xE330A81AUL, 0x115B2B19UL, 0x020BD8EDUL, 0xF0605BEEUL,
0x24AA3F05UL, 0xD6C1BC06UL, 0xC5914FF2UL, 0x37FACCF1UL,
0x69E9F0D5UL, 0x9B8273D6UL, 0x88D28022UL, 0x7AB90321UL,
0xAE7367CAUL, 0x5C18E4C9UL, 0x4F48173DUL, 0xBD23943EUL,
0xF36E6F75UL, 0x0105EC76UL, 0x12551F82UL, 0xE03E9C81UL,
0x34F4F86AUL, 0xC69F7B69UL, 0xD5CF889DUL, 0x27A40B9EUL,
0x79B737BAUL, 0x8BDCB4B9UL, 0x988C474DUL, 0x6AE7C44EUL,
0xBE2DA0A5UL, 0x4C4623A6UL, 0x5F16D052UL, 0xAD7D5351UL,
};
```

```
#endif
```

```
/* Example of table build routine */
```

```
#include <stdio.h>
#include <stdlib.h>
```

```
#define OUTPUT_FILE    "crc32cr.h"
#define CRC32C_POLY    0x1EDC6F41UL
```

```
static FILE *tf;
```

```
static uint32_t
reflect_32(uint32_t b)
{
    int i;
    uint32_t rw = 0UL;

    for (i = 0; i < 32; i++) {
        if (b & 1)
            rw |= 1UL << (31 - i);
        b >>= 1;
    }
    return (rw);
}
```

```

static uint32_t
build_crc_table (int index)
{
    int i;
    uint32_t rb;

    rb = reflect_32(index);

    for (i = 0; i < 8; i++) {
        if (rb & 0x80000000UL)
            rb = (rb << 1) ^ (uint32_t)CRC32C_POLY;
        else
            rb <<= 1;
    }
    return (reflect_32(rb));
}

int
main (void)
{
    int i;

    printf("\nGenerating CRC32c table file <%s>.\n",
        OUTPUT_FILE);
    if ((tf = fopen(OUTPUT_FILE, "w")) == NULL) {
        printf("Unable to open %s.\n", OUTPUT_FILE);
        exit (1);
    }
    fprintf(tf, "#ifndef __crc32cr_h__\n");
    fprintf(tf, "#define __crc32cr_h__\n\n");
    fprintf(tf, "#define CRC32C_POLY 0x%08XUL\n",
        (uint32_t)CRC32C_POLY);
    fprintf(tf,
        "#define CRC32C(c,d) (c=(c>>8)^crc_c[(c^(d))&0xFF])\n");
    fprintf(tf, "\nuint32_t crc_c[256] =\n{\n");
    for (i = 0; i < 256; i++) {
        fprintf(tf, "0x%08XUL, ", build_crc_table (i));
        if ((i & 3) == 3)
            fprintf(tf, "\n");
        else
            fprintf(tf, " ");
    }
    fprintf(tf, "};\n\n#endif\n");

    if (fclose(tf) != 0)
        printf("Unable to close <%s>.\n", OUTPUT_FILE);
    else
        printf("\nThe CRC32c table has been written to <%s>.\n",

```

```
        OUTPUT_FILE);
    return (0);
}

/* Example of crc insertion */

#include "crc32cr.h"

uint32_t
generate_crc32c(unsigned char *buffer, unsigned int length)
{
    unsigned int i;
    uint32_t crc32 = 0xffffffffUL;
    uint32_t result;
    uint32_t byte0, byte1, byte2, byte3;

    for (i = 0; i < length; i++) {
        CRC32C(crc32, buffer[i]);
    }

    result = ~crc32;

    /* result now holds the negated polynomial remainder,
     * since the table and algorithm are "reflected" [williams95].
     * That is, result has the same value as if we mapped the message
     * to a polynomial, computed the host-bit-order polynomial
     * remainder, performed final negation, and then did an
     * end-for-end bit-reversal.
     * Note that a 32-bit bit-reversal is identical to four in-place
     * 8-bit bit-reversals followed by an end-for-end byteswap.
     * In other words, the bits of each byte are in the right order,
     * but the bytes have been byteswapped. So, we now do an explicit
     * byteswap. On a little-endian machine, this byteswap and
     * the final ntohl cancel out and could be elided.
     */

    byte0 = result & 0xff;
    byte1 = (result>>8) & 0xff;
    byte2 = (result>>16) & 0xff;
    byte3 = (result>>24) & 0xff;
    crc32 = ((byte0 << 24) |
              (byte1 << 16) |
              (byte2 << 8) |
              byte3);
    return (crc32);
}

int
```

```
insert_crc32(unsigned char *buffer, unsigned int length)
{
    SCTP_message *message;
    uint32_t crc32;

    message = (SCTP_message *)buffer;
    message->common_header.checksum = 0UL;
    crc32 = generate_crc32c(buffer, length);
    /* and insert it into the message */
    message->common_header.checksum = htonl(crc32);
    return (1);
}

int
validate_crc32(unsigned char *buffer, unsigned int length)
{
    SCTP_message *message;
    unsigned int i;
    uint32_t original_crc32;
    uint32_t crc32;

    /* save and zero checksum */
    message = (SCTP_message *)buffer;
    original_crc32 = ntohl(message->common_header.checksum);
    message->common_header.checksum = 0L;
    crc32 = generate_crc32c(buffer, length);
    return ((original_crc32 == crc32) ? 1 : -1);
}
<CODE ENDS>
```

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Propagating Explicit Congestion Notification Across IP Tunnel Headers
Separated by a Shim
draft-ietf-tsvwg-rfc6040update-shim-14

Abstract

RFC 6040 on "Tunnelling of Explicit Congestion Notification" made the rules for propagation of ECN consistent for all forms of IP in IP tunnel. This specification updates RFC 6040 to clarify that its scope includes tunnels where two IP headers are separated by at least one shim header that is not sufficient on its own for wide area packet forwarding. It surveys widely deployed IP tunnelling protocols that use such shim header(s) and updates the specifications of those that do not mention ECN propagation (L2TPv2, L2TPv3, GRE, Teredo and AMT). This specification also updates RFC 6040 with configuration requirements needed to make any legacy tunnel ingress safe.

Status of This Memo

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1. Introduction

RFC 6040 on "Tunnelling of Explicit Congestion Notification" [RFC6040] made the rules for propagation of Explicit Congestion Notification (ECN [RFC3168]) consistent for all forms of IP in IP tunnel.

A common pattern for many tunnelling protocols is to encapsulate an inner IP header (v4 or v6) with shim header(s) then an outer IP header (v4 or v6). Some of these shim headers are designed as generic encapsulations, so they do not necessarily directly encapsulate an inner IP header. Instead they can encapsulate headers such as link-layer (L2) protocols that in turn often encapsulate IP.

To clear up confusion, this specification clarifies that the scope of RFC 6040 includes any IP-in-IP tunnel, including those with shim header(s) and other encapsulations between the IP headers. Where necessary, it updates the specifications of the relevant encapsulation protocols with the specific text necessary to comply with RFC 6040.

This specification also updates RFC 6040 to state how operators ought to configure a legacy tunnel ingress to avoid unsafe system configurations.

2. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119 [RFC2119] when, and only when, they appear in all capitals, as shown here.

This specification uses the terminology defined in RFC 6040 [RFC6040].

3. Scope of RFC 6040

In section 1.1 of RFC 6040, its scope is defined as:

"...ECN field processing at encapsulation and decapsulation for any IP-in-IP tunnelling, whether IPsec or non-IPsec tunnels. It applies irrespective of whether IPv4 or IPv6 is used for either the inner or outer headers. ..."

This was intended to include cases where shim header(s) sit between the IP headers. Many tunnelling implementers have interpreted the scope of RFC 6040 as it was intended, but it is ambiguous. Therefore, this specification updates RFC 6040 by adding the following scoping text after the sentences quoted above:

It applies in cases where an outer IP header encapsulates an inner IP header either directly or indirectly by encapsulating other headers that in turn encapsulate (or might encapsulate) an inner IP header.

There is another problem with the scope of RFC 6040. Like many IETF specifications, RFC 6040 is written as a specification that implementations can choose to claim compliance with. This means it does not cover two important cases:

1. those cases where it is infeasible for an implementation to access an inner IP header when adding or removing an outer IP header;
2. those implementations that choose not to propagate ECN between IP headers.

However, the ECN field is a non-optional part of the IP header (v4 and v6). So any implementation that creates an outer IP header has to give the ECN field some value. There is only one safe value a tunnel ingress can use if it does not know whether the egress supports propagation of the ECN field; it has to clear the ECN field in any outer IP header to 0b00.

However, an RFC has no jurisdiction over implementations that choose not to comply with it or cannot comply with it, including all those implementations that pre-dated the RFC. Therefore it would have been unreasonable to add such a requirement to RFC 6040. Nonetheless, to ensure safe propagation of the ECN field over tunnels, it is reasonable to add requirements on operators, to ensure they configure their tunnels safely (where possible). Before stating these configuration requirements in Section 4, the factors that determine whether propagating ECN is feasible or desirable will be briefly introduced.

3.1. Feasibility of ECN Propagation between Tunnel Headers

In many cases shim header(s) and an outer IP header are always added to (or removed from) an inner IP packet as part of the same procedure. We call this a tightly coupled shim header. Processing the shim and outer together is often necessary because the shim(s) are not sufficient for packet forwarding in their own right; not unless complemented by an outer header. In these cases it will often be feasible for an implementation to propagate the ECN field between the IP headers.

In some cases a tunnel adds an outer IP header and a tightly coupled shim header to an inner header that is not an IP header, but that in turn encapsulates an IP header (or might encapsulate an IP header). For instance an inner Ethernet (or other link layer) header might encapsulate an inner IP header as its payload. We call this a tightly coupled shim over an encapsulating header.

Digging to arbitrary depths to find an inner IP header within an encapsulation is strictly a layering violation so it cannot be a required behaviour. Nonetheless, some tunnel endpoints already look within a L2 header for an IP header, for instance to map the Diffserv codepoint between an encapsulated IP header and an outer IP header

[RFC2983]. In such cases at least, it should be feasible to also (independently) propagate the ECN field between the same IP headers. Thus, access to the ECN field within an encapsulating header can be a useful and benign optimization. The guidelines in section 5 of [I-D.ietf-tsvwg-ecn-encap-guidelines] give the conditions for this layering violation to be benign.

3.2. Desirability of ECN Propagation between Tunnel Headers

Developers and network operators are encouraged to implement and deploy tunnel endpoints compliant with RFC 6040 (as updated by the present specification) in order to provide the benefits of wider ECN deployment [RFC8087]. Nonetheless, propagation of ECN between IP headers, whether separated by shim headers or not, has to be optional to implement and to use, because:

- o Legacy implementations of tunnels without any ECN support already exist
- o A network might be designed so that there is usually no bottleneck within the tunnel
- o If the tunnel endpoints would have to search within an L2 header to find an encapsulated IP header, it might not be worth the potential performance hit

4. Making a non-ECN Tunnel Ingress Safe by Configuration

Even when no specific attempt has been made to implement propagation of the ECN field at a tunnel ingress, it ought to be possible for the operator to render a tunnel ingress safe by configuration. The main safety concern is to disable (clear to zero) the ECN capability in the outer IP header at the ingress if the egress of the tunnel does not implement ECN logic to propagate any ECN markings into the packet forwarded beyond the tunnel. Otherwise the non-ECN egress could discard any ECN marking introduced within the tunnel, which would break all the ECN-based control loops that regulate the traffic load over the tunnel.

Therefore this specification updates RFC 6040 by inserting the following text at the end of section 4.3:

"

Whether or not an ingress implementation claims compliance with RFC 6040, RFC 4301 or RFC3168, when the outer tunnel header is IP (v4 or v6), if possible, the operator MUST configure the ingress to zero the outer ECN field in any of the following cases:

- * if it is known that the tunnel egress does not support any of the RFCs that define propagation of the ECN field (RFC 6040, RFC 4301 or the full functionality mode of RFC 3168)
- * or if the behaviour of the egress is not known or an egress with unknown behaviour might be dynamically paired with the ingress.
- * or if an IP header might be encapsulated within a non-IP header that the tunnel ingress is encapsulating, but the ingress does not inspect within the encapsulation.

For the avoidance of doubt, the above only concerns the outer IP header. The ingress MUST NOT alter the ECN field of the arriving IP header that will become the inner IP header.

In order that the network operator can comply with the above safety rules, even if an implementation of a tunnel ingress does not claim to support RFC 6040, RFC 4301 or the full functionality mode of RFC 3168:

- * it MUST NOT treat the former ToS octet (IPv4) or the former Traffic Class octet (IPv6) as a single 8-bit field, as the resulting linkage of ECN and Diffserv field propagation between inner and outer is not consistent with the definition of the 6-bit Diffserv field in [RFC2474] and [RFC3260];
- * it SHOULD be able to be configured to zero the ECN field of the outer header.

"

For instance, if a tunnel ingress with no ECN-specific logic had a configuration capability to refer to the last 2 bits of the old ToS Byte of the outer (e.g. with a 0x3 mask) and set them to zero, while also being able to allow the DSCP to be re-mapped independently, that would be sufficient to satisfy both the above implementation requirements.

There might be concern that the above "MUST NOT" makes compliant implementations non-compliant at a stroke. However, by definition it solely applies to equipment that provides Diffserv configuration. Any such Diffserv equipment that is configuring treatment of the former ToS octet (IPv4) or the former Traffic Class octet (IPv6) as a single 8-bit field must have always been non-compliant with the definition of the 6-bit Diffserv field in [RFC2474] and [RFC3260]. If a tunnel ingress does not have any ECN logic, copying the ECN field as a side-effect of copying the DSCP is a seriously unsafe bug

that risks breaking the feedback loops that regulate load on a tunnel.

Zeroing the outer ECN field of all packets in all circumstances would be safe, but it would not be sufficient to claim compliance with RFC 6040 because it would not meet the aim of introducing ECN support to tunnels (see Section 4.3 of [RFC6040]).

5. ECN Propagation and Fragmentation/Reassembly

The following requirements update RFC6040, which omitted handling of the ECN field during fragmentation or reassembly. These changes might alter how many ECN-marked packets are propagated by a tunnel that fragments packets, but this would not raise any backward compatibility issues:

If a tunnel ingress fragments a packet, it MUST set the outer ECN field of all the fragments to the same value as it would have set if it had not fragmented the packet.

Section 5.3 of [RFC3168] specifies ECN requirements for reassembly of sets of outer fragments [I-D.ietf-intarea-tunnels] into packets. The following two additional requirements apply at a tunnel egress:

- o During reassembly of outer fragments [I-D.ietf-intarea-tunnels], if the ECN fields of the outer headers being reassembled into a single packet consist of a mixture of Not-ECT and other ECN codepoints, the packet MUST be discarded.
- o If there is mix of ECT(0) and ECT(1) fragments, then the reassembled packet MUST be set to either ECT(0) or ECT(1). In this case, reassembly SHOULD take into account that the RFC series has so far ensured that ECT(0) and ECT(1) can either be considered equivalent, or they can provide 2 levels of congestion severity, where the ranking of severity from highest to lowest is CE, ECT(1), ECT(0) [RFC6040].

6. IP-in-IP Tunnels with Tightly Coupled Shim Headers

There follows a list of specifications of encapsulations with tightly coupled shim header(s), in rough chronological order. The list is confined to standards track or widely deployed protocols. The list is not necessarily exhaustive so, for the avoidance of doubt, the scope of RFC 6040 is defined in Section 3 and is not limited to this list.

- o PPTP (Point-to-Point Tunneling Protocol) [RFC2637];

- o L2TP (Layer 2 Tunneling Protocol), specifically L2TPv2 [RFC2661] and L2TPv3 [RFC3931], which not only includes all the L2-specific specializations of L2TP, but also derivatives such as the Keyed IPv6 Tunnel [RFC8159];
- o GRE (Generic Routing Encapsulation) [RFC2784] and NVGRE (Network Virtualization using GRE) [RFC7637];
- o GTP (GPRS Tunneling Protocol), specifically GTPv1 [GTPv1], GTP v1 User Plane [GTPv1-U], GTP v2 Control Plane [GTPv2-C];
- o Teredo [RFC4380];
- o CAPWAP (Control And Provisioning of Wireless Access Points) [RFC5415];
- o LISP (Locator/Identifier Separation Protocol) [RFC6830];
- o AMT (Automatic Multicast Tunneling) [RFC7450];
- o VXLAN (Virtual eXtensible Local Area Network) [RFC7348] and VXLAN-GPE [I-D.ietf-nvo3-vxlan-gpe];
- o The Network Service Header (NSH [RFC8300]) for Service Function Chaining (SFC);
- o Geneve [RFC8926];
- o GUE (Generic UDP Encapsulation) [I-D.ietf-intarea-gue];
- o Direct tunnelling of an IP packet within a UDP/IP datagram (see Section 3.1.11 of [RFC8085]);
- o TCP Encapsulation of IKE and IPsec Packets (see Section 12.5 of [RFC8229]).

Some of the listed protocols enable encapsulation of a variety of network layer protocols as inner and/or outer. This specification applies in the cases where there is an inner and outer IP header as described in Section 3. Otherwise [I-D.ietf-tsvwg-ecn-encap-guidelines] gives guidance on how to design propagation of ECN into other protocols that might encapsulate IP.

Where protocols in the above list need to be updated to specify ECN propagation and they are under IETF change control, update text is given in the following subsections. For those not under IETF control, it is RECOMMENDED that implementations of encapsulation and decapsulation comply with RFC 6040. It is also RECOMMENDED that

their specifications are updated to add a requirement to comply with RFC 6040 (as updated by the present document).

PPTP is not under the change control of the IETF, but it has been documented in an informational RFC [RFC2637]. However, there is no need for the present specification to update PPTP because L2TP has been developed as a standardized replacement.

NVGRE is not under the change control of the IETF, but it has been documented in an informational RFC [RFC7637]. NVGRE is a specific use-case of GRE (it re-purposes the key field from the initial specification of GRE [RFC1701] as a Virtual Subnet ID). Therefore the text that updates GRE in Section 6.1.2 below is also intended to update NVGRE.

Although the definition of the various GTP shim headers is under the control of the 3GPP, it is hard to determine whether the 3GPP or the IETF controls standardization of the `_process_` of adding both a GTP and an IP header to an inner IP header. Nonetheless, the present specification is provided so that the 3GPP can refer to it from any of its own specifications of GTP and IP header processing.

The specification of CAPWAP already specifies RFC 3168 ECN propagation and ECN capability negotiation. Without modification the CAPWAP specification already interworks with the backward compatible updates to RFC 3168 in RFC 6040.

LISP made the ECN propagation procedures in RFC 3168 mandatory from the start. RFC 3168 has since been updated by RFC 6040, but the changes are backwards compatible so there is still no need for LISP tunnel endpoints to negotiate their ECN capabilities.

VXLAN is not under the change control of the IETF but it has been documented in an informational RFC. In contrast, VXLAN-GPE (Generic Protocol Extension) is being documented under IETF change control. It is RECOMMENDED that VXLAN and VXLAN-GPE implementations comply with RFC 6040 when the VXLAN header is inserted between (or removed from between) IP headers. The authors of any future update to these specifications are encouraged to add a requirement to comply with RFC 6040 as updated by the present specification.

The Network Service Header (NSH [RFC8300]) has been defined as a shim-based encapsulation to identify the Service Function Path (SFP) in the Service Function Chaining (SFC) architecture [RFC7665]. A proposal has been made for the processing of ECN when handling transport encapsulation [I-D.ietf-sfc-nsh-ecn-support].

The specifications of Geneve and GUE already refer to RFC 6040 for ECN encapsulation.

Section 3.1.11 of RFC 8085 already explains that a tunnel that encapsulates an IP header within a UDP/IP datagram needs to follow RFC 6040 when propagating the ECN field between inner and outer IP headers. The requirements in Section 4 update RFC 6040, and hence implicitly update the UDP usage guidelines in RFC 8085 to add the important but previously unstated requirement that, if the UDP tunnel egress does not, or might not, support ECN propagation, a UDP tunnel ingress has to clear the outer IP ECN field to 0b00, e.g. by configuration.

Section 12.5 of TCP Encapsulation of IKE and IPsec Packets [RFC8229] already recommends the compatibility mode of RFC 6040 in this case, because there is not a one-to-one mapping between inner and outer packets.

6.1. Specific Updates to Protocols under IETF Change Control

6.1.1. L2TP (v2 and v3) ECN Extension

The L2TP terminology used here is defined in [RFC2661] and [RFC3931].

L2TPv3 [RFC3931] is used as a shim header between any packet-switched network (PSN) header (e.g. IPv4, IPv6, MPLS) and many types of layer 2 (L2) header. The L2TPv3 shim header encapsulates an L2-specific sub-layer then an L2 header that is likely to contain an inner IP header (v4 or v6). Then this whole stack of headers can be encapsulated optionally within an outer UDP header then an outer PSN header that is typically IP (v4 or v6).

L2TPv2 is used as a shim header between any PSN header and a PPP header, which is in turn likely to encapsulate an IP header.

Even though these shims are rather fat (particularly in the case of L2TPv3), they still fit the definition of a tightly coupled shim header over an encapsulating header (Section 3.1), because all the headers encapsulating the L2 header are added (or removed) together. L2TPv2 and L2TPv3 are therefore within the scope of RFC 6040, as updated by Section 3 above.

L2TP maintainers are RECOMMENDED to implement the ECN extension to L2TPv2 and L2TPv3 defined in Section 6.1.1.2 below, in order to provide the benefits of ECN [RFC8087], whenever a node within an L2TP tunnel becomes the bottleneck for an end-to-end traffic flow.

6.1.1.1. Safe Configuration of a 'Non-ECN' Ingress LCCE

The following text is appended to both Section 5.3 of [RFC2661] and Section 4.5 of [RFC3931] as an update to the base L2TPv2 and L2TPv3 specifications:

The operator of an LCCE that does not support the ECN Extension in Section 6.1.1.2 of RFCXXXX MUST follow the configuration requirements in Section 4 of RFCXXXX to ensure it clears the outer IP ECN field to 0b00 when the outer PSN header is IP (v4 or v6). {RFCXXXX refers to the present document so it will need to be inserted by the RFC Editor}

In particular, for an LCCE implementation that does not support the ECN Extension, this means that configuration of how it propagates the ECN field between inner and outer IP headers MUST be independent of any configuration of the Diffserv extension of L2TP [RFC3308].

6.1.1.2. ECN Extension for L2TP (v2 or v3)

When the outer PSN header and the payload inside the L2 header are both IP (v4 or v6), to comply with RFC 6040, an LCCE will follow the rules for propagation of the ECN field at ingress and egress in Section 4 of RFC 6040 [RFC6040].

Before encapsulating any data packets, RFC 6040 requires an ingress LCCE to check that the egress LCCE supports ECN propagation as defined in RFC 6040 or one of its compatible predecessors ([RFC4301] or the full functionality mode of [RFC3168]). If the egress supports ECN propagation, the ingress LCCE can use the normal mode of encapsulation (copying the ECN field from inner to outer). Otherwise, the ingress LCCE has to use compatibility mode [RFC6040] (clearing the outer IP ECN field to 0b00).

An LCCE can determine the remote LCCE's support for ECN either statically (by configuration) or by dynamic discovery during setup of each control connection between the LCCEs, using the Capability AVP defined in Section 6.1.1.2.1 below.

Where the outer PSN header is some protocol other than IP that supports ECN, the appropriate ECN propagation specification will need to be followed, e.g. "Explicit Congestion Marking in MPLS" [RFC5129]. Where no specification exists for ECN propagation by a particular PSN, [I-D.ietf-tsvwg-ecn-encap-guidelines] gives general guidance on how to design ECN propagation into a protocol that encapsulates IP.

6.1.1.2.1. LCCE Capability AVP for ECN Capability Negotiation

The LCCE Capability Attribute-Value Pair (AVP) defined here has Attribute Type ZZ. The Attribute Value field for this AVP is a bit-mask with the following 16-bit format:

```

      0                               1
      0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5
+---+---+---+---+---+---+---+---+---+
|X X X X X X X X X X X X X X X E|
+---+---+---+---+---+---+---+---+

```

Figure 1: Value Field for the LCCE Capability Attribute

This AVP MAY be present in the following message types: SCCRQ and SCCRP (Start-Control-Connection-Request and Start-Control-Connection-Reply). This AVP MAY be hidden (the H-bit set to 0 or 1) and is optional (M-bit not set). The length (before hiding) of this AVP MUST be 8 octets. The Vendor ID is the IETF Vendor ID of 0.

Bit 15 of the Value field of the LCCE Capability AVP is defined as the ECN Capability flag (E). When the ECN Capability flag is set to 1, it indicates that the sender supports ECN propagation. When the ECN Capability flag is cleared to zero, or when no LCCE Capability AVP is present, it indicates that the sender does not support ECN propagation. All the other bits are reserved. They MUST be cleared to zero when sent and ignored when received or forwarded.

An LCCE initiating a control connection will send a Start-Control-Connection-Request (SCCRQ) containing an LCCE Capability AVP with the ECN Capability flag set to 1. If the tunnel terminator supports ECN, it will return a Start-Control-Connection-Reply (SCCRP) that also includes an LCCE Capability AVP with the ECN Capability flag set to 1. Then, for any sessions created by that control connection, both ends of the tunnel can use the normal mode of RFC 6040, i.e. it can copy the IP ECN field from inner to outer when encapsulating data packets.

If, on the other hand, the tunnel terminator does not support ECN it will ignore the ECN flag in the LCCE Capability AVP and send an SCCRP to the tunnel initiator without a Capability AVP (or with a Capability AVP but with the ECN Capability flag cleared to zero). The tunnel initiator interprets the absence of the ECN Capability flag in the SCCRP as an indication that the tunnel terminator is incapable of supporting ECN. When encapsulating data packets for any sessions created by that control connection, the tunnel initiator will then use the compatibility mode of RFC 6040 to clear the ECN field of the outer IP header to 0b00.

If the tunnel terminator does not support this ECN extension, the network operator is still expected to configure it to comply with the safety provisions set out in Section 6.1.1.1 above, when it acts as an ingress LCCE.

6.1.2. GRE

The GRE terminology used here is defined in [RFC2784]. GRE is often used as a tightly coupled shim header between IP headers. Sometimes the GRE shim header encapsulates an L2 header, which might in turn encapsulate an IP header. Therefore GRE is within the scope of RFC 6040 as updated by Section 3 above.

GRE tunnel endpoint maintainers are RECOMMENDED to support [RFC6040] as updated by the present specification, in order to provide the benefits of ECN [RFC8087] whenever a node within a GRE tunnel becomes the bottleneck for an end-to-end IP traffic flow tunnelled over GRE using IP as the delivery protocol (outer header).

GRE itself does not support dynamic set-up and configuration of tunnels. However, control plane protocols such as Mobile IPv4 (MIP4) [RFC5944], Mobile IPv6 (MIP6) [RFC6275], Proxy Mobile IP (PMIP) [RFC5845] and IKEv2 [RFC7296] are sometimes used to set up GRE tunnels dynamically.

When these control protocols set up IP-in-IP or IPSec tunnels, it is likely that they propagate the ECN field as defined in RFC 6040 or one of its compatible predecessors (RFC 4301 or the full functionality mode of RFC 3168). However, if they use a GRE encapsulation, this presumption is less sound.

Therefore, If the outer delivery protocol is IP (v4 or v6) the operator is obliged to follow the safe configuration requirements in Section 4 above. Section 6.1.2.1 below updates the base GRE specification with this requirement, to emphasize its importance.

Where the delivery protocol is some protocol other than IP that supports ECN, the appropriate ECN propagation specification will need to be followed, e.g Explicit Congestion Marking in MPLS [RFC5129]. Where no specification exists for ECN propagation by a particular PSN, [I-D.ietf-tsvwg-ecn-encap-guidelines] gives more general guidance on how to propagate ECN to and from protocols that encapsulate IP.

6.1.2.1. Safe Configuration of a 'Non-ECN' GRE Ingress

The following text is appended to Section 3 of [RFC2784] as an update to the base GRE specification:

The operator of a GRE tunnel ingress MUST follow the configuration requirements in Section 4 of RFCXXXX when the outer delivery protocol is IP (v4 or v6). {RFCXXXX refers to the present document so it will need to be inserted by the RFC Editor}

6.1.3. Teredo

Teredo [RFC4380] provides a way to tunnel IPv6 over an IPv4 network, with a UDP-based shim header between the two.

For Teredo tunnel endpoints to provide the benefits of ECN, the Teredo specification would have to be updated to include negotiation of the ECN capability between Teredo tunnel endpoints. Otherwise it would be unsafe for a Teredo tunnel ingress to copy the ECN field to the IPv6 outer.

It is believed that current implementations do not support propagation of ECN, but that they do safely zero the ECN field in the outer IPv6 header. However the specification does not mention anything about this.

To make existing Teredo deployments safe, it would be possible to add ECN capability negotiation to those that are subject to remote OS update. However, for those implementations not subject to remote OS update, it will not be feasible to require them to be configured correctly, because Teredo tunnel endpoints are generally deployed on hosts.

Therefore, until ECN support is added to the specification of Teredo, the only feasible further safety precaution available here is to update the specification of Teredo implementations with the following text, as a new section 5.1.3:

"5.1.3 Safe 'Non-ECN' Teredo Encapsulation

A Teredo tunnel ingress implementation that does not support ECN propagation as defined in RFC 6040 or one of its compatible predecessors (RFC 4301 or the full functionality mode of RFC 3168) MUST zero the ECN field in the outer IPv6 header."

6.1.4. AMT

Automatic Multicast Tunneling (AMT [RFC7450]) is a tightly coupled shim header that encapsulates an IP packet and is itself encapsulated within a UDP/IP datagram. Therefore AMT is within the scope of RFC 6040 as updated by Section 3 above.

AMT tunnel endpoint maintainers are RECOMMENDED to support [RFC6040] as updated by the present specification, in order to provide the benefits of ECN [RFC8087] whenever a node within an AMT tunnel becomes the bottleneck for an IP traffic flow tunnelled over AMT.

To comply with RFC 6040, an AMT relay and gateway will follow the rules for propagation of the ECN field at ingress and egress respectively, as described in Section 4 of RFC 6040 [RFC6040].

Before encapsulating any data packets, RFC 6040 requires an ingress AMT relay to check that the egress AMT gateway supports ECN propagation as defined in RFC 6040 or one of its compatible predecessors (RFC 4301 or the full functionality mode of RFC 3168). If the egress gateway supports ECN, the ingress relay can use the normal mode of encapsulation (copying the IP ECN field from inner to outer). Otherwise, the ingress relay has to use compatibility mode, which means it has to clear the outer ECN field to zero [RFC6040].

An AMT tunnel is created dynamically (not manually), so the relay will need to determine the remote gateway's support for ECN using the ECN capability declaration defined in Section 6.1.4.2 below.

6.1.4.1. Safe Configuration of a 'Non-ECN' Ingress AMT Relay

The following text is appended to Section 4.2.2 of [RFC7450] as an update to the AMT specification:

The operator of an AMT relay that does not support RFC 6040 or one of its compatible predecessors (RFC 4301 or the full functionality mode of RFC 3168) MUST follow the configuration requirements in Section 4 of RFCXXXX to ensure it clears the outer IP ECN field to zero. {RFCXXXX refers to the present document so it will need to be inserted by the RFC Editor}

6.1.4.2. ECN Capability Declaration of an AMT Gateway

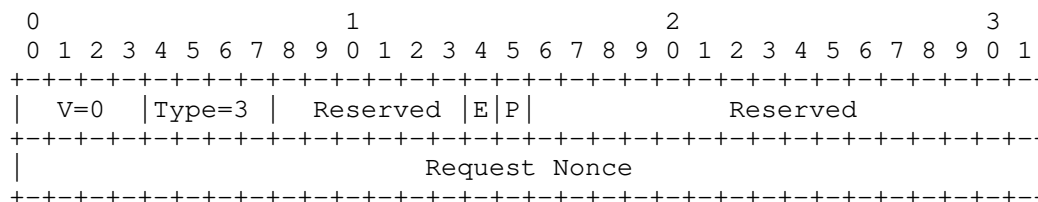


Figure 2: Updated AMT Request Message Format

Bit 14 of the AMT Request Message counting from 0 (or bit 7 of the Reserved field counting from 1) is defined here as the AMT Gateway ECN Capability flag (E), as shown in Figure 2. The definitions of all other fields in the AMT Request Message are unchanged from RFC 7450.

When the E flag is set to 1, it indicates that the sender of the message supports RFC 6040 ECN propagation. When it is cleared to zero, it indicates the sender of the message does not support RFC 6040 ECN propagation. An AMT gateway "that supports RFC 6040 ECN propagation" means one that propagates the ECN field to the forwarded data packet based on the combination of arriving inner and outer ECN fields, as defined in Section 4 of RFC 6040.

The other bits of the Reserved field remain reserved. They will continue to be cleared to zero when sent and ignored when either received or forwarded, as specified in Section 5.1.3.3. of RFC 7450.

An AMT gateway that does not support RFC 6040 MUST NOT set the E flag of its Request Message to 1.

An AMT gateway that supports RFC 6040 ECN propagation MUST set the E flag of its Relay Discovery Message to 1.

The action of the corresponding AMT relay that receives a Request message with the E flag set to 1 depends on whether the relay itself supports RFC 6040 ECN propagation:

- o If the relay supports RFC 6040 ECN propagation, it will store the ECN capability of the gateway along with its address. Then whenever it tunnels datagrams towards this gateway, it MUST use the normal mode of RFC 6040 to propagate the ECN field when encapsulating datagrams (i.e. it copies the IP ECN field from inner to outer).

- o If the discovered AMT relay does not support RFC 6040 ECN propagation, it will ignore the E flag in the Reserved field, as per section 5.1.3.3. of RFC 7450.

If the AMT relay does not support RFC 6040 ECN propagation, the network operator is still expected to configure it to comply with the safety provisions set out in Section 6.1.4.1 above.

7. IANA Considerations

IANA is requested to assign the following L2TP Control Message Attribute Value Pair:

Attribute Type	Description	Reference
ZZ	ECN Capability	RFCXXXX

[TO BE REMOVED: This registration should take place at the following location: <https://www.iana.org/assignments/l2tp-parameters/l2tp-parameters.xhtml>]

8. Security Considerations

The Security Considerations in [RFC6040] and [I-D.ietf-tsvwg-ecn-encap-guidelines] apply equally to the scope defined for the present specification.

9. Comments Solicited

Comments and questions are encouraged and very welcome. They can be addressed to the IETF Transport Area working group mailing list <tsvwg@ietf.org>, and/or to the authors.

10. Acknowledgements

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Sliding Window Random Linear Code (RLC) Forward Erasure Correction (FEC)
Schemes for FECFRAME
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Abstract

This document describes two fully-specified Forward Erasure Correction (FEC) Schemes for Sliding Window Random Linear Codes (RLC), one for RLC over the Galois Field (A.K.A. Finite Field) $GF(2)$, a second one for RLC over the Galois Field $GF(2^{8})$, each time with the possibility of controlling the code density. They can protect arbitrary media streams along the lines defined by FECFRAME extended to sliding window FEC codes. These sliding window FEC codes rely on an encoding window that slides over the source symbols, generating new repair symbols whenever needed. Compared to block FEC codes, these sliding window FEC codes offer key advantages with real-time flows in terms of reduced FEC-related latency while often providing improved packet erasure recovery capabilities.

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1. Introduction

Application-Level Forward Erasure Correction (AL-FEC) codes, or simply FEC codes, are a key element of communication systems. They are used to recover from packet losses (or erasures) during content delivery sessions to a potentially large number of receivers (multicast/broadcast transmissions). This is the case with the FLUTE/ALC protocol [RFC6726] when used for reliable file transfers over lossy networks, and the FECFRAME protocol [RFC6363] when used for reliable continuous media transfers over lossy networks.

The present document only focuses on the FECFRAME protocol, used in multicast/broadcast delivery mode, in particular for contents that feature stringent real-time constraints: each source packet has a maximum validity period after which it will not be considered by the destination application.

1.1. Limits of Block Codes with Real-Time Flows

With FECFRAME, there is a single FEC encoding point (either an end-host/server (source) or a middlebox) and a single FEC decoding point per receiver (either an end-host (receiver) or middlebox). In this context, currently standardized AL-FEC codes for FECFRAME like Reed-Solomon [RFC6865], LDPC-Staircase [RFC6816], or Raptor/RaptorQ [RFC6681], are all linear block codes: they require the data flow to be segmented into blocks of a predefined maximum size.

To define this block size, it is required to find an appropriate balance between robustness and decoding latency: the larger the block size, the higher the robustness (e.g., in case of long packet erasure bursts), but also the higher the maximum decoding latency (i.e., the maximum time required to recover a lost (erased) packet thanks to FEC protection). Therefore, with a multicast/broadcast session where different receivers experience different packet loss rates, the block size should be chosen by considering the worst communication conditions one wants to support, but without exceeding the desired maximum decoding latency. This choice then impacts the FEC-related latency of all receivers, even those experiencing a good communication quality, since no FEC encoding can happen until all the source data of the block is available at the sender, which directly depends on the block size.

1.2. Lower Latency and Better Protection of Real-Time Flows with the Sliding Window RLC Codes

This document introduces two fully-specified FEC Schemes that do not follow the block code approach: the Sliding Window Random Linear Codes (RLC) over either Galois Fields (A.K.A. Finite Fields) $GF(2)$ (the "binary case") or $GF(2^{8})$, each time with the possibility of controlling the code density. These FEC Schemes are used to protect arbitrary media streams along the lines defined by FECFRAME extended to sliding window FEC codes [fecframe-ext]. These FEC Schemes, and more generally Sliding Window FEC codes, are recommended for instance, with media that feature real-time constraints sent within a multicast/broadcast session [Roca17].

The RLC codes belong to the broad class of sliding-window AL-FEC codes (A.K.A. convolutional codes) [RFC8406]. The encoding process is based on an encoding window that slides over the set of source packets (in fact source symbols as we will see in Section 3.2), this window being either of fixed size or variable size (A.K.A. an elastic window). Repair symbols are generated on-the-fly, by computing a random linear combination of the source symbols present in the current encoding window, and passed to the transport layer.

At the receiver, a linear system is managed from the set of received source and repair packets. New variables (representing source symbols) and equations (representing the linear combination carried by each repair symbol received) are added upon receiving new packets. Variables and the equations they are involved in are removed when they are too old with respect to their validity period (real-time constraints). Lost source symbols are then recovered thanks to this linear system whenever its rank permits to solve it (at least partially).

The protection of any multicast/broadcast session needs to be dimensioned by considering the worst communication conditions one wants to support. This is also true with RLC (more generally any sliding window) code. However, the receivers experiencing a good to medium communication quality will observe a reduced FEC-related latency compared to block codes [Roca17] since an isolated lost source packet is quickly recovered with the following repair packet. On the opposite, with a block code, recovering an isolated lost source packet always requires waiting for the first repair packet to arrive after the end of the block. Additionally, under certain situations (e.g., with a limited FEC-related latency budget and with constant bitrate transmissions after FECFRAME encoding), sliding window codes can more efficiently achieve a target transmission quality (e.g., measured by the residual loss after FEC decoding) by sending fewer repair packets (i.e., higher code rate) than block codes.

1.3. Small Transmission Overheads with the Sliding Window RLC FEC Scheme

The Sliding Window RLC FEC Scheme is designed to limit the packet header overhead. The main requirement is that each repair packet header must enable a receiver to reconstruct the set of source symbols plus the associated coefficients used during the encoding process. In order to minimize packet overhead, the set of source symbols in the encoding window as well as the set of coefficients over $GF(2^m)$ (where m is 1 or 8, depending on the FEC Scheme) used in the linear combination are not individually listed in the repair packet header. Instead, each FEC Repair Packet header contains:

- o the Encoding Symbol Identifier (ESI) of the first source symbol in the encoding window as well as the number of symbols (since this number may vary with a variable size, elastic window). These two pieces of information enable each receiver to reconstruct the set of source symbols considered during encoding, the only constraint being that there cannot be any gap;
- o the seed and density threshold parameters used by a coding coefficients generation function (Section 3.6). These two pieces

of information enable each receiver to generate the same set of coding coefficients over $GF(2^m)$ as the sender;

Therefore, no matter the number of source symbols present in the encoding window, each FEC Repair Packet features a fixed 64-bit long header, called Repair FEC Payload ID (Figure 8). Similarly, each FEC Source Packet features a fixed 32-bit long trailer, called Explicit Source FEC Payload ID (Figure 6), that contains the ESI of the first source symbol (Section 3.2).

1.4. Document Organization

This fully-specified FEC Scheme follows the structure required by [RFC6363], section 5.6. "FEC Scheme Requirements", namely:

3. Procedures: This section describes procedures specific to this FEC Scheme, namely: RLC parameters derivation, ADUI and source symbols mapping, pseudo-random number generator, and coding coefficients generation function;
4. Formats and Codes: This section defines the Source FEC Payload ID and Repair FEC Payload ID formats, carrying the signaling information associated to each source or repair symbol. It also defines the FEC Framework Configuration Information (FFCI) carrying signaling information for the session;
5. FEC Code Specification: Finally this section provides the code specification.

2. Definitions and Abbreviations

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

This document uses the following definitions and abbreviations:

a^b a to the power of b

$GF(q)$ denotes a finite field (also known as the Galois Field) with q elements. We assume that $q = 2^m$ in this document

m defines the length of the elements in the finite field, in bits.

In this document, m is equal to 1 or 8

ADU: Application Data Unit

ADUI: Application Data Unit Information (includes the F, L and padding fields in addition to the ADU)

E: size of an encoding symbol (i.e., source or repair symbol), assumed fixed (in bytes)

br_in: transmission bitrate at the input of the FECFRAME sender, assumed fixed (in bits/s)
br_out: transmission bitrate at the output of the FECFRAME sender, assumed fixed (in bits/s)
max_lat: maximum FEC-related latency within FECFRAME (a decimal number expressed in seconds)
cr: RLC coding rate, ratio between the total number of source symbols and the total number of source plus repair symbols
ew_size: encoding window current size at a sender (in symbols)
ew_max_size: encoding window maximum size at a sender (in symbols)
dw_max_size: decoding window maximum size at a receiver (in symbols)
ls_max_size: linear system maximum size (or width) at a receiver (in symbols)
WSR: window size ratio parameter used to derive ew_max_size (encoder) and ls_max_size (decoder).
PRNG: pseudo-random number generator
TinyMT32: PRNG used in this specification.
DT: coding coefficients density threshold, an integer between 0 and 15 (inclusive) the controls the fraction of coefficients that are non zero

3. Common Procedures

This section introduces the procedures that are used by these FEC Schemes.

3.1. Codec Parameters

A codec implementing the Sliding Window RLC FEC Scheme relies on several parameters:

Maximum FEC-related latency budget, max_lat (a decimal number expressed in seconds) with real-time flows:
a source ADU flow can have real-time constraints, and therefore any FECFRAME related operation should take place within the validity period of each ADU (Appendix D describes an exception to this rule). When there are multiple flows with different real-time constraints, we consider the most stringent constraints (see [RFC6363], Section 10.2, item 6, for recommendations when several flows are globally protected). The maximum FEC-related latency budget, max_lat, accounts for all sources of latency added by FEC encoding (at a sender) and FEC decoding (at a receiver). Other sources of latency (e.g., added by network communications) are out of scope and must be considered separately (said differently, they have already been deducted from max_lat). max_lat can be regarded as the latency budget permitted for all FEC-related operations. This is an input parameter that enables a FECFRAME sender to derive other internal parameters (see Appendix C);

Encoding window current (resp. maximum) size, `ew_size` (resp. `ew_max_size`) (in symbols):

at a FECFRAME sender, during FEC encoding, a repair symbol is computed as a linear combination of the `ew_size` source symbols present in the encoding window. The `ew_max_size` is the maximum size of this window, while `ew_size` is the current size. For example, in the common case at session start, upon receiving new source ADUs, the `ew_size` progressively increases until it reaches its maximum value, `ew_max_size`. We have:

$$0 < \text{ew_size} \leq \text{ew_max_size}$$

Decoding window maximum size, `dw_max_size` (in symbols): at a FECFRAME receiver, `dw_max_size` is the maximum number of received or lost source symbols that are still within their latency budget;

Linear system maximum size, `ls_max_size` (in symbols): at a FECFRAME receiver, the linear system maximum size, `ls_max_size`, is the maximum number of received or lost source symbols in the linear system (i.e., the variables). It SHOULD NOT be smaller than `dw_max_size` since it would mean that, even after receiving a sufficient number of FEC Repair Packets, a lost ADU may not be recovered just because the associated source symbols have been prematurely removed from the linear system, which is usually counter-productive. On the opposite, the linear system MAY grow beyond the `dw_max_size` (Appendix D);

Symbol size, `E` (in bytes): the `E` parameter determines the source and repair symbol sizes (necessarily equal). This is an input parameter that enables a FECFRAME sender to derive other internal parameters, as explained below. An implementation at a sender MUST fix the `E` parameter and MUST communicate it as part of the FEC Scheme-Specific Information (Section 4.1.1.2).

Code rate, `cr`: The code rate parameter determines the amount of redundancy added to the flow. More precisely the `cr` is the ratio between the total number of source symbols and the total number of source plus repair symbols and by definition: $0 < \text{cr} \leq 1$. This is an input parameter that enables a FECFRAME sender to derive other internal parameters, as explained below. However, there is no need to communicate the `cr` parameter per se (it's not required to process a repair symbol at a receiver). This code rate parameter can be static. However, in specific use-cases (e.g., with unicast transmissions in presence of a feedback mechanism that estimates the communication quality, out of scope of FECFRAME), the code rate may be adjusted dynamically.

Appendix C proposes non normative techniques to derive those parameters, depending on the use-case specificities.

3.2. ADU, ADUI and Source Symbols Mappings

At a sender, an ADU coming from the application is not directly mapped to source symbols. When multiple source flows (e.g., media streams) are mapped onto the same FECFRAME instance, each flow is assigned its own Flow ID value (see below). This Flow ID is then prepended to each ADU before FEC encoding. This way, FEC decoding at a receiver also recovers this Flow ID and the recovered ADU can be assigned to the right source flow (note that the 5-tuple used to identify the right source flow of a received ADU is absent with a recovered ADU since it is not FEC protected).

Additionally, since ADUs are of variable size, padding is needed so that each ADU (with its flow identifier) contribute to an integral number of source symbols. This requires adding the original ADU length to each ADU before doing FEC encoding. Because of these requirements, an intermediate format, the ADUI, or ADU Information, is considered [RFC6363].

For each incoming ADU, an ADUI MUST be created as follows. First of all, 3 bytes are prepended (Figure 1):

Flow ID (F) (8-bit field): this unsigned byte contains the integer identifier associated to the source ADU flow to which this ADU belongs. It is assumed that a single byte is sufficient, which implies that no more than 256 flows will be protected by a single FECFRAME session instance.

Length (L) (16-bit field): this unsigned integer contains the length of this ADU, in network byte order (i.e., big endian). This length is for the ADU itself and does not include the F, L, or Pad fields.

Then, zero padding is added to the ADU if needed:

Padding (Pad) (variable size field): this field contains zero padding to align the F, L, ADU and padding up to a size that is multiple of E bytes (i.e., the source and repair symbol length).

The data unit resulting from the ADU and the F, L, and Pad fields is called ADUI. Since ADUs can have different sizes, this is also the case for ADUIs. However, an ADUI always contributes to an integral number of source symbols.

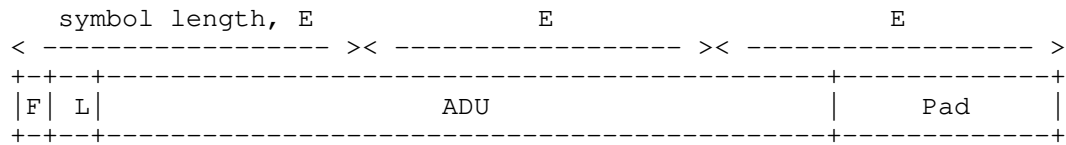


Figure 1: ADUI Creation example (here 3 source symbols are created for this ADUI).

Note that neither the initial 3 bytes nor the optional padding are sent over the network. However, they are considered during FEC encoding, and a receiver who lost a certain FEC Source Packet (e.g., the UDP datagram containing this FEC Source Packet when UDP is used as the transport protocol) will be able to recover the ADUI if FEC decoding succeeds. Thanks to the initial 3 bytes, this receiver will get rid of the padding (if any) and identify the corresponding ADU flow.

3.3. Encoding Window Management

Source symbols and the corresponding ADUs are removed from the encoding window:

- o when the sliding encoding window has reached its maximum size, `ew_max_size`. In that case the oldest symbol MUST be removed before adding a new symbol, so that the current encoding window size always remains inferior or equal to the maximum size: `ew_size <= ew_max_size`;
- o when an ADU has reached its maximum validity duration in case of a real-time flow. When this happens, all source symbols corresponding to the ADUI that expired SHOULD be removed from the encoding window;

Source symbols are added to the sliding encoding window each time a new ADU arrives, once the ADU-to-source symbols mapping has been performed (Section 3.2). The current size of the encoding window, `ew_size`, is updated after adding new source symbols. This process may require to remove old source symbols so that: `ew_size <= ew_max_size`.

Note that a FEC codec may feature practical limits in the number of source symbols in the encoding window (e.g., for computational complexity reasons). This factor may further limit the `ew_max_size` value, in addition to the maximum FEC-related latency budget (Section 3.1).

3.4. Source Symbol Identification

Each source symbol is identified by an Encoding Symbol ID (ESI), an unsigned integer. The ESI of source symbols MUST start with value 0 for the first source symbol and MUST be managed sequentially. Wrapping to zero happens after reaching the maximum value made possible by the ESI field size (this maximum value is FEC Scheme dependant, for instance, $2^{32}-1$ with FEC Schemes XXX and YYY).

No such consideration applies to repair symbols.

3.5. Pseudo-Random Number Generator (PRNG)

In order to compute coding coefficients (see Section 3.6), the RLC FEC Schemes rely on the TinyMT32 PRNG defined in [tinymt32] with two additional functions defined in this section.

This PRNG MUST first be initialized with a 32-bit unsigned integer, used as a seed, with:

```
void tinymt32_init (tinymt32_t * s, uint32_t seed);
```

With the FEC Schemes defined in this document, the seed is in practice restricted to a value between 0 and 0xFFFF inclusive (note that this PRNG accepts a seed value equal to 0), since this is the Repair_Key 16-bit field value of the Repair FEC Payload ID (Section 4.1.3). In practice, how to manage the seed and Repair_Key values (both are equal) is left to the implementer, using a monotonically increasing counter being one possibility (Section 6.1). In addition to the seed, this function takes as parameter a pointer to an instance of a tinymt32_t structure that is used to keep the internal state of the PRNG.

Then, each time a new pseudo-random integer between 0 and 15 inclusive (4-bit pseudo-random integer) is needed, the following function is used:

```
uint32_t tinymt32_rand16 (tinymt32_t * s);
```

This function takes as parameter a pointer to the same tinymt32_t structure (that is left unchanged between successive calls to the function).

Similarly, each time a new pseudo-random integer between 0 and 255 inclusive (8-bit pseudo-random integer) is needed, the following function is used:

```
uint32_t tinymt32_rand256 (tinymt32_t * s);
```


These two functions keep respectively the 4 or 8 less significant bits of the 32-bit pseudo-random number generated by the `tinymt32_generate_uint32()` function of [tinymt32]. This is done by computing the result of a binary AND between the `tinymt32_generate_uint32()` output and respectively the `0xF` or `0xFF` constants, using 32-bit unsigned integer operations. Figure 2 shows a possible implementation. This is a C language implementation, written for C99 [C99]. Test results discussed in Appendix B show that this simple technique, applied to this PRNG, is in line with the RLC FEC Schemes needs.

```
<CODE BEGINS>
/**
 * This function outputs a pseudo-random integer in [0 .. 15] range.
 *
 * @param s      pointer to tinymt internal state.
 * @return       unsigned integer between 0 and 15 inclusive.
 */
uint32_t tinymt32_rand16(tinymt32_t *s)
{
    return (tinymt32_generate_uint32(s) & 0xF);
}

/**
 * This function outputs a pseudo-random integer in [0 .. 255] range.
 *
 * @param s      pointer to tinymt internal state.
 * @return       unsigned integer between 0 and 255 inclusive.
 */
uint32_t tinymt32_rand256(tinymt32_t *s)
{
    return (tinymt32_generate_uint32(s) & 0xFF);
}
<CODE ENDS>
```

Figure 2: 4-bit and 8-bit mapping functions for TinyMT32

Any implementation of this PRNG MUST have the same output as that provided by the reference implementation of [tinymt32]. In order to increase the compliancy confidence, three criteria are proposed: the one described in [tinymt32] (for the TinyMT32 32-bit unsigned integer generator), and the two others detailed in Appendix A (for the mapping to 4-bit and 8-bit intervals). Because of the way the mapping functions work, it is unlikely that an implementation that fulfills the first criterion fails to fulfill the two others.

3.6. Coding Coefficients Generation Function

The coding coefficients, used during the encoding process, are generated at the RLC encoder by the `generate_coding_coefficients()` function each time a new repair symbol needs to be produced. The fraction of coefficients that are non zero (i.e., the density) is controlled by the DT (Density Threshold) parameter. DT has values between 0 (the minimum value) and 15 (the maximum value), and the average probability of having a non zero coefficient equals $(DT + 1) / 16$. In particular, when DT equals 15 the function guaranties that all coefficients are non zero (i.e., maximum density).

These considerations apply to both the RLC over GF(2) and RLC over GF($2^{^8}$), the only difference being the value of the m parameter. With the RLC over GF(2) FEC Scheme (Section 5), m is equal to 1. With RLC over GF($2^{^8}$) FEC Scheme (Section 4), m is equal to 8.

Figure 3 shows the reference `generate_coding_coefficients()` implementation. This is a C language implementation, written for C99 [C99].

```
<CODE BEGINS>
#include <string.h>

/*
 * Fills in the table of coding coefficients (of the right size)
 * provided with the appropriate number of coding coefficients to
 * use for the repair symbol key provided.
 *
 * (in) repair_key    key associated to this repair symbol. This
 *                    parameter is ignored (useless) if m=1 and dt=15
 * (in/out) cc_tab    pointer to a table of the right size to store
 *                    coding coefficients. All coefficients are
 *                    stored as bytes, regardless of the m parameter,
 *                    upon return of this function.
 * (in) cc_nb         number of entries in the cc_tab table. This
 *                    value is equal to the current encoding window
 *                    size.
 * (in) dt            integer between 0 and 15 (inclusive) that
 *                    controls the density. With value 15, all
 *                    coefficients are guaranteed to be non zero
 *                    (i.e. equal to 1 with GF(2) and equal to a
 *                    value in {1,... 255} with GF( $2^{^8}$ )), otherwise
 *                    a fraction of them will be 0.
 * (in) m             Finite Field GF( $2^{^m}$ ) parameter. In this
 *                    document only values 1 and 8 are considered.
 * (out)              returns 0 in case of success, an error code
 *                    different than 0 otherwise.
```

```
*/
int generate_coding_coefficients (uint16_t  repair_key,
                                uint8_t*   cc_tab,
                                uint16_t   cc_nb,
                                uint8_t    dt,
                                uint8_t    m)
{
    uint32_t    i;
    tinymt32_t  s;    /* PRNG internal state */

    if (dt > 15) {
        return -1; /* error, bad dt parameter */
    }
    switch (m) {
    case 1:
        if (dt == 15) {
            /* all coefficients are 1 */
            memset(cc_tab, 1, cc_nb);
        } else {
            /* here coefficients are either 0 or 1 */
            tinymt32_init(&s, repair_key);
            for (i = 0 ; i < cc_nb ; i++) {
                cc_tab[i] = (tinymt32_rand16(&s) <= dt) ? 1 : 0;
            }
        }
        break;

    case 8:
        tinymt32_init(&s, repair_key);
        if (dt == 15) {
            /* coefficient 0 is avoided here in order to include
             * all the source symbols */
            for (i = 0 ; i < cc_nb ; i++) {
                do {
                    cc_tab[i] = (uint8_t) tinymt32_rand256(&s);
                } while (cc_tab[i] == 0);
            }
        } else {
            /* here a certain number of coefficients should be 0 */
            for (i = 0 ; i < cc_nb ; i++) {
                if (tinymt32_rand16(&s) <= dt) {
                    do {
                        cc_tab[i] = (uint8_t) tinymt32_rand256(&s);
                    } while (cc_tab[i] == 0);
                } else {
                    cc_tab[i] = 0;
                }
            }
        }
    }
}
```

```

        }
        break;

    default:
        return -2; /* error, bad parameter m */
    }
    return 0; /* success */
}
<CODE ENDS>

```

Figure 3: Coding Coefficients Generation Function Reference Implementation

3.7. Finite Fields Operations

3.7.1. Finite Field Definitions

The two RLC FEC Schemes specified in this document reuse the Finite Fields defined in [RFC5510], section 8.1. More specifically, the elements of the field $GF(2^m)$ are represented by polynomials with binary coefficients (i.e., over $GF(2)$) and degree lower or equal to $m-1$. The addition between two elements is defined as the addition of binary polynomials in $GF(2)$, which is equivalent to a bitwise XOR operation on the binary representation of these elements.

With $GF(2^8)$, multiplication between two elements is the multiplication modulo a given irreducible polynomial of degree 8. The following irreducible polynomial is used for $GF(2^8)$:

$$x^8 + x^4 + x^3 + x^2 + 1$$

With $GF(2)$, multiplication corresponds to a logical AND operation.

3.7.2. Linear Combination of Source Symbols Computation

The two RLC FEC Schemes require the computation of a linear combination of source symbols, using the coding coefficients produced by the `generate_coding_coefficients()` function and stored in the `cc_tab[]` array.

With the RLC over $GF(2^8)$ FEC Scheme, a linear combination of the `ew_size` source symbol present in the encoding window, say `src_0` to `src_ew_size_1`, in order to generate a repair symbol, is computed as follows. For each byte of position `i` in each source and the repair symbol, where `i` belongs to `[0; E-1]`, compute:

$$\text{repair}[i] = \text{cc_tab}[0] * \text{src_0}[i] \text{ XOR } \text{cc_tab}[1] * \text{src_1}[i] \text{ XOR } \dots \text{ XOR } \text{cc_tab}[\text{ew_size} - 1] * \text{src_ew_size_1}[i]$$

where $*$ is the multiplication over $GF(2^{^8})$. In practice various optimizations need to be used in order to make this computation efficient (see in particular [PGM13]).

With the RLC over $GF(2)$ FEC Scheme (binary case), a linear combination is computed as follows. The repair symbol is the XOR sum of all the source symbols corresponding to a coding coefficient $cc_tab[j]$ equal to 1 (i.e., the source symbols corresponding to zero coding coefficients are ignored). The XOR sum of the byte of position i in each source is computed and stored in the corresponding byte of the repair symbol, where i belongs to $[0; E-1]$. In practice, the XOR sums will be computed several bytes at a time (e.g., on 64 bit words, or on arrays of 16 or more bytes when using SIMD CPU extensions).

With both FEC Schemes, the details of how to optimize the computation of these linear combinations are of high practical importance but out of scope of this document.

4. Sliding Window RLC FEC Scheme over $GF(2^{^8})$ for Arbitrary Packet Flows

This fully-specified FEC Scheme defines the Sliding Window Random Linear Codes (RLC) over $GF(2^{^8})$.

4.1. Formats and Codes

4.1.1. FEC Framework Configuration Information

Following the guidelines of [RFC6363], section 5.6, this section provides the FEC Framework Configuration Information (or FFCI). This FFCI needs to be shared (e.g., using SDP) between the FECFRAME sender and receiver instances in order to synchronize them. It includes a FEC Encoding ID, mandatory for any FEC Scheme specification, plus scheme-specific elements.

4.1.1.1. FEC Encoding ID

- o FEC Encoding ID: the value assigned to this fully specified FEC Scheme MUST be XXXX, as assigned by IANA (Section 10).

When SDP is used to communicate the FFCI, this FEC Encoding ID is carried in the 'encoding-id' parameter.

4.1.1.2. FEC Scheme-Specific Information

The FEC Scheme-Specific Information (FSSI) includes elements that are specific to the present FEC Scheme. More precisely:

Encoding symbol size (E): a non-negative integer that indicates the size of each encoding symbol in bytes;

Window Size Ratio (WSR) parameter: a non-negative integer between 0 and 255 (both inclusive) used to initialize window sizes. A value of 0 indicates this parameter is not considered (e.g., a fixed encoding window size may be chosen). A value between 1 and 255 inclusive is required by certain of the parameter derivation techniques described in Appendix C;

This element is required both by the sender (RLC encoder) and the receiver(s) (RLC decoder).

When SDP is used to communicate the FFCI, this FEC Scheme-specific information is carried in the 'fssi' parameter in textual representation as specified in [RFC6364]. For instance:

```
fssi=E:1400,WSR:191
```

In that case the name values "E" and "WSR" are used to convey the E and WSR parameters respectively.

If another mechanism requires the FSSI to be carried as an opaque octet string, the encoding format consists of the following three octets, where the E field is carried in "big-endian" or "network order" format, that is, most significant byte first:

Encoding symbol length (E): 16-bit field;
Window Size Ratio Parameter (WSR): 8-bit field.

These three octets can be communicated as such, or for instance, be subject to an additional Base64 encoding.

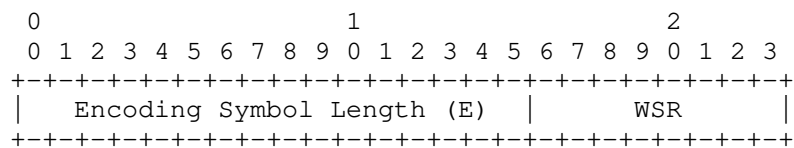


Figure 4: FSSI Encoding Format

4.1.2. Explicit Source FEC Payload ID

A FEC Source Packet MUST contain an Explicit Source FEC Payload ID that is appended to the end of the packet as illustrated in Figure 5.

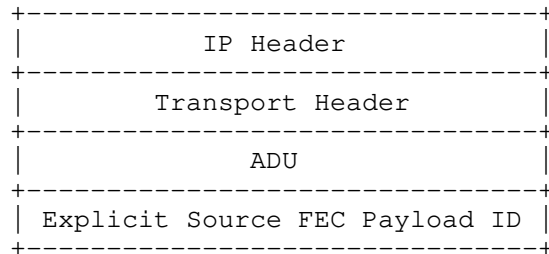


Figure 5: Structure of an FEC Source Packet with the Explicit Source FEC Payload ID

More precisely, the Explicit Source FEC Payload ID is composed of the following field, carried in "big-endian" or "network order" format, that is, most significant byte first (Figure 6):

Encoding Symbol ID (ESI) (32-bit field): this unsigned integer identifies the first source symbol of the ADUI corresponding to this FEC Source Packet. The ESI is incremented for each new source symbol, and after reaching the maximum value ($2^{32}-1$), wrapping to zero occurs.

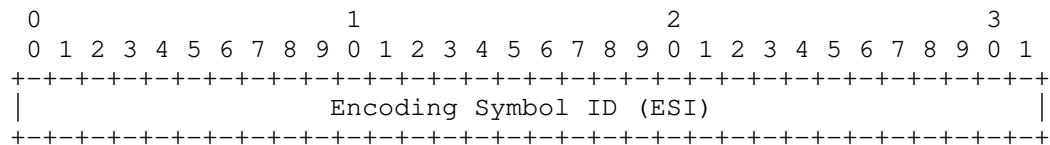


Figure 6: Source FEC Payload ID Encoding Format

4.1.3. Repair FEC Payload ID

A FEC Repair Packet MAY contain one or more repair symbols. When there are several repair symbols, all of them MUST have been generated from the same encoding window, using Repair_Key values that are managed as explained below. A receiver can easily deduce the number of repair symbols within a FEC Repair Packet by comparing the received FEC Repair Packet size (equal to the UDP payload size when UDP is the underlying transport protocol) and the symbol size, E, communicated in the FFCI.

A FEC Repair Packet MUST contain a Repair FEC Payload ID that is prepended to the repair symbol as illustrated in Figure 7.

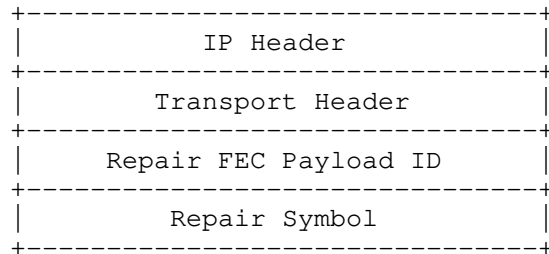


Figure 7: Structure of an FEC Repair Packet with the Repair FEC Payload ID

More precisely, the Repair FEC Payload ID is composed of the following fields where all integer fields are carried in "big-endian" or "network order" format, that is, most significant byte first (Figure 8):

Repair_Key (16-bit field): this unsigned integer is used as a seed by the coefficient generation function (Section 3.6) in order to generate the desired number of coding coefficients. This repair key may be a monotonically increasing integer value that loops back to 0 after reaching 65535 (see Section 6.1). When a FEC Repair Packet contains several repair symbols, this repair key value is that of the first repair symbol. The remaining repair keys can be deduced by incrementing by 1 this value, up to a maximum value of 65535 after which it loops back to 0.

Density Threshold for the coding coefficients, DT (4-bit field): this unsigned integer carries the Density Threshold (DT) used by the coding coefficient generation function Section 3.6. More precisely, it controls the probability of having a non zero coding coefficient, which equals $(DT+1) / 16$. When a FEC Repair Packet contains several repair symbols, the DT value applies to all of them;

Number of Source Symbols in the encoding window, NSS (12-bit field):

this unsigned integer indicates the number of source symbols in the encoding window when this repair symbol was generated. When a FEC Repair Packet contains several repair symbols, this NSS value applies to all of them;

ESI of First Source Symbol in the encoding window, FSS_ESI (32-bit field):

this unsigned integer indicates the ESI of the first source symbol in the encoding window when this repair symbol was generated.

When a FEC Repair Packet contains several repair symbols, this FSS_ESI value applies to all of them;

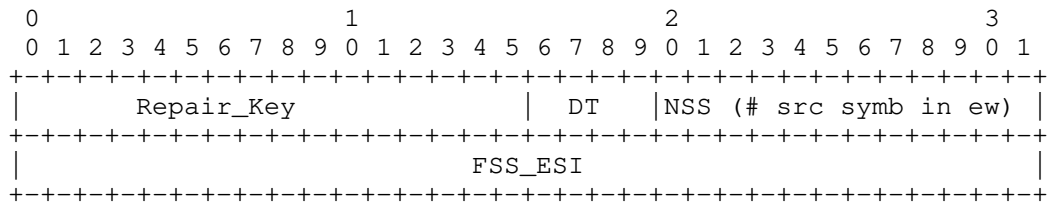


Figure 8: Repair FEC Payload ID Encoding Format

4.2. Procedures

All the procedures of Section 3 apply to this FEC Scheme.

5. Sliding Window RLC FEC Scheme over GF(2) for Arbitrary Packet Flows

This fully-specified FEC Scheme defines the Sliding Window Random Linear Codes (RLC) over GF(2) (binary case).

5.1. Formats and Codes

5.1.1. FEC Framework Configuration Information

5.1.1.1. FEC Encoding ID

- o FEC Encoding ID: the value assigned to this fully specified FEC Scheme MUST be YYYY, as assigned by IANA (Section 10).

When SDP is used to communicate the FFCI, this FEC Encoding ID is carried in the 'encoding-id' parameter.

5.1.1.2. FEC Scheme-Specific Information

All the considerations of Section 4.1.1.2 apply here.

5.1.2. Explicit Source FEC Payload ID

All the considerations of Section 4.1.2 apply here.

5.1.3. Repair FEC Payload ID

All the considerations of Section 4.1.3 apply here, with the only exception that the Repair_Key field is useless if DT = 15 (indeed, in that case all the coefficients are necessarily equal to 1 and the coefficient generation function does not use any PRNG). When DT = 15

the FECFRAME sender MUST set the Repair_Key field to zero on transmission and a receiver MUST ignore it on receipt.

5.2. Procedures

All the procedures of Section 3 apply to this FEC Scheme.

6. FEC Code Specification

6.1. Encoding Side

This section provides a high level description of a Sliding Window RLC encoder.

Whenever a new FEC Repair Packet is needed, the RLC encoder instance first gathers the ew_size source symbols currently in the sliding encoding window. Then it chooses a repair key, which can be a monotonically increasing integer value, incremented for each repair symbol up to a maximum value of 65535 (as it is carried within a 16-bit field) after which it loops back to 0. This repair key is communicated to the coefficient generation function (Section 3.6) in order to generate ew_size coding coefficients. Finally, the FECFRAME sender computes the repair symbol as a linear combination of the ew_size source symbols using the ew_size coding coefficients (Section 3.7). When E is small and when there is an incentive to pack several repair symbols within the same FEC Repair Packet, the appropriate number of repair symbols are computed. In that case the repair key for each of them MUST be incremented by 1, keeping the same ew_size source symbols, since only the first repair key will be carried in the Repair FEC Payload ID. The FEC Repair Packet can then be passed to the transport layer for transmission. The source versus repair FEC packet transmission order is out of scope of this document and several approaches exist that are implementation-specific.

Other solutions are possible to select a repair key value when a new FEC Repair Packet is needed, for instance, by choosing a random integer between 0 and 65535. However, selecting the same repair key as before (which may happen in case of a random process) is only meaningful if the encoding window has changed, otherwise the same FEC Repair Packet will be generated. In any case, choosing the repair key is entirely at the discretion of the sender, since it is communicated to the receiver(s) in each Repair FEC Payload ID. A receiver should not make any assumption on the way the repair key is managed.

6.2. Decoding Side

This section provides a high level description of a Sliding Window RLC decoder.

A FECFRAME receiver needs to maintain a linear system whose variables are the received and lost source symbols. Upon receiving a FEC Repair Packet, a receiver first extracts all the repair symbols it contains (in case several repair symbols are packed together). For each repair symbol, when at least one of the corresponding source symbols it protects has been lost, the receiver adds an equation to the linear system (or no equation if this repair packet does not change the linear system rank). This equation of course re-uses the `ew_size` coding coefficients that are computed by the same coefficient generation function (Section 3.6), using the repair key and encoding window descriptions carried in the Repair FEC Payload ID. Whenever possible (i.e., when a sub-system covering one or more lost source symbols is of full rank), decoding is performed in order to recover lost source symbols. Gaussian elimination is one possible algorithm to solve this linear system. Each time an ADUI can be totally recovered, padding is removed (thanks to the Length field, `L`, of the ADUI) and the ADU is assigned to the corresponding application flow (thanks to the Flow ID field, `F`, of the ADUI). This ADU is finally passed to the corresponding upper application. Received FEC Source Packets, containing an ADU, MAY be passed to the application either immediately or after some time to guaranty an ordered delivery to the application. This document does not mandate any approach as this is an operational and management decision.

With real-time flows, a lost ADU that is decoded after the maximum latency or an ADU received after this delay has no value to the application. This raises the question of deciding whether or not an ADU is late. This decision MAY be taken within the FECFRAME receiver (e.g., using the decoding window, see Section 3.1) or within the application (e.g., using RTP timestamps within the ADU). Deciding which option to follow and whether or not to pass all ADUs, including those assumed late, to the application are operational decisions that depend on the application and are therefore out of scope of this document. Additionally, Appendix D discusses a backward compatible optimization whereby late source symbols MAY still be used within the FECFRAME receiver in order to improve transmission robustness.

7. Implementation Status

Editor's notes: RFC Editor, please remove this section motivated by RFC 6982 before publishing the RFC. Thanks.

An implementation of the Sliding Window RLC FEC Scheme for FECFRAME exists:

- o Organisation: Inria
- o Description: This is an implementation of the Sliding Window RLC FEC Scheme limited to $GF(2^{8})$. It relies on a modified version of our OpenFEC (<http://openfec.org>) FEC code library. It is integrated in our FECFRAME software (see [fecframe-ext]).
- o Maturity: prototype.
- o Coverage: this software complies with the Sliding Window RLC FEC Scheme.
- o Licensing: proprietary.
- o Contact: vincent.roca@inria.fr

8. Security Considerations

The FEC Framework document [RFC6363] provides a fairly comprehensive analysis of security considerations applicable to FEC Schemes. Therefore, the present section follows the security considerations section of [RFC6363] and only discusses specific topics.

8.1. Attacks Against the Data Flow

8.1.1. Access to Confidential Content

The Sliding Window RLC FEC Scheme specified in this document does not change the recommendations of [RFC6363]. To summarize, if confidentiality is a concern, it is RECOMMENDED that one of the solutions mentioned in [RFC6363] is used with special considerations to the way this solution is applied (e.g., is encryption applied before or after FEC protection, within the end-system or in a middlebox), to the operational constraints (e.g., performing FEC decoding in a protected environment may be complicated or even impossible) and to the threat model.

8.1.2. Content Corruption

The Sliding Window RLC FEC Scheme specified in this document does not change the recommendations of [RFC6363]. To summarize, it is RECOMMENDED that one of the solutions mentioned in [RFC6363] is used on both the FEC Source and Repair Packets.

8.2. Attacks Against the FEC Parameters

The FEC Scheme specified in this document defines parameters that can be the basis of attacks. More specifically, the following parameters of the FFCI may be modified by an attacker who targets receivers (Section 4.1.1.2):

- o FEC Encoding ID: changing this parameter leads a receiver to consider a different FEC Scheme. The consequences are severe, the format of the Explicit Source FEC Payload ID and Repair FEC Payload ID of received packets will probably differ, leading to various malfunctions. Even if the original and modified FEC Schemes share the same format, FEC decoding will either fail or lead to corrupted decoded symbols. This will happen if an attacker turns value YYYY (i.e., RLC over $GF(2)$) to value XXXX (RLC over $GF(2^{8})$), an additional consequence being a higher processing overhead at the receiver. In any case, the attack results in a form of Denial of Service (DoS) or corrupted content.
- o Encoding symbol length (E): setting this E parameter to a different value will confuse a receiver. If the size of a received FEC Repair Packet is no longer multiple of the modified E value, a receiver quickly detects a problem and SHOULD reject the packet. If the new E value is a sub-multiple of the original E value (e.g., half the original value), then receivers may not detect the problem immediately. For instance, a receiver may think that a received FEC Repair Packet contains more repair symbols (e.g., twice as many if E is reduced by half), leading to malfunctions whose nature depends on implementation details. Here also, the attack always results in a form of DoS or corrupted content.

It is therefore RECOMMENDED that security measures be taken to guarantee the FFCI integrity, as specified in [RFC6363]. How to achieve this depends on the way the FFCI is communicated from the sender to the receiver, which is not specified in this document.

Similarly, attacks are possible against the Explicit Source FEC Payload ID and Repair FEC Payload ID. More specifically, in case of a FEC Source Packet, the following value can be modified by an attacker who targets receivers:

- o Encoding Symbol ID (ESI): changing the ESI leads a receiver to consider a wrong ADU, resulting in severe consequences, including corrupted content passed to the receiving application;

And in case of a FEC Repair Packet:

- o Repair Key: changing this value leads a receiver to generate a wrong coding coefficient sequence, and therefore any source symbol decoded using the repair symbols contained in this packet will be corrupted;
- o DT: changing this value also leads a receiver to generate a wrong coding coefficient sequence, and therefore any source symbol decoded using the repair symbols contained in this packet will be corrupted. In addition, if the DT value is significantly

- increased, it will generate a higher processing overhead at a receiver. In case of very large encoding windows, this may impact the terminal performance;
- o NSS: changing this value leads a receiver to consider a different set of source symbols, and therefore any source symbol decoded using the repair symbols contained in this packet will be corrupted. In addition, if the NSS value is significantly increased, it will generate a higher processing overhead at a receiver, which may impact the terminal performance;
 - o FSS_ESI: changing this value also leads a receiver to consider a different set of source symbols and therefore any source symbol decoded using the repair symbols contained in this packet will be corrupted.

It is therefore RECOMMENDED that security measures are taken to guarantee the FEC Source and Repair Packets as stated in [RFC6363].

8.3. When Several Source Flows are to be Protected Together

The Sliding Window RLC FEC Scheme specified in this document does not change the recommendations of [RFC6363].

8.4. Baseline Secure FEC Framework Operation

The Sliding Window RLC FEC Scheme specified in this document does not change the recommendations of [RFC6363] concerning the use of the IPsec/ESP security protocol as a mandatory to implement (but not mandatory to use) security scheme. This is well suited to situations where the only insecure domain is the one over which the FEC Framework operates.

8.5. Additional Security Considerations for Numerical Computations

In addition to the above security considerations, inherited from [RFC6363], the present document introduces several formulae, in particular in Appendix C.1. It is RECOMMENDED to check that the computed values stay within reasonable bounds since numerical overflows, caused by an erroneous implementation or an erroneous input value, may lead to hazardous behaviours. However, what "reasonable bounds" means is use-case and implementation dependent and is not detailed in this document.

Appendix C.2 also mentions the possibility of "using the timestamp field of an RTP packet header" when applicable. A malicious attacker may deliberately corrupt this header field in order to trigger hazardous behaviours at a FECFRAME receiver. Protection against this type of content corruption can be addressed with the above recommendations on a baseline secure operation. In addition, it is

also RECOMMENDED to check that the timestamp value be within reasonable bounds.

9. Operations and Management Considerations

The FEC Framework document [RFC6363] provides a fairly comprehensive analysis of operations and management considerations applicable to FEC Schemes. Therefore, the present section only discusses specific topics.

9.1. Operational Recommendations: Finite Field GF(2) Versus GF(2⁸)

The present document specifies two FEC Schemes that differ on the Finite Field used for the coding coefficients. It is expected that the RLC over GF(2⁸) FEC Scheme will be mostly used since it warrants a higher packet loss protection. In case of small encoding windows, the associated processing overhead is not an issue (e.g., we measured decoding speeds between 745 Mbps and 2.8 Gbps on an ARM Cortex-A15 embedded board in [Roca17] depending on the code rate and the channel conditions, using an encoding window of size 18 or 23 symbols; see the above article for the details). Of course the CPU overhead will increase with the encoding window size, because more operations in the GF(2⁸) finite field will be needed.

The RLC over GF(2) FEC Scheme offers an alternative. In that case operations symbols can be directly XOR-ed together which warrants high bitrate encoding and decoding operations, and can be an advantage with large encoding windows. However, packet loss protection is significantly reduced by using this FEC Scheme.

9.2. Operational Recommendations: Coding Coefficients Density Threshold

In addition to the choice of the Finite Field, the two FEC Schemes define a coding coefficient density threshold (DT) parameter. This parameter enables a sender to control the code density, i.e., the proportion of coefficients that are non zero on average. With RLC over GF(2⁸), it is usually appropriate that small encoding windows be associated to a density threshold equal to 15, the maximum value, in order to warrant a high loss protection.

On the opposite, with larger encoding windows, it is usually appropriate that the density threshold be reduced. With large encoding windows, an alternative can be to use RLC over GF(2) and a density threshold equal to 7 (i.e., an average density equal to 1/2) or smaller.

Note that using a density threshold equal to 15 with RLC over GF(2) is equivalent to using an XOR code that computes the XOR sum of all

the source symbols in the encoding window. In that case: (1) only a single repair symbol can be produced for any encoding window, and (2) the `repair_key` parameter becomes useless (the coding coefficients generation function does not rely on the PRNG).

10. IANA Considerations

This document registers two values in the "FEC Framework (FECFRAME) FEC Encoding IDs" registry [RFC6363] as follows:

- o YYYY refers to the Sliding Window Random Linear Codes (RLC) over GF(2) FEC Scheme for Arbitrary Packet Flows, as defined in Section 5 of this document.
- o XXXX refers to the Sliding Window Random Linear Codes (RLC) over GF(2⁸) FEC Scheme for Arbitrary Packet Flows, as defined in Section 4 of this document.

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Appendix A. TinyMT32 Validation Criteria (Normative)

PRNG determinism, for a given seed, is a requirement. Consequently, in order to validate an implementation of the TinyMT32 PRNG, the following criteria MUST be met.

The first criterion focusses on the `tinymt32_rand256()`, where the 32-bit integer of the core TinyMT32 PRNG is scaled down to an 8-bit integer. Using a seed value of 1, the first 50 values returned by: `tinymt32_rand256()` as 8-bit unsigned integers MUST be equal to values provided in Figure 9, to be read line by line.

37	225	177	176	21
246	54	139	168	237
211	187	62	190	104
135	210	99	176	11
207	35	40	113	179
214	254	101	212	211
226	41	234	232	203
29	194	211	112	107
217	104	197	135	23
89	210	252	109	166

Figure 9: First 50 decimal values (to be read per line) returned by `tinymt32_rand256()` as 8-bit unsigned integers, with a seed value of 1.

The second criterion focusses on the `tinymt32_rand16()`, where the 32-bit integer of the core TinyMT32 PRNG is scaled down to a 4-bit integer. Using a seed value of 1, the first 50 values returned by: `tinymt32_rand16()` as 4-bit unsigned integers MUST be equal to values provided in Figure 10, to be read line by line.

5	1	1	0	5
6	6	11	8	13
3	11	14	14	8
7	2	3	0	11
15	3	8	1	3
6	14	5	4	3
2	9	10	8	11
13	2	3	0	11
9	8	5	7	7
9	2	12	13	6

Figure 10: First 50 decimal values (to be read per line) returned by `tinymt32_rand16()` as 4-bit unsigned integers, with a seed value of 1.

Appendix B. Assessing the PRNG Adequacy (Informational)

This annex discusses the adequacy of the TinyMT32 PRNG and the `tinymt32_rand16()` and `tinymt32_rand256()` functions, to the RLC FEC Schemes. The goal is to assess the adequacy of these two functions in producing coding coefficients that are sufficiently different from one another, across various repair symbols with repair key values in sequence (we can expect this approach to be commonly used by implementers, see Section 6.1). This section is purely informational and does not claim to be a solid evaluation.

The two RLC FEC Schemes use the PRNG to produce pseudo-random coding coefficients (Section 3.6), each time a new repair symbol is needed. A different repair key is used for each repair symbol, usually by incrementing the repair key value (Section 6.1). For each repair symbol, a limited number of pseudo-random numbers is needed, depending on the DT and encoding window size (Section 3.6), using either `tinymt32_rand16()` or `tinymt32_rand256()`. Therefore we are more interested in the randomness of small sequences of random numbers mapped to 4-bit or 8-bit integers, than in the randomness of a very large sequence of random numbers which is not representative of the usage of the PRNG.

Evaluation of `tinymt32_rand16()`: We first generate a huge number (1,000,000,000) of small sequences (20 pseudo-random numbers per sequence), increasing the seed value for each sequence, and perform statistics on the number of occurrences of each of the 16 possible values across all sequences. In this first test we consider 32-bit seed values in order to assess the PRNG quality after output truncation to 4 bits.

value	occurrences	percentage (%) (total of 20000000000)
0	1250036799	6.2502
1	1249995831	6.2500
2	1250038674	6.2502
3	1250000881	6.2500
4	1250023929	6.2501
5	1249986320	6.2499
6	1249995587	6.2500
7	1250020363	6.2501
8	1249995276	6.2500
9	1249982856	6.2499
10	1249984111	6.2499
11	1250009551	6.2500
12	1249955768	6.2498
13	1249994654	6.2500
14	1250000569	6.2500
15	1249978831	6.2499

Figure 11: `tinymt32_rand16()`: occurrence statistics across a huge number (1,000,000,000) of small sequences (20 pseudo-random numbers per sequence), with 0 as the first PRNG seed.

The results (Figure 11) show that all possible values are almost equally represented, or said differently, that the `tinymt32_rand16()` output converges to a uniform distribution where each of the 16 possible values would appear exactly $1 / 16 * 100 = 6.25\%$ of times.

Since the RLC FEC Schemes use of this PRNG will be limited to 16-bit seed values, we carried out the same test for the first 2^{16} seed values only. The distribution (not shown) is of course less uniform, with value occurrences ranging between 6.2121% (i.e., 81,423 occurrences out of a total of $65536 * 20 = 1,310,720$) and 6.2948% (i.e., 82,507 occurrences). However, we do not believe it significantly impacts the RLC FEC Scheme behavior.

Other types of biases may exist that may be visible with smaller tests, for instance to evaluate the convergence speed to a uniform distribution. We therefore perform 200 tests, each of them consisting in producing 200 sequences, keeping only the first value of each sequence. We use non overlapping repair keys for each sequence, starting with value 0 and increasing it after each use.

value	min occurrences	max occurrences	average occurrences
0	4	21	6.3675
1	4	22	6.0200
2	4	20	6.3125
3	5	23	6.1775
4	5	24	6.1000
5	4	21	6.5925
6	5	30	6.3075
7	6	22	6.2225
8	5	26	6.1750
9	3	21	5.9425
10	5	24	6.3175
11	4	22	6.4300
12	5	21	6.1600
13	5	22	6.3100
14	4	26	6.3950
15	4	21	6.1700

Figure 12: `tinymt32_rand16()`: occurrence statistics across 200 tests, each of them consisting in 200 sequences of 1 pseudo-random number each, with non overlapping PRNG seeds in sequence starting from 0.

Figure 12 shows across all 200 tests, for each of the 16 possible pseudo-random number values, the minimum (resp. maximum) number of times it appeared in a test, as well as the average number of occurrences across the 200 tests. Although the distribution is not perfect, there is no major bias. On the opposite, in the same conditions, the Park-Miller linear congruential PRNG of [RFC5170] with a result scaled down to 4-bit values, using seeds in sequence starting from 1, returns systematically 0 as the first value during some time, then after a certain repair key value threshold, it systematically returns 1, etc.

Evaluation of `tinymt32_rand256()`: The same approach is used here. Results (not shown) are similar: occurrences vary between 7,810,3368 (i.e., 0.3905%) and 7,814,7952 (i.e., 0.3907%). Here also we see a convergence to the theoretical uniform distribution where each of the 256 possible values would appear exactly $1 / 256 * 100 = 0.390625\%$ of times.

Appendix C. Possible Parameter Derivation (Informational)

Section 3.1 defines several parameters to control the encoder or decoder. This annex proposes techniques to derive these parameters according to the target use-case. This annex is informational, in the sense that using a different derivation technique will not prevent the encoder and decoder to interoperate: a decoder can still recover an erased source symbol without any error. However, in case

of a real-time flow, an inappropriate parameter derivation may lead to the decoding of erased source packets after their validity period, making them useless to the target application. This annex proposes an approach to reduce this risk, among other things.

The FEC Schemes defined in this document can be used in various manners, depending on the target use-case:

- o the source ADU flow they protect may or may not have real-time constraints;
- o the source ADU flow may be a Constant Bitrate (CBR) or Variable BitRate (VBR) flow;
- o with a VBR source ADU flow, the flow's minimum and maximum bitrates may or may not be known;
- o and the communication path between encoder and decoder may be a CBR communication path (e.g., as with certain LTE-based broadcast channels) or not (general case, e.g., with Internet).

The parameter derivation technique should be suited to the use-case, as described in the following sections.

C.1. Case of a CBR Real-Time Flow

In the following, we consider a real-time flow with `max_lat` latency budget. The encoding symbol size, `E`, is constant. The code rate, `cr`, is also constant, its value depending on the expected communication loss model (this choice is out of scope of this document).

In a first configuration, the source ADU flow bitrate at the input of the FECFRAME sender is fixed and equal to `br_in` (in bits/s), and this value is known by the FECFRAME sender. It follows that the transmission bitrate at the output of the FECFRAME sender will be higher, depending on the added repair flow overhead. In order to comply with the maximum FEC-related latency budget, we have:

$$\text{dw_max_size} = (\text{max_lat} * \text{br_in}) / (8 * E)$$

assuming that the encoding and decoding times are negligible with respect to the target `max_lat`. This is a reasonable assumption in many situations (e.g., see Section 9.1 in case of small window sizes). Otherwise the `max_lat` parameter should be adjusted in order to avoid the problem. In any case, interoperability will never be compromised by choosing a too large value.

In a second configuration, the FECFRAME sender generates a fixed bitrate flow, equal to the CBR communication path bitrate equal to `br_out` (in bits/s), and this value is known by the FECFRAME sender,

as in [Roca17]. The maximum source flow bitrate needs to be such that, with the added repair flow overhead, the total transmission bitrate remains inferior or equal to `br_out`. We have:

$$\text{dw_max_size} = (\text{max_lat} * \text{br_out} * \text{cr}) / (8 * E)$$

assuming here also that the encoding and decoding times are negligible with respect to the target `max_lat`.

For decoding to be possible within the latency budget, it is required that the encoding window maximum size be smaller than or at most equal to the decoding window maximum size. The `ew_max_size` is the main parameter at a FECFRAME sender, but its exact value has no impact on the the FEC-related latency budget. The `ew_max_size` parameter is computed as follows:

$$\text{ew_max_size} = \text{dw_max_size} * \text{WSR} / 255$$

In line with [Roca17], `WSR = 191` is considered as a reasonable value (the resulting encoding to decoding window size ratio is then close to 0.75), but other values between 1 and 255 inclusive are possible, depending on the use-case.

The `dw_max_size` is computed by a FECFRAME sender but not explicitly communicated to a FECFRAME receiver. However, a FECFRAME receiver can easily evaluate the `ew_max_size` by observing the maximum Number of Source Symbols (NSS) value contained in the Repair FEC Payload ID of received FEC Repair Packets (Section 4.1.3). A receiver can then easily compute `dw_max_size`:

$$\text{dw_max_size} = \text{max_NSS_observed} * 255 / \text{WSR}$$

A receiver can then chose an appropriate linear system maximum size:

$$\text{ls_max_size} \geq \text{dw_max_size}$$

It is good practice to use a larger value for `ls_max_size` as explained in Appendix D, which does not impact maximum latency nor interoperability.

In any case, for a given use-case (i.e., for target encoding and decoding devices and desired protection levels in front of communication impairments) and for the computed `ew_max_size`, `dw_max_size` and `ls_max_size` values, it is RECOMMENDED to check that the maximum encoding time and maximum memory requirements at a FECFRAME sender, and maximum decoding time and maximum memory requirements at a FECFRAME receiver, stay within reasonable bounds. When assuming that the encoding and decoding times are negligible

with respect to the target `max_lat`, this should be verified as well, otherwise the `max_lat` SHOULD be adjusted accordingly.

The particular case of session start needs to be managed appropriately since the `ew_size`, starting at zero, increases each time a new source ADU is received by the FECFRAME sender, until it reaches the `ew_max_size` value. Therefore a FECFRAME receiver SHOULD continuously observe the received FEC Repair Packets, since the NSS value carried in the Repair FEC Payload ID will increase too, and adjust its `ls_max_size` accordingly if need be. With a CBR flow, session start is expected to be the only moment when the encoding window size will increase. Similarly, with a CBR real-time flow, the session end is expected to be the only moment when the encoding window size will progressively decrease. No adjustment of the `ls_max_size` is required at the FECFRAME receiver in that case.

C.2. Other Types of Real-Time Flow

In the following, we consider a real-time source ADU flow with a `max_lat` latency budget and a variable bitrate (VBR) measured at the entry of the FECFRAME sender. A first approach consists in considering the smallest instantaneous bitrate of the source ADU flow, when this parameter is known, and to reuse the derivation of Appendix C.1. Considering the smallest bitrate means that the encoding and decoding window maximum size estimations are pessimistic: these windows have the smallest size required to enable on-time decoding at a FECFRAME receiver. If the instantaneous bitrate is higher than this smallest bitrate, this approach leads to an encoding window that is unnecessarily small, which reduces robustness in front of long erasure bursts.

Another approach consists in using ADU timing information (e.g., using the timestamp field of an RTP packet header, or registering the time upon receiving a new ADU). From the global FEC-related latency budget, the FECFRAME sender can derive a practical maximum latency budget for encoding operations, `max_lat_for_encoding`. For the FEC Schemes specified in this document, this latency budget SHOULD be computed with:

$$\text{max_lat_for_encoding} = \text{max_lat} * \text{WSR} / 255$$

It follows that any source symbols associated to an ADU that has timed-out with respect to `max_lat_for_encoding` SHOULD be removed from the encoding window. With this approach there is no pre-determined `ew_size` value: this value fluctuates over the time according to the instantaneous source ADU flow bitrate. For practical reasons, a FECFRAME sender may still require that `ew_size` does not increase beyond a maximum value (Appendix C.3).

With both approaches, and no matter the choice of the FECFRAME sender, a FECFRAME receiver can still easily evaluate the `ew_max_size` by observing the maximum Number of Source Symbols (NSS) value contained in the Repair FEC Payload ID of received FEC Repair Packets. A receiver can then compute `dw_max_size` and derive an appropriate `ls_max_size` as explained in Appendix C.1.

When the observed NSS fluctuates significantly, a FECFRAME receiver may want to adapt its `ls_max_size` accordingly. In particular when the NSS is significantly reduced, a FECFRAME receiver may want to reduce the `ls_max_size` too in order to limit computation complexity. A balance must be found between using an `ls_max_size` "too large" (which increases computation complexity and memory requirements) and the opposite (which reduces recovery performance).

C.3. Case of a Non Real-Time Flow

Finally there are configurations where a source ADU flow has no real-time constraints. FECFRAME and the FEC Schemes defined in this document can still be used. The choice of appropriate parameter values can be directed by practical considerations. For instance, it can derive from an estimation of the maximum memory amount that could be dedicated to the linear system at a FECFRAME receiver, or the maximum computation complexity at a FECFRAME receiver, both of them depending on the `ls_max_size` parameter. The same considerations also apply to the FECFRAME sender, where the maximum memory amount and computation complexity depend on the `ew_max_size` parameter.

Here also, the NSS value contained in FEC Repair Packets is used by a FECFRAME receiver to determine the current coding window size and `ew_max_size` by observing its maximum value over the time.

Appendix D. Decoding Beyond Maximum Latency Optimization (Informational)

This annex introduces non normative considerations. It is provided as suggestions, without any impact on interoperability. For more information see [Roca16].

With a real-time source ADU flow, it is possible to improve the decoding performance of sliding window codes without impacting maximum latency, at the cost of extra memory and CPU overhead. The optimization consists, for a FECFRAME receiver, to extend the linear system beyond the decoding window maximum size, by keeping a certain number of old source symbols whereas their associated ADUs timed-out:

$$ls_max_size > dw_max_size$$

Usually the following choice is a good trade-off between decoding performance and extra CPU overhead:

```
ls_max_size = 2 * dw_max_size
```

When the `dw_max_size` is very small, it may be preferable to keep a minimum `ls_max_size` value (e.g., `LS_MIN_SIZE_DEFAULT = 40` symbols). Going below this threshold will not save a significant amount of memory nor CPU cycles. Therefore:

```
ls_max_size = max(2 * dw_max_size, LS_MIN_SIZE_DEFAULT)
```

Finally, it is worth noting that a receiver that benefits from an FEC protection significantly higher than what is required to recover from packet losses, can choose to reduce the `ls_max_size`. In that case lost ADUs will be recovered without relying on this optimization.

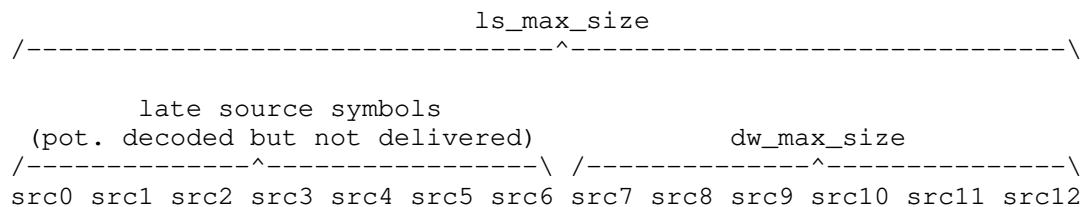


Figure 13: Relationship between parameters to decode beyond maximum latency.

It means that source symbols, and therefore ADUs, may be decoded even if the added latency exceeds the maximum value permitted by the application (the "late source symbols" of Figure 13). It follows that the corresponding ADUs will not be useful to the application. However, decoding these "late symbols" significantly improves the global robustness in bad reception conditions and is therefore recommended for receivers experiencing bad communication conditions [Roca16]. In any case whether or not to use this optimization and what exact value to use for the `ls_max_size` parameter are local decisions made by each receiver independently, without any impact on the other receivers nor on the source.

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TinyMT32 Pseudo Random Number Generator (PRNG)
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Abstract

This document describes the TinyMT32 Pseudo Random Number Generator (PRNG) that produces 32-bit pseudo-random unsigned integers and aims at having a simple-to-use and deterministic solution. This PRNG is a small-sized variant of Mersenne Twister (MT) PRNG. The main advantage of TinyMT32 over MT is the use of a small internal state, compatible with most target platforms that include embedded devices, while keeping a reasonably good randomness that represents a significant improvement compared to the Park-Miller Linear Congruential PRNG. However, neither the TinyMT nor MT PRNG are meant to be used for cryptographic applications.

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1. Introduction

This document specifies the TinyMT32 PRNG, as a specialization of the reference implementation version 1.1 (2015/04/24) by Mutsuo Saito and Makoto Matsumoto, from Hiroshima University, that can be found at [TinyMT-web] (TinyMT web site) and [TinyMT-dev] (Github site). This specialisation aims at having a simple-to-use and deterministic PRNG, as explained below. However, the TinyMT32 PRNG is not meant to be used for cryptographic applications.

TinyMT is a new small-sized variant introduced in 2011 of the Mersenne Twister (MT) PRNG [MT98]. This document focusses on the TinyMT32 variant (rather than TinyMT64) of the TinyMT PRNG, which outputs 32-bit unsigned integers.

The purpose of TinyMT is not to replace Mersenne Twister: TinyMT has a far shorter period ($2^{127} - 1$) than MT. The merit of TinyMT is in the small size of the internal state of 127 bits, far smaller than the 19937 bits of MT. The outputs of TinyMT satisfy several statistical tests for non-cryptographic randomness, including BigCrush in TestU01 [TestU01] and AdaptiveCrush [AdaptiveCrush],

leaving it well-placed for non-cryptographic usage, especially given the small size of its internal state (see [TinyMT-web]). From this point of view, TinyMT32 represents a major improvement with respect to the Park-Miller Linear Congruential PRNG (e.g., as specified in [RFC5170]) that suffers several known limitations (see for instance [PTVF92], section 7.1, p. 279, and [RLC-ID], Appendix B).

The TinyMT32 PRNG initialization depends, among other things, on a parameter set, namely (mat1, mat2, tmat). In order to facilitate the use of this PRNG and make the sequence of pseudo-random numbers depend only on the seed value, this specification requires the use of a specific parameter set (see Section 3.1). This is a major difference with respect to the implementation version 1.1 (2015/04/24) that leaves this parameter set unspecified.

Finally, the determinism of this PRNG, for a given seed, has been carefully checked (see Section 3.3). It means that the same sequence of pseudo-random numbers should be generated, no matter the target execution platform and compiler, for a given initial seed value. This determinism can be a key requirement as it the case with [RLC-ID] that normatively depends on this specification.

2. Definitions

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

3. TinyMT32 PRNG Specification

3.1. TinyMT32 Source Code

The TinyMT32 PRNG requires to be initialized with a parameter set that needs to be well chosen. In this specification, for the sake of simplicity, the following parameter set MUST be used:

```
o mat1 = 0x8f7011ee = 2406486510
o mat2 = 0xfc78ff1f = 4235788063
o tmat = 0x3793fdff = 932445695
```

This parameter set is the first entry of the precalculated parameter sets in file tinymt32dc/tinymt32dc.0.1048576.txt, by Kenji Rikitake, and available at [TinyMT-params]. This is also the parameter set used in [KR12].

The TinyMT32 PRNG reference implementation is reproduced in Figure 1. This is a C language implementation, written for C99 [C99]. This reference implementation differs from the original source code as follows:

- o the original copyright and license have been removed by the original authors who are now authors of this document, in accordance with BCP 78 and the IETF Trust's Legal Provisions Relating to IETF Documents (<http://trustee.ietf.org/license-info>);
- o the source code initially spread over the `tinymt32.h` and `tinymt32.c` files has been merged;
- o the unused parts of the original source code have been removed. This is the case of the `tinymt32_init_by_array()` alternative initialisation function. This is also the case of the `period_certification()` function after having checked it is not required with the chosen parameter set;
- o the unused constants `TINYMT32_MEXP` and `TINYMT32_MUL` have been removed;
- o the appropriate parameter set has been added to the initialization function;
- o the function order has been changed;
- o certain internal variables have been renamed for compactness purposes;
- o the `const` qualifier has been added to the constant definitions;
- o the code that was dependant on the representation of negative integers by 2's complements has been replaced by a more portable version;

<CODE BEGINS>

```
/**
 * Tiny Mersenne Twister only 127 bit internal state.
 * Derived from the reference implementation version 1.1 (2015/04/24)
 * by Mutsuo Saito (Hiroshima University) and Makoto Matsumoto
 * (Hiroshima University).
 */
#include <stdint.h>

/**
 * tinymt32 internal state vector and parameters
 */
typedef struct {
    uint32_t status[4];
    uint32_t mat1;
    uint32_t mat2;
    uint32_t tmat;
} tinymt32_t;

static void tinymt32_next_state (tinymt32_t* s);
```



```
static uint32_t tinymt32_temper (tinymt32_t* s);

/**
 * Parameter set to use for this IETF specification. Don't change.
 * This parameter set is the first entry of the precalculated
 * parameter sets in file tinymt32dc/tinymt32dc.0.1048576.txt, by
 * Kenji Rikitake, available at:
 *   https://github.com/jjlbdx/tinymtdc-longbatch/
 * It is also the parameter set used:
 *   Rikitake, K., "TinyMT Pseudo Random Number Generator for
 *   Erlang", ACM 11th SIGPLAN Erlang Workshop (Erlang'12),
 *   September, 2012.
 */
const uint32_t TINYMT32_MAT1_PARAM = UINT32_C(0x8f7011ee);
const uint32_t TINYMT32_MAT2_PARAM = UINT32_C(0xfc78ff1f);
const uint32_t TINYMT32_TMAT_PARAM = UINT32_C(0x3793fdff);

/**
 * This function initializes the internal state array with a
 * 32-bit unsigned integer seed.
 * @param s pointer to tinymt internal state.
 * @param seed a 32-bit unsigned integer used as a seed.
 */
void tinymt32_init (tinymt32_t* s, uint32_t seed)
{
    const uint32_t MIN_LOOP = 8;
    const uint32_t PRE_LOOP = 8;
    s->status[0] = seed;
    s->status[1] = s->mat1 = TINYMT32_MAT1_PARAM;
    s->status[2] = s->mat2 = TINYMT32_MAT2_PARAM;
    s->status[3] = s->tmat = TINYMT32_TMAT_PARAM;
    for (int i = 1; i < MIN_LOOP; i++) {
        s->status[i & 3] ^= i + UINT32_C(1812433253)
            * (s->status[(i - 1) & 3]
                ^ (s->status[(i - 1) & 3] >> 30));
    }
    /**
     * NB: the parameter set of this specification warrants
     * that none of the possible 232 seeds leads to an
     * all-zero 127-bit internal state. Therefore, the
     * period_certification() function of the original
     * TinyMT32 source code has been safely removed. If
     * another parameter set is used, this function will
     * have to be re-introduced here.
     */
    for (int i = 0; i < PRE_LOOP; i++) {
        tinymt32_next_state(s);
    }
}
```

```

}

/**
 * This function outputs a 32-bit unsigned integer from
 * the internal state.
 * @param s      pointer to tinynt internal state.
 * @return       32-bit unsigned integer r ( $0 \leq r < 2^{32}$ ).
 */
uint32_t tinynt32_generate_uint32 (tinynt32_t* s)
{
    tinynt32_next_state(s);
    return tinynt32_temper(s);
}

/**
 * Internal tinynt32 constants and functions.
 * Users should not call these functions directly.
 */
const uint32_t TINYMT32_SH0 = 1;
const uint32_t TINYMT32_SH1 = 10;
const uint32_t TINYMT32_SH8 = 8;
const uint32_t TINYMT32_MASK = UINT32_C(0x7fffffff);

/**
 * This function changes the internal state of tinynt32.
 * @param s      pointer to tinynt internal state.
 */
static void tinynt32_next_state (tinynt32_t* s)
{
    uint32_t x;
    uint32_t y;

    y = s->status[3];
    x = (s->status[0] & TINYMT32_MASK)
        ^ s->status[1]
        ^ s->status[2];
    x ^= (x << TINYMT32_SH0);
    y ^= (y >> TINYMT32_SH0) ^ x;
    s->status[0] = s->status[1];
    s->status[1] = s->status[2];
    s->status[2] = x ^ (y << TINYMT32_SH1);
    s->status[3] = y;
    /*
     * The if (y & 1) {...} block below replaces:
     *     s->status[1] ^= -((int32_t)(y & 1)) & s->mat1;
     *     s->status[2] ^= -((int32_t)(y & 1)) & s->mat2;
     * The adopted code is equivalent to the original code
     * but does not depend on the representation of negative

```

```

    * integers by 2's complements. It is therefore more
    * portable, but includes an if-branch which may slow
    * down the generation speed.
    */
    if (y & 1) {
        s->status[1] ^= s->mat1;
        s->status[2] ^= s->mat2;
    }
}

/**
 * This function outputs a 32-bit unsigned integer from
 * the internal state.
 * @param s      pointer to tinymt internal state.
 * @return       32-bit unsigned pseudo-random number.
 */
static uint32_t tinymt32_temper (tinymt32_t* s)
{
    uint32_t t0, t1;
    t0 = s->status[3];
    t1 = s->status[0] + (s->status[2] >> TINYMT32_SH8);
    t0 ^= t1;
    /*
     * The if (t1 & 1) {...} block below replaces:
     *     t0 ^= -((int32_t)(t1 & 1)) & s->tmat;
     * The adopted code is equivalent to the original code
     * but does not depend on the representation of negative
     * integers by 2's complements. It is therefore more
     * portable, but includes an if-branch which may slow
     * down the generation speed.
     */
    if (t1 & 1) {
        t0 ^= s->tmat;
    }
    return t0;
}
<CODE ENDS>

```

Figure 1: TinyMT32 Reference Implementation

3.2. TinyMT32 Usage

This PRNG MUST first be initialized with the following function:

```
void tinymt32_init (tinymt32_t* s, uint32_t seed);
```

It takes as input a 32-bit unsigned integer used as a seed (note that value 0 is permitted by TinyMT32). This function also takes as input

a pointer to an instance of a `tinymt32_t` structure that needs to be allocated by the caller but left uninitialized. This structure will then be updated by the various TinyMT32 functions in order to keep the internal state of the PRNG. The use of this structure admits several instances of this PRNG to be used in parallel, each of them having its own instance of the structure.

Then, each time a new 32-bit pseudo-random unsigned integer between 0 and $2^{32} - 1$ inclusive is needed, the following function is used:

```
uint32_t tinymt32_generate_uint32 (tinymt32_t * s);
```

Of course, the `tinymt32_t` structure must be left unchanged by the caller between successive calls to this function.

3.3. Specific Implementation Validation and Deterministic Behavior

PRNG determinism, for a given seed, can be a requirement (e.g., with [RLC-ID]). Consequently, any implementation of the TinyMT32 PRNG in line with this specification **MUST** have the same output as that provided by the reference implementation of Figure 1. In order to increase the compliancy confidence, this document proposes the following criteria. Using a seed value of 1, the first 50 values returned by `tinymt32_generate_uint32(s)` as 32-bit unsigned integers are equal to values provided in Figure 2, to be read line by line. Note that these values come from the `tinymt/check32.out.txt` file provided by the PRNG authors to validate implementations of TinyMT32, as part of the MersenneTwister-Lab/TinyMT Github repository.

```
2545341989  981918433 3715302833 2387538352 3591001365
3820442102 2114400566 2196103051 2783359912 764534509
 643179475 1822416315 881558334 4207026366 3690273640
3240535687 2921447122 3984931427 4092394160 44209675
2188315343 2908663843 1834519336 3774670961 3019990707
4065554902 1239765502 4035716197 3412127188 552822483
 161364450 353727785 140085994 149132008 2547770827
4064042525 4078297538 2057335507 622384752 2041665899
2193913817 1080849512 33160901 662956935 642999063
3384709977 1723175122 3866752252 521822317 2292524454
```

Figure 2: First 50 decimal values (to be read per line) returned by `tinymt32_generate_uint32(s)` as 32-bit unsigned integers, with a seed value of 1.

In particular, the deterministic behavior of the Figure 1 source code has been checked across several platforms: high-end laptops running 64-bits Mac OSX and Linux/Ubuntu; a board featuring a 32-bits ARM Cortex-A15 and running 32-bit Linux/Ubuntu; several embedded cards

featuring either an ARM Cortex-M0+, a Cortex-M3 or a Cortex-M4 32-bit microcontroller, all of them running RIOT [Baccelli18]; two low-end embedded cards featuring either a 16-bit microcontroller (TI MSP430) or a 8-bit microcontroller (Arduino ATMEGA2560), both of them running RIOT.

This specification only outputs 32-bit unsigned pseudo-random numbers and does not try to map this output to a smaller integer range (e.g., between 10 and 49 inclusive). If a specific use-case needs such a mapping, it will have to provide its own function. In that case, if PRNG determinism is also required, the use of floating point (single or double precision) to perform this mapping should probably be avoided, these calculations leading potentially to different rounding errors across different target platforms. Great care should also be put on not introducing biases in the randomness of the mapped output (it may be the case with some mapping algorithms) incompatible with the use-case requirements. The details of how to perform such a mapping are out-of-scope of this document.

4. Security Considerations

The authors do not believe the present specification generates specific security risks per se. However, neither the TinyMT nor MT PRNG are meant to be used for cryptographic applications.

5. IANA Considerations

This document does not require any IANA action.

6. Acknowledgments

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Considerations around Transport Header Confidentiality, Network
Operations, and the Evolution of Internet Transport Protocols
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Abstract

To protect user data and privacy, Internet transport protocols have supported payload encryption and authentication for some time. Such encryption and authentication is now also starting to be applied to the transport protocol headers. This helps avoid transport protocol ossification by middleboxes, mitigate attacks against the transport protocol, and protect metadata about the communication. Current operational practice in some networks inspect transport header information within the network, but this is no longer possible when those transport headers are encrypted.

This document discusses the possible impact when network traffic uses a protocol with an encrypted transport header. It suggests issues to consider when designing new transport protocols or features.

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1. Introduction

The transport layer supports the end-to-end flow of data across a network path, providing features such as connection establishment, reliability, framing, ordering, congestion control, flow control, etc., as needed to support applications. One of the core functions of an Internet transport is to discover and adapt to the characteristics of the network path that is currently being used.

For some years, it has been common for the transport layer payload to be protected by encryption and authentication, but for the transport layer headers to be sent unprotected. Examples of protocols that behave in this manner include Transport Layer Security (TLS) over TCP [RFC8446], Datagram TLS [RFC6347] [I-D.ietf-tls-dtls13], the Secure Real-time Transport Protocol [RFC3711], and tcpcrypt [RFC8548]. The use of unencrypted transport headers has led some network operators, researchers, and others to develop tools and processes that rely on observations of transport headers both in aggregate and at the flow level to infer details of the network's behaviour and inform operational practice.

Transport protocols are now being developed that encrypt some or all of the transport headers, in addition to the transport payload data. The QUIC transport protocol [I-D.ietf-quic-transport] is an example of such a protocol. Such transport header encryption makes it difficult to observe transport protocol behaviour from the vantage point of the network. This document discusses some implications of transport header encryption for network operators and researchers that have previously observed transport headers, and highlights some issues to consider for transport protocol designers.

As discussed in [RFC7258], the IETF has concluded that Pervasive Monitoring (PM) is a technical attack that needs to be mitigated in the design of IETF protocols. This document supports that conclusion. It also recognises that RFC7258 states "Making networks unmanageable to mitigate PM is not an acceptable outcome, but ignoring PM would go against the consensus documented here. An appropriate balance will emerge over time as real instances of this tension are considered". This document is written to provide input to the discussion around what is an appropriate balance, by highlighting some implications of transport header encryption.

Current uses of transport header information by network devices on the Internet path are explained. These uses can be beneficial or malicious. This is written to provide input to the discussion around what is an appropriate balance, by highlighting some implications of transport header encryption.

2. Current uses of Transport Headers within the Network

In response to pervasive monitoring [RFC7624] revelations and the IETF consensus that "Pervasive Monitoring is an Attack" [RFC7258], efforts are underway to increase encryption of Internet traffic. Applying confidentiality to transport header fields can improve privacy, and can help to mitigate certain attacks or manipulation of packets by devices on the network path, but it can also affect network operations and measurement [RFC8404].

When considering what parts of the transport headers should be encrypted to provide confidentiality, and what parts should be visible to network devices (including non-encrypted but authenticated headers), it is necessary to consider both the impact on network operations and management, and the implications for ossification and user privacy [Measurement]. Different parties will view the relative importance of these concerns differently. For some, the benefits of encrypting all the transport headers outweigh the impact of doing so; others might analyse the security, privacy, and ossification impacts and arrive at a different trade-off.

This section reviews examples of the observation of transport layer headers within the network by devices on the network path, or using information exported by an on-path device. Unencrypted transport headers provide information that can support network operations and management, and this section notes some ways in which this has been done. Unencrypted transport header information also contributes metadata that can be exploited for purposes unrelated to network transport measurement, diagnostics or troubleshooting (e.g., to block or to throttle traffic from a specific content provider), and this section also notes some threats relating to unencrypted transport headers.

Exposed transport information also provides a source of information that contributes to linked data sets, which could be exploited to deduce private information, e.g., user patterns, user location, tracking behaviour, etc. This might reveal information the parties did not intend to be revealed. [RFC6973] aims to make designers, implementers, and users of Internet protocols aware of privacy-related design choices in IETF protocols.

This section does not consider intentional modification of transport headers by middleboxes, such as devices performing Network Address Translation (NAT) or Firewalls.

2.1. To Separate Flows in Network Devices

Some network layer mechanisms separate network traffic by flow, without resorting to identifying the type of traffic. Hash-based load-sharing sharing across paths (e.g., equal cost multi path, ECMP), sharing across a group of links (e.g., using a link aggregation group, LAG), ensuring equal access to link capacity (e.g., fair queuing, FQ), or distributing traffic to servers (e.g., load balancing). To prevent packet reordering, forwarding engines can consistently forward the same transport flows along the same forwarding path, often achieved by calculating a hash using an n-tuple gleaned from a combination of link header information through to transport header information. This n-tuple can use the MAC address, IP addresses, and can include observable transport header information.

When transport header information cannot be observed, there can be less information to separate flows at equipment along the path. Flow separation might not be possible when, a transport that forms traffic into an encrypted aggregate. For IPv6, the Flow Label [RFC6437] can be used even when all transport information is encrypted, enabling Flow Label-based ECMP [RFC6438] and Load-Sharing [RFC7098].

2.2. To Identify Transport Protocols and Flows

Information in exposed transport layer headers can be used by the network to identify transport protocols and flows [RFC8558]. The ability to identify transport protocols, flows, and sessions is a common function performed, for example, by measurement activities, Quality of Service (QoS) classifiers, and firewalls. These functions can be beneficial, and performed with the consent of, and in support of, the end user. Alternatively, the same mechanisms could be used to support practises that might be adversarial to the end user, including blocking, de-prioritising, and monitoring traffic without consent.

Observable transport header information, together with information in the network header, has been used to identify flows and their connection state, together with the set of protocol options being used. Transport protocols, such as TCP [RFC7414] and the Stream Control Transport Protocol (SCTP) [RFC4960], specify a standard base header that includes sequence number information and other data. They also have the possibility to negotiate additional headers at connection setup, identified by an option number in the transport header.

In some uses, an assigned transport port (e.g., 0..49151) can identify the upper-layer protocol or service [RFC7605]. However,

port information alone is not sufficient to guarantee identification. Applications can use arbitrary ports and do not need to use assigned port numbers. The use of an assigned port number is also not limited to the protocol for which the port is intended. Multiple sessions can also be multiplexed on a single port, and ports can be re-used by subsequent sessions.

Some flows can be identified by observing signalling data (e.g., [RFC3261], [RFC8837]) or through the use of magic numbers placed in the first byte(s) of a datagram payload [RFC7983].

When transport header information cannot be observed, this removes information that could have been used to classify flows by passive observers along the path. More ambitious ways could be used to collect, estimate, or infer flow information, including heuristics based on the analysis of traffic patterns, such as classification of flows relying on timing, volumes of information, and correlation between multiple flows. For example, an operator that cannot access the Session Description Protocol (SDP) session descriptions [RFC4566] to classify a flow as audio traffic, might instead use (possibly less-reliable) heuristics to infer that short UDP packets with regular spacing carry audio traffic. Operational practises aimed at inferring transport parameters are out of scope for this document, and are only mentioned here to recognise that encryption does not prevent operators from attempting to apply practises that were used with unencrypted transport headers.

The IAB [RFC8546] have provided a summary of expected implications of increased encryption on network functions that use the observable headers and describe the expected benefits of designs that explicitly declare protocol invariant header information that can be used for this purpose.

2.3. To Understand Transport Protocol Performance

This subsection describes use by the network of exposed transport layer headers to understand transport protocol performance and behaviour.

2.3.1. Using Information Derived from Transport Layer Headers

Observable transport headers enable explicit measurement and analysis of protocol performance, and detection of network anomalies at any point along the Internet path. Some operators use passive monitoring to manage their portion of the Internet by characterising the performance of link/network segments. Inferences from transport headers are used to derive performance metrics:

Traffic Rate and Volume: Per-application traffic rate and volume measures can be used to characterise the traffic that uses a network segment or the pattern of network usage. Observing the protocol sequence number and packet size offers one way to measure this (e.g., measurements observing counters in periodic reports such as RTCP; or measurements observing protocol sequence numbers in statistical samples of packet flows, or specific control packets, such as those observed at the start and end of a flow).

Measurements can be per endpoint, or for an endpoint aggregate. These could be used to assess usage or for subscriber billing.

Such measurements can be used to trigger traffic shaping, and to associate QoS support within the network and lower layers. This can be done with consent and in support of an end user, to improve quality of service; or could be used by the network to de-prioritise certain flows without user consent.

The traffic rate and volume can be determined providing that the packets belonging to individual flows can be identified, but there might be no additional information about a flow when the transport headers cannot be observed.

Loss Rate and Loss Pattern: Flow loss rate can be derived (e.g., from transport sequence numbers or inferred from observing transport protocol interactions) and has been used as a metric for performance assessment and to characterise transport behaviour. Network operators have used the variation in patterns to detect changes in the offered service. Understanding the location and root cause of loss can help an operator determine whether this requires corrective action.

There are various causes of loss, including: corruption of link frames (e.g., due to interference on a radio link), buffering loss (e.g., overflow due to congestion, Active Queue Management, AQM [RFC7567], or inadequate provision following traffic pre-emption), and policing (traffic management [RFC2475]). Understanding flow loss rates requires maintaining per-flow state (flow identification often requires transport layer information) and either observing the increase in sequence numbers in the network or transport headers, or comparing a per-flow packet counter with the number of packets that the flow actually sent. Per-hop loss can also sometimes be monitored at the interface level by devices on the network path, or using in-situ methods operating over a network segment (see Section 3.3).

The pattern of loss can provide insight into the cause of loss. Losses can often occur as bursts, randomly-timed events, etc. It

can also be valuable to understand the conditions under which loss occurs. This usually requires relating loss to the traffic flowing at a network node or segment at the time of loss. Transport header information can help identify cases where loss could have been wrongly identified, or where the transport did not require retransmission of a lost packet.

Throughput and Goodput: Throughput is the amount of payload data sent by a flow per time interval. Goodput (the subset of throughput consisting of useful traffic) (see Section 2.5 of [RFC7928] and [RFC5166]) is a measure of useful data exchanged. The throughput of a flow can be determined in the absence of transport header information, providing that the individual flow can be identified, and the overhead known. Goodput requires ability to differentiate loss and retransmission of packets, for example by observing packet sequence numbers in the TCP or RTP headers [RFC3550].

Latency: Latency is a key performance metric that impacts application and user-perceived response times. It often indirectly impacts throughput and flow completion time. This determines the reaction time of the transport protocol itself, impacting flow setup, congestion control, loss recovery, and other transport mechanisms. The observed latency can have many components [Latency]. Of these, unnecessary/unwanted queueing in buffers of the network devices on the path has often been observed as a significant factor [bufferbloat]. Once the cause of unwanted latency has been identified, this can often be eliminated.

To measure latency across a part of a path, an observation point [RFC7799] can measure the experienced round trip time (RTT) using packet sequence numbers and acknowledgements, or by observing header timestamp information. Such information allows an observation point on the network path to determine not only the path RTT, but also allows measurement of the upstream and downstream contribution to the RTT. This could be used to locate a source of latency, e.g., by observing cases where the median RTT is much greater than the minimum RTT for a part of a path.

The service offered by network operators can benefit from latency information to understand the impact of configuration changes and to tune deployed services. Latency metrics are key to evaluating and deploying AQM [RFC7567], DiffServ [RFC2474], and Explicit Congestion Notification (ECN) [RFC3168] [RFC8087]. Measurements could identify excessively large buffers, indicating where to deploy or configure AQM. An AQM method is often deployed in combination with other techniques, such as scheduling [RFC7567] [RFC8290] and although parameter-less methods are desired

[RFC7567], current methods often require tuning [RFC8290] [RFC8289] [RFC8033] because they cannot scale across all possible deployment scenarios.

Latency and round-trip time information can potentially expose some information useful for approximate geolocation, as discussed in [PAM-RTT].

Variation in delay: Some network applications are sensitive to (small) changes in packet timing (jitter). Short and long-term delay variation can impact on the latency of a flow and hence the perceived quality of applications using a network path. For example, jitter metrics are often cited when characterising paths supporting real-time traffic. The expected performance of such applications, can be inferred from a measure of the variation in delay observed along a portion of the path [RFC3393] [RFC5481]. The requirements resemble those for the measurement of latency.

Flow Reordering: Significant packet reordering within a flow can impact time-critical applications and can be interpreted as loss by reliable transports. Many transport protocol techniques are impacted by reordering (e.g., triggering TCP retransmission or re-buffering of real-time applications). Packet reordering can occur for many reasons, from equipment design to misconfiguration of forwarding rules. Flow identification is often required to avoid significant packet mis-ordering (e.g., when using ECMP, or LAG). Network tools can detect and measure unwanted/excessive reordering, and the impact on transport performance.

There have been initiatives in the IETF transport area to reduce the impact of reordering within a transport flow, possibly leading to a reduction in the requirements for preserving ordering. These have potential to simplify network equipment design as well as the potential to improve robustness of the transport service. Measurements of reordering can help understand the present level of reordering, and inform decisions about how to progress new mechanisms.

Techniques for measuring reordering typically observe packet sequence numbers. Metrics have been defined that evaluate whether a network path has maintained packet order on a packet-by-packet basis [RFC4737] [RFC5236]. Some protocols provide in-built monitoring and reporting functions. Transport fields in the RTP header [RFC3550] [RFC4585] can be observed to derive traffic volume measurements and provide information on the progress and quality of a session using RTP. Metadata assists in understanding the context under which the data was collected, including the time, observation point [RFC7799], and way in which metrics were

accumulated. The RTCP protocol directly reports some of this information in a form that can be directly visible by devices on the network path.

In some cases, measurements could involve active injection of test traffic to perform a measurement (see Section 3.4 of [RFC7799]). However, most operators do not have access to user equipment, therefore the point of test is normally different from the transport endpoint. Injection of test traffic can incur an additional cost in running such tests (e.g., the implications of capacity tests in a mobile network segment are obvious). Some active measurements [RFC7799] (e.g., response under load or particular workloads) perturb other traffic, and could require dedicated access to the network segment.

Passive measurements (see Section 3.6 of [RFC7799]) can have advantages in terms of eliminating unproductive test traffic, reducing the influence of test traffic on the overall traffic mix, and the ability to choose the point of observation (see Section 2.4.1). Measurements can rely on observing packet headers, which is not possible if those headers are encrypted, but could utilise information about traffic volumes or patterns of interaction to deduce metrics.

Passive packet sampling techniques are also often used to scale the processing involved in observing packets on high rate links. This exports only the packet header information of (randomly) selected packets. Interpretation of the exported information relies on understanding of the header information. The utility of these measurements depends on the type of network segment/link and number of mechanisms used by the network devices. Simple routers are relatively easy to manage, but a device with more complexity demands understanding of the choice of many system parameters.

2.3.2. Using Information Derived from Network Layer Header Fields

Information from the transport header can be used by a multi-field (MF) classifier as a part of policy framework. Policies are commonly used for management of the QoS or Quality of Experience (QoE) in resource-constrained networks, or by firewalls to implement access rules (see also Section 2.2.2 of [RFC8404]). Policies can support user applications/services or protect against unwanted, or lower priority traffic (Section 2.4.4).

Transport layer information can also be explicitly carried in network-layer header fields that are not encrypted, serving as a replacement/addition to the exposed transport header information [RFC8558]. This information can enable a different forwarding

treatment by the devices forming the network path, even when a transport employs encryption to protect other header information.

On the one hand, the user of a transport that multiplexes multiple sub-flows might want to obscure the presence and characteristics of these sub-flows. On the other hand, an encrypted transport could set the network-layer information to indicate the presence of sub-flows, and to reflect the service requirements of individual sub-flows. There are several ways this could be done:

IP Address: Applications normally expose the endpoint addresses used in the forwarding decisions in network devices. Address and other protocol information can be used by a MF-classifier to determine how traffic is treated [RFC2475], and hence affect the quality of experience for a flow. Common issues concerning IP address sharing are described in [RFC6269].

Using the IPv6 Network-Layer Flow Label: A number of Standards Track and Best Current Practice RFCs (e.g., [RFC8085], [RFC6437], [RFC6438]) encourage endpoints to set the IPv6 flow label field of the network-layer header. IPv6 "source nodes SHOULD assign each unrelated transport connection and application data stream to a new flow" [RFC6437]. A multiplexing transport could choose to use multiple flow labels to allow the network to independently forward sub-flows. RFC6437 provides further guidance on choosing a flow label value, stating these "should be chosen such that their bits exhibit a high degree of variability", and chosen so that "third parties should be unlikely to be able to guess the next value that a source of flow labels will choose".

Once set, a flow label can provide information that can help inform network-layer queueing and forwarding, including use with IPsec, [RFC6294] and use with Equal Cost Multi-Path routing and Link Aggregation [RFC6438].

The choice of how to assign a flow label needs to avoid introducing linkages between flows that a network device could not otherwise observe. Inappropriate use by the transport can have privacy implications (e.g., assigning the same label to two independent flows that ought not to be classified the same).

Using the Network-Layer Differentiated Services Code Point: Applications can expose their delivery expectations to network devices by setting the Differentiated Services Code Point (DSCP) field of IPv4 and IPv6 packets [RFC2474]. For example, WebRTC applications identify different forwarding treatments for individual sub-flows (audio vs. video) based on the value of the DSCP field [I-D.ietf-tsvwg-rtcweb-qos]). This provides explicit

information to inform network-layer queueing and forwarding, rather than an operator inferring traffic requirements from transport and application headers via a multi-field classifier. Inappropriate use by the transport can have privacy implications (e.g., assigning a different DSCP to a subflow could assist in a network device discovering the traffic pattern used by an application). The field is mutable, i.e., some network devices can be expected to change this field. Since the DSCP value can impact the quality of experience for a flow, observations of service performance have to consider this field when a network path supports differentiated service treatment.

Using Explicit Congestion Marking: ECN [RFC3168] is a transport mechanism that uses the ECN field in the network-layer header. Use of ECN explicitly informs the network-layer that a transport is ECN-capable, and requests ECN treatment of the flow. An ECN-capable transport can offer benefits when used over a path with equipment that implements an AQM method with CE marking of IP packets [RFC8087], since it can react to congestion without also having to recover from lost packets.

ECN exposes the presence of congestion. The reception of CE-marked packets can be used to estimate the level of incipient congestion on the upstream portion of the path from the point of observation (Section 2.5 of [RFC8087]). Interpreting the marking behaviour (i.e., assessing congestion and diagnosing faults) requires context from the transport layer, such as path RTT.

AQM and ECN offer a range of algorithms and configuration options. Tools therefore have to be available to network operators and researchers to understand the implication of configuration choices and transport behaviour as the use of ECN increases and new methods emerge [RFC7567].

Network-Layer Options Network protocols can carry optional headers (see Section 5.1). These can explicitly expose transport header information to on-path devices operating at the network layer (as discussed further in Section 6).

IPv4 [RFC0791] has provision for optional header fields. IP routers can examine these headers and are required to ignore IPv4 options that they do not recognise. Many current paths include network devices that forward packets that carry options on a slower processing path. Some network devices (e.g., firewalls) can be (and are) configured to drop these packets [RFC7126]. BCP 186 [RFC7126] provides Best Current Practice guidance on how operators should treat IPv4 packets that specify options.

IPv6 can encode optional network-layer information in separate headers that may be placed between the IPv6 header and the upper-layer header [RFC8200]. (e.g., the IPv6 Alternate Marking Method [I-D.ietf-6man-ipv6-alt-mark], which can be used to measure packet loss and delay metrics). The Hop-by-Hop options header, when present, immediately follows the IPv6 header. IPv6 permits this header to be examined by any node along the path if explicitly configured [RFC8200].

Careful use of the network layer features (e.g., Extension Headers can Section 5) help provide similar information in the case where the network is unable to inspect transport protocol headers.

2.4. To Support Network Operations

Some network operators make use of on-path observations of transport headers to analyse the service offered to the users of a network segment, and to inform operational practice, and can help detect and locate network problems. [RFC8517] gives an operator's perspective about such use.

When observable transport header information is not available, those seeking an understanding of transport behaviour and dynamics might learn to work without that information. Alternatively, they might use more limited measurements combined with pattern inference and other heuristics to infer network behaviour (see Section 2.1.1 of [RFC8404]). Operational practises aimed at inferring transport parameters are out of scope for this document, and are only mentioned here to recognise that encryption does not necessarily stop operators from attempting to apply practises that have been used with unencrypted transport headers.

This section discusses topics concerning observation of transport flows, with a focus on transport measurement.

2.4.1. Problem Location

Observations of transport header information can be used to locate the source of problems or to assess the performance of a network segment. Often issues can only be understood in the context of the other flows that share a particular path, particular device configuration, interface port, etc. A simple example is monitoring of a network device that uses a scheduler or active queue management technique [RFC7567], where it could be desirable to understand whether the algorithms are correctly controlling latency, or if overload protection is working. This implies knowledge of how traffic is assigned to any sub-queues used for flow scheduling, but can require information about how the traffic dynamics impact active

queue management, starvation prevention mechanisms, and circuit-breakers.

Sometimes correlating observations of headers at multiple points along the path (e.g., at the ingress and egress of a network segment), allows an observer to determine the contribution of a portion of the path to an observed metric. e.g., to locate a source of delay, jitter, loss, reordering, or congestion marking.

2.4.2. Network Planning and Provisioning

Traffic rate and volume measurements are used to help plan deployment of new equipment and configuration in networks. Data is also valuable to equipment vendors who want to understand traffic trends and patterns of usage as inputs to decisions about planning products and provisioning for new deployments.

Trends in aggregate traffic can be observed and can be related to the endpoint addresses being used, but when transport header information is not observable, it might be impossible to correlate patterns in measurements with changes in transport protocols. This increases the dependency on other indirect sources of information to inform planning and provisioning.

2.4.3. Compliance with Congestion Control

The traffic that can be observed by on-path network devices (the "wire image") is a function of transport protocol design/options, network use, applications, and user characteristics. In general, when only a small proportion of the traffic has a specific (different) characteristic, such traffic seldom leads to operational concern, although the ability to measure and monitor it is lower. The desire to understand the traffic and protocol interactions typically grows as the proportion of traffic increases. The challenges increase when multiple instances of an evolving protocol contribute to the traffic that share network capacity.

Operators can manage traffic load (e.g., when the network is severely overloaded) by deploying rate-limiters, traffic shaping, or network transport circuit breakers [RFC8084]. The information provided by observing transport headers is a source of data that can help to inform such mechanisms.

Congestion Control Compliance of Traffic: Congestion control is a key transport function [RFC2914]. Many network operators implicitly accept that TCP traffic complies with a behaviour that is acceptable for the shared Internet. TCP algorithms have been continuously improved over decades, and have reached a level of

efficiency and correctness that is difficult to match in custom application-layer mechanisms [RFC8085].

A standards-compliant TCP stack provides congestion control that is judged safe for use across the Internet. Applications developed on top of well-designed transports can be expected to appropriately control their network usage, reacting when the network experiences congestion, by back-off and reduce the load placed on the network. This is the normal expected behaviour for IETF-specified transports (e.g., TCP and SCTP).

Congestion Control Compliance for UDP traffic: UDP provides a minimal message-passing datagram transport that has no inherent congestion control mechanisms. Because congestion control is critical to the stable operation of the Internet, applications and other protocols that choose to use UDP as a transport have to employ mechanisms to prevent collapse, avoid unacceptable contributions to jitter/latency, and to establish an acceptable share of capacity with concurrent traffic [RFC8085].

UDP flows that expose a well-known header can be observed to gain understanding of the dynamics of a flow and its congestion control behaviour. For example, tools exist to monitor various aspects of RTP header information and RTCP reports for real-time flows (see Section 2.3). The Secure RTP and RTCP extensions [RFC3711] were explicitly designed to expose some header information to enable such observation, while protecting the payload data.

A network operator can observe the headers of transport protocols layered above UDP to understand if the datagram flows comply with congestion control expectations. This can help inform a decision on whether it might be appropriate to deploy methods such as rate-limiters to enforce acceptable usage. The available information determines the level of precision with which flows can be classified and the design space for conditioning mechanisms (e.g., rate limiting, circuit breaker techniques [RFC8084], or blocking of uncharacterised traffic) [RFC5218].

When anomalies are detected, tools can interpret the transport header information to help understand the impact of specific transport protocols (or protocol mechanisms) on the other traffic that shares a network. An observer on the network path can gain an understanding of the dynamics of a flow and its congestion control behaviour. Analysing observed flows can help to build confidence that an application flow backs-off its share of the network load under persistent congestion, and hence to understand whether the behaviour is appropriate for sharing limited network capacity. For example, it is common to visualise plots of TCP sequence numbers versus time for

a flow to understand how a flow shares available capacity, deduce its dynamics in response to congestion, etc.

The ability to identify sources and flows that contribute to persistent congestion is important to the safe operation of network infrastructure, and can inform configuration of network devices to complement the endpoint congestion avoidance mechanisms [RFC7567] [RFC8084] to avoid a portion of the network being driven into congestion collapse [RFC2914].

2.4.4. To Characterise "Unknown" Network Traffic

The patterns and types of traffic that share Internet capacity change over time as networked applications, usage patterns and protocols continue to evolve.

Encryption can increase the volume of "unknown" or "uncharacterised" traffic seen by the network. If these traffic patterns form a small part of the traffic aggregate passing through a network device or segment of the network path, the dynamics of the uncharacterised traffic might not have a significant collateral impact on the performance of other traffic that shares this network segment. Once the proportion of this traffic increases, monitoring the traffic can determine if appropriate safety measures have to be put in place.

Tracking the impact of new mechanisms and protocols requires traffic volume to be measured and new transport behaviours to be identified. This is especially true of protocols operating over a UDP substrate. The level and style of encryption needs to be considered in determining how this activity is performed.

Traffic that cannot be classified typically receives a default treatment. Some networks block or rate-limit traffic that cannot be classified.

2.4.5. To Support Network Security Functions

On-path observation of the transport headers of packets can be used for various security functions. For example, Denial of Service (DoS) and Distributed DoS (DDoS) attacks against the infrastructure or against an endpoint can be detected and mitigated by characterising anomalous traffic (see Section 2.4.4) on a shorter timescale. Other uses include support for security audits (e.g., verifying the compliance with cipher suites), client and application fingerprinting for inventory, and to provide alerts for network intrusion detection and other next generation firewall functions.

When using an encrypted transport, endpoints can directly provide information to support these security functions. Another method, if the endpoints do not provide this information, is to use an on-path network device that relies on pattern inferences in the traffic, and heuristics or machine learning instead of processing observed header information. An endpoint could also explicitly cooperate with an on-path device (e.g., a QUIC endpoint could share information about current uses of connection IDs).

2.4.6. Network Diagnostics and Troubleshooting

Operators monitor the health of a network segment to support a variety of operational tasks [RFC8404] including procedures to provide early warning and trigger action: to diagnose network problems, to manage security threats (including DoS), to evaluate equipment or protocol performance, or to respond to user performance questions. Information about transport flows can assist in setting buffer sizes, and help identify whether link/network tuning is effective. Information can also support debugging and diagnosis of the root causes of faults that concern a particular user's traffic and can support post-mortem investigation after an anomaly. Section 3.1.2 and Section 5 of [RFC8404] provide further examples.

Network segments vary in their complexity. The design trade-offs for radio networks are often very different from those of wired networks [RFC8462]. A radio-based network (e.g., cellular mobile, enterprise Wireless LAN (WLAN), satellite access/back-haul, point-to-point radio) adds a subsystem that performs radio resource management, with impact on the available capacity, and potentially loss/reordering of packets. This impact can differ by traffic type, and can be correlated with link propagation and interference. These can impact the cost and performance of a provided service, and is expected to increase in importance as operators bring together heterogeneous types of network equipment and deploy opportunistic methods to access shared radio spectrum.

2.4.7. Tooling and Network Operations

A variety of open source and proprietary tools have been deployed that use the transport header information observable with widely used protocols such as TCP or RTP/UDP/IP. Tools that dissect network traffic flows can alert to potential problems that are hard to derive from volume measurements, link statistics or device measurements alone.

Any introduction of a new transport protocol, protocol feature, or application might require changes to such tools, and so could impact operational practice and policies. Such changes have associated

costs that are incurred by the network operators that need to update their tooling or develop alternative practises that work without access to the changed/removed information.

The use of encryption has the desirable effect of preventing unintended observation of the payload data and these tools seldom seek to observe the payload, or other application details. A flow that hides its transport header information could imply "don't touch" to some operators. This might limit a trouble-shooting response to "can't help, no trouble found".

An alternative that does not require access to observable transport headers is to access endpoint diagnostic tools or to include user involvement in diagnosing and troubleshooting unusual use cases or to troubleshoot non-trivial problems. Another approach is to use traffic pattern analysis. Such tools can provide useful information during network anomalies (e.g., detecting significant reordering, high or intermittent loss), however indirect measurements need to be carefully designed to provide information for diagnostics and troubleshooting.

If new protocols, or protocol extensions, are made to closely resemble or match existing mechanisms, then the changes to tooling and the associated costs can be small. Equally, more extensive changes to the transport tend to require more extensive, and more expensive, changes to tooling and operational practice. Protocol designers can mitigate these costs by explicitly choosing to expose selected information as invariants that are guaranteed not to change for a particular protocol (e.g., the header invariants and the spin-bit in QUIC [I-D.ietf-quic-transport]). Specification of common log formats and development of alternative approaches can also help mitigate the costs of transport changes.

2.5. To Mitigate the Effects of Constrained Networks

Some link and network segments are constrained by the capacity they can offer, by the time it takes to access capacity (e.g., due to under-lying radio resource management methods), or by asymmetries in the design (e.g., many link are designed so that the capacity available is different in the forward and return directions; some radio technologies have different access methods in the forward and return directions resulting from differences in the power budget).

The impact of path constraints can be mitigated using a proxy operating at or above the transport layer to use an alternate transport protocol.

In many cases, one or both endpoints are unaware of the characteristics of the constraining link or network segment and mitigations are applied below the transport layer: Packet classification and QoS methods (described in various sections) can be beneficial in differentially prioritising certain traffic when there is a capacity constraint or additional delay in scheduling link transmissions. Another common mitigation is to apply header compression over the specific link or subnetwork (see Section 2.5.1).

2.5.1. To Provide Header Compression

Header compression saves link capacity by compressing network and transport protocol headers on a per-hop basis. This has been widely used with low bandwidth dial-up access links, and still finds application on wireless links that are subject to capacity constraints. These methods are effective for bit-congestive links sending small packets (e.g., reducing the cost for sending control packets or small data packets over radio links).

Examples of header compression include use with TCP/IP and RTP/UDP/IP flows [RFC2507], [RFC6846], [RFC2508], [RFC5795], [RFC8724]. Successful compression depends on observing the transport headers and understanding of the way fields change between packets, and is hence incompatible with header encryption. Devices that compress transport headers are dependent on a stable header format, implying ossification of that format.

Introducing a new transport protocol, or changing the format of the transport header information, will limit the effectiveness of header compression until the network devices are updated. Encrypting the transport protocol headers will tend to cause the header compression to fall back to compressing only the network layer headers, with a significant reduction in efficiency. This can limit connectivity if the resulting flow exceeds the link capacity, or if the packets are dropped because they exceed the link MTU.

The Secure RTP (SRTP) extensions [RFC3711] were explicitly designed to leave the transport protocol headers unencrypted, but authenticated, since support for header compression was considered important.

2.6. To Verify SLA Compliance

Observable transport headers coupled with published transport specifications allow operators and regulators to explore and verify compliance with Service Level Agreements (SLAs). It can also be used to understand whether a service is providing differential treatment to certain flows.

When transport header information cannot be observed, other methods have to be found to confirm that the traffic produced conforms to the expectations of the operator or developer.

Independently verifiable performance metrics can be utilised to demonstrate regulatory compliance in some jurisdictions, and as a basis for informing design decisions. This can bring assurance to those operating networks, often avoiding deployment of complex techniques that routinely monitor and manage Internet traffic flows (e.g., avoiding the capital and operational costs of deploying flow rate-limiting and network circuit-breaker methods [RFC8084]).

3. Research, Development and Deployment

Research and development of new protocols and mechanisms need to be informed by measurement data (as described in the previous section). Data can also help promote acceptance of proposed standards specifications by the wider community (e.g., as a method to judge the safety for Internet deployment).

Observed data is important to ensure the health of the research and development communities, and provides data needed to evaluate new proposals for standardisation. Open standards motivate a desire to include independent observation and evaluation of performance and deployment data. Independent data helps compare different methods, judge the level of deployment and ensure the wider applicability of the results. This is important when considering when a protocol or mechanism should be standardised for use in the general Internet. This, in turn, demands control/understanding about where and when measurement samples are collected. This requires consideration of the methods used to observe information and the appropriate balance between encrypting all and no transport header information.

There can be performance and operational trade-offs in exposing selected information to network tools. This section explores key implications of tools and procedures that observe transport protocols, but does not endorse or condemn any specific practises.

3.1. Independent Measurement

Encrypting transport header information has implications on the way network data is collected and analysed. Independent observation by multiple actors is currently used by the transport community to maintain an accurate understanding of the network within transport area working groups, IRTF research groups, and the broader research community. This is important to be able to provide accountability, and demonstrate that protocols behave as intended, although when providing or using such information, it is important to consider the

privacy of the user and their incentive for providing accurate and detailed information.

Protocols that expose the state of the transport protocol in their header (e.g., timestamps used to calculate the RTT, packet numbers used to assess congestion and requests for retransmission) provide an incentive for a sending endpoint to provide consistent information, because a protocol will not work otherwise. An on-path observer can have confidence that well-known (and ossified) transport header information represents the actual state of the endpoints, when this information is necessary for the protocol's correct operation.

Encryption of transport header information could reduce the range of actors that can observe useful data. This would limit the information sources available to the Internet community to understand the operation of new transport protocols, reducing information to inform design decisions and standardisation of the new protocols and related operational practises. The cooperating dependence of network, application, and host to provide communication performance on the Internet is uncertain when only endpoints (i.e., at user devices and within service platforms) can observe performance, and when performance cannot be independently verified by all parties.

3.2. Measurable Transport Protocols

Transport protocol evolution, and the ability to measure and understand the impact of protocol changes, have to proceed hand-in-hand. A transport protocol that provides observable headers can be used to provide open and verifiable measurement data. Observation of pathologies has a critical role in the design of transport protocol mechanisms and development of new mechanisms and protocols, and aides understanding of the interactions between cooperating protocols and network mechanisms, the implications of sharing capacity with other traffic and the impact of different patterns of usage. The ability of other stakeholders to review transport header traces helps develop insight into the performance and the traffic contribution of specific variants of a protocol.

Development of new transport protocol mechanisms has to consider the scale of deployment and the range of environments in which the transport is used. Experience has shown that it is often difficult to correctly implement new mechanisms [RFC8085], and that mechanisms often evolve as a protocol matures, or in response to changes in network conditions, changes in network traffic, or changes to application usage. Analysis is especially valuable when based on the behaviour experienced across a range of topologies, vendor equipment, and traffic patterns.

Encryption enables a transport protocol to choose which internal state to reveal to devices on the network path, what information to encrypt, and what fields to grease [RFC8701]. A new design can provide summary information regarding its performance, congestion control state, etc., or to make available explicit measurement information. For example, [I-D.ietf-quic-transport] specifies a way for a QUIC endpoint to optionally set the spin-bit to explicitly reveal the RTT of an encrypted transport session to the on-path network devices. There is a choice of what information to expose. For some operational uses, the information has to contain sufficient detail to understand, and possibly reconstruct, the network traffic pattern for further testing. The interpretation of the information needs to consider whether this information reflects the actual transport state of the endpoints. This might require the trust of transport protocol implementers, to correctly reveal the desired information.

New transport protocol formats are expected to facilitate an increased pace of transport evolution, and with it the possibility to experiment with and deploy a wide range of protocol mechanisms. At the time of writing, there has been interest in a wide range of new transport methods, e.g., Larger Initial Window, Proportional Rate Reduction (PRR), congestion control methods based on measuring bottleneck bandwidth and round-trip propagation time, the introduction of AQM techniques and new forms of ECN response (e.g., Data Centre TCP, DCTCP, and methods proposed for L4S). The growth and diversity of applications and protocols using the Internet also continues to expand. For each new method or application, it is desirable to build a body of data reflecting its behaviour under a wide range of deployment scenarios, traffic load, and interactions with other deployed/candidate methods.

3.3. Other Sources of Information

Some measurements that traditionally rely on observable transport information could be completed by utilising endpoint-based logging (e.g., based on Quic-Trace [Quic-Trace] and qlog [I-D.marx-qlog-main-schema]). Such information has a diversity of uses, including developers wishing to debug/understand the transport/application protocols with which they work, researchers seeking to spot trends and anomalies, and to characterise variants of protocols. A standard format for endpoint logging could allow these to be shared (after appropriate anonymisation) to understand performance and pathologies.

When measurement datasets are made available by servers or client endpoints, additional metadata, such as the state of the network and conditions in which the system was observed, is often necessary to

interpret this data to answer questions about network performance or understand a pathology. Collecting and coordinating such metadata is more difficult when the observation point is at a different location to the bottleneck or device under evaluation [RFC7799].

Despite being applicable in some scenarios, endpoint logs do not provide equivalent information to on-path measurements made by devices in the network. In particular, endpoint logs contain only a part of the information to understand the operation of network devices and identify issues such as link performance or capacity sharing between multiple flows. An analysis can require coordination between actors at different layers to successfully characterise flows and correlate the performance or behaviour of a specific mechanism with an equipment configuration and traffic using operational equipment along a network path (e.g., combining transport and network measurements to explore congestion control dynamics, to understand the implications of traffic on designs for active queue management or circuit breakers).

Another source of information could arise from operations, administration and management (OAM) (see Section 6) information data records could be embedded into header information at different layers to support functions such as performance evaluation, path-tracing, path verification information, classification and a diversity of other uses.

In-situ OAM (IOAM) data fields [I-D.ietf-ippm-ioam-data] can be encapsulated into a variety of protocols to record operational and telemetry information in an existing packet, while that packet traverses a part of the path between two points in a network (e.g., within a particular IOAM management domain). The IOAM-Data-Fields are independent from the protocols into which the IOAM-Data-Fields are encapsulated. For example, IOAM can provide proof that a certain traffic flow takes a pre-defined path, SLA verification for the live data traffic, and statistics relating to traffic distribution.

4. Encryption and Authentication of Transport Headers

There are several motivations for transport header encryption.

One motive to encrypt transport headers is to prevent network ossification from network devices that inspect well-known transport headers. Once a network device observes a transport header and becomes reliant upon using it, the overall use of that field can become ossified, preventing new versions of the protocol and mechanisms from being deployed. Examples include:

- o During the development of TLS 1.3 [RFC8446], the design needed to function in the presence of deployed middleboxes that relied on the presence of certain header fields exposed in TLS 1.2 [RFC5426].
- o The design of Multipath TCP (MPTCP) [RFC8684] had to account for middleboxes (known as "TCP Normalizers") that monitor the evolution of the window advertised in the TCP header and then reset connections when the window did not grow as expected.
- o TCP Fast Open [RFC7413] can experience problems due to middleboxes that modify the transport header of packets by removing "unknown" TCP options. Segments with unrecognised TCP options can be dropped, segments that contain data and set the SYN bit can be dropped, and some middleboxes that disrupt connections that send data before completion of the three-way handshake.
- o Other examples of TCP ossification have included middleboxes that modify transport headers by rewriting TCP sequence and acknowledgement numbers, but are unaware of the (newer) TCP selective acknowledgement (SACK) option and therefore fail to correctly rewrite the SACK information to match the changes made to the fixed TCP header, preventing correct SACK operation.

In all these cases, middleboxes with a hard-coded, but incomplete, understanding of a specific transport behaviour (i.e., TCP), interacted poorly with transport protocols after the transport behaviour was changed. In some cases, the middleboxes modified or replaced information in the transport protocol header.

Transport header encryption prevents an on-path device from observing the transport headers, and therefore stops ossified mechanisms being used that directly rely on or infer semantics of the transport header information. This encryption is normally combined with authentication of the protected information. RFC 8546 summarises this approach, stating that it is "The wire image, not the protocol's specification, determines how third parties on the network paths among protocol participants will interact with that protocol" (Section 1 of [RFC8546]), and it can be expected that header information that is not encrypted will become ossified.

Encryption does not itself prevent ossification of the network service. People seeking to understand or classify network traffic could still come to rely on pattern inferences and other heuristics or machine learning to derive measurement data and as the basis for network forwarding decisions [RFC8546]. This can also create dependencies on the transport protocol, or the patterns of traffic it can generate, also resulting in ossification of the service.

Another motivation for using transport header encryption is to improve privacy and to decrease opportunities for surveillance. Users value the ability to protect their identity and location, and defend against analysis of the traffic. Revelations about the use of pervasive surveillance [RFC7624] have, to some extent, eroded trust in the service offered by network operators and have led to an increased use of encryption. Concerns have also been voiced about the addition of metadata to packets by third parties to provide analytics, customisation, advertising, cross-site tracking of users, to bill the customer, or to selectively allow or block content.

Whatever the reasons, the IETF is designing protocols that include transport header encryption (e.g., QUIC [I-D.ietf-quic-transport]) to supplement the already widespread payload encryption, and to further limit exposure of transport metadata to the network.

If a transport protocol uses header encryption, the designers have to decide whether to encrypt all, or a part of, the transport layer information. Section 4 of [RFC8558] states: "Anything exposed to the path should be done with the intent that it be used by the network elements on the path".

Certain transport header fields can be made observable to on-path network devices, or can define new fields designed to explicitly expose observable transport layer information to the network. Where exposed fields are intended to be immutable (i.e., can be observed, but not modified by a network device), the endpoints are encouraged to use authentication to provide a cryptographic integrity check that can detect if these immutable fields have been modified by network devices. Authentication can help to prevent attacks that rely on sending packets that fake exposed control signals in transport headers (e.g., TCP RST spoofing). Making a part of a transport header observable or exposing new header fields can lead to ossification of that part of a header as network devices come to rely on observations of the exposed fields.

The use of transport header authentication and encryption therefore exposes a tussle between middlebox vendors, operators, researchers, applications developers, and end-users:

- o On the one hand, future Internet protocols that support transport header encryption assist in the restoration of the end-to-end nature of the Internet by returning complex processing to the endpoints. Since middleboxes cannot modify what they cannot see, the use of transport header encryption can improve application and end-user privacy by reducing leakage of transport metadata to operators that deploy middleboxes.

- o On the other hand, encryption of transport layer information has implications for network operators and researchers seeking to understand the dynamics of protocols and traffic patterns, since it reduces the information that is available to them.

The following briefly reviews some security design options for transport protocols. A Survey of the Interaction between Security Protocols and Transport Services [RFC8922] provides more details concerning commonly used encryption methods at the transport layer.

Security work typically employs a design technique that seeks to expose only what is needed [RFC3552]. This approach provides incentives to not reveal any information that is not necessary for the end-to-end communication. The IETF has provided guidelines for writing Security Considerations for IETF specifications [RFC3552].

Endpoint design choices impacting privacy also need to be considered as a part of the design process [RFC6973]. The IAB has provided guidance for analyzing and documenting privacy considerations within IETF specifications [RFC6973].

Authenticating the Transport Protocol Header: Transport layer header information can be authenticated. An example transport authentication mechanism is TCP-Authentication (TCP-AO) [RFC5925]. This TCP option authenticates the IP pseudo header, TCP header, and TCP data. TCP-AO protects the transport layer, preventing attacks from disabling the TCP connection itself and provides replay protection. Such authentication might interact with middleboxes, depending on their behaviour [RFC3234].

The IPsec Authentication Header (AH) [RFC4302] was designed to work at the network layer and authenticate the IP payload. This approach authenticates all transport headers, and verifies their integrity at the receiver, preventing modification by network devices on the path. The IPsec Encapsulating Security Payload (ESP) [RFC4303] can also provide authentication and integrity without confidentiality using the NULL encryption algorithm [RFC2410]. SRTP [RFC3711] is another example of a transport protocol that allows header authentication.

Integrity Check Transport protocols usually employ integrity checks on the transport header information. Security method usually employ stronger checks and can combine this with authentication. An integrity check that protects the immutable transport header fields, but can still expose the transport header information in the clear, allows on-path network devices to observe these fields. An integrity check is not able to prevent modification by network devices on the path, but can prevent a receiving endpoint from

accepting changes and avoid impact on the transport protocol operation, including some types of attack.

Selectively Encrypting Transport Headers and Payload: A transport protocol design that encrypts selected header fields, allows specific transport header fields to be made observable by network devices on the path. This information is explicitly exposed either in a transport header field or lower layer protocol header. A design that only exposes immutable fields can also perform end-to-end authentication of these fields across the path to prevent undetected modification of the immutable transport headers.

Mutable fields in the transport header provide opportunities where on-path network devices can modify the transport behaviour (e.g., the extended headers described in [I-D.trammell-plus-abstract-mech]). An example of a method that encrypts some, but not all, transport header information is GRE-in-UDP [RFC8086] when used with GRE encryption.

Optional Encryption of Header Information: There are implications to the use of optional header encryption in the design of a transport protocol, where support of optional mechanisms can increase the complexity of the protocol and its implementation, and in the management decisions that have to be made to use variable format fields. Instead, fields of a specific type ought to be sent with the same level of confidentiality or integrity protection.

Greasing: Protocols often provide extensibility features, reserving fields or values for use by future versions of a specification. The specification of receivers has traditionally ignored unspecified values, however on-path network devices have emerged that ossify to require a certain value in a field, or re-use a field for another purpose. When the specification is later updated, it is impossible to deploy the new use of the field, and forwarding of the protocol could even become conditional on a specific header field value.

A protocol can intentionally vary the value, format, and/or presence of observable transport header fields at random [RFC8701]. This prevents a network device ossifying the use of a specific observable field and can ease future deployment of new uses of the value or code-point. This is not a security mechanism, although the use can be combined with an authentication mechanism.

Different transports use encryption to protect their header information to varying degrees. The trend is towards increased protection.

5. Intentionally Exposing Transport Information to the Network

A transport protocol can choose to expose certain transport information to on-path devices operating at the network layer by sending observable fields. One approach is to make an explicit choice not to encrypt certain transport header fields, making this transport information observable by an on-path network device. Another approach is to expose transport information in a network-layer extension header (see Section 5.1). Both are examples of explicit information intended to be used by network devices on the path [RFC8558].

Whatever the mechanism used to expose the information, a decision to expose only specific information places the transport endpoint in control of what to expose outside of the encrypted transport header. This decision can then be made independently of the transport protocol functionality. This can be done by exposing part of the transport header or as a network layer option/extension.

5.1. Exposing Transport Information in Extension Headers

At the network-layer, packets can carry optional headers that explicitly expose transport header information to the on-path devices operating at the network layer (Section 2.3.2). For example, an endpoint that sends an IPv6 Hop-by-Hop option [RFC8200] can provide explicit transport layer information that can be observed and used by network devices on the path. New hop-by-hop options are not recommended in RFC 8200 [RFC8200] "because nodes may be configured to ignore the Hop-by-Hop Options header, drop packets containing a Hop-by-Hop Options header, or assign packets containing a Hop-by-Hop Options header to a slow processing path. Designers considering defining new hop-by-hop options need to be aware of this likely behavior."

Network-layer optional headers explicitly indicate the information that is exposed, whereas use of exposed transport header information first requires an observer to identify the transport protocol and its format. (See Section 2.2.)

An arbitrary path can include one or more network devices that drop packets that include a specific header or option used for this purpose (see [RFC7872]). This could impact the proper functioning of the protocols using the path. Protocol methods can be designed to probe to discover whether the specific option(s) can be used along the current path, enabling use on arbitrary paths.

5.2. Common Exposed Transport Information

There are opportunities for multiple transport protocols to consistently supply common observable information [RFC8558]. A common approach can result in an open definition of the observable fields. This has the potential that the same information can be utilised across a range of operational and analysis tools.

5.3. Considerations for Exposing Transport Information

Considerations concerning what information, if any, it is appropriate to expose include:

- o On the one hand, explicitly exposing derived fields containing relevant transport information (e.g., metrics for loss, latency, etc) can avoid network devices needing to derive this information from other header fields. This could result in development and evolution of transport-independent tools around a common observable header, and permit transport protocols to also evolve independently of this ossified header [RFC8558].
- o On the other hand, protocols and implementations might be designed to avoid consistently exposing external information that corresponds to the actual internal information used by the protocol itself. An endpoint/protocol could choose to expose transport header information to optimise the benefit it gets from the network [RFC8558]. The value of this information for analysing operation of the transport layer would be enhanced if the exposed information could be verified to match the transport protocol's observed behavior.

The motivation to include actual transport header information and the implications of network devices using this information has to be considered when proposing such a method. RFC 8558 summarises this as "When signals from endpoints to the path are independent from the signals used by endpoints to manage the flow's state mechanics, they may be falsified by an endpoint without affecting the peer's understanding of the flow's state. For encrypted flows, this divergence is not detectable by on-path devices [RFC8558].

6. Addition of Transport OAM Information to Network-Layer Headers

Even when the transport headers are encrypted, on-path devices can make measurements by utilising additional protocol headers carrying OAM information in an additional packet header. OAM information can be included with packets to perform functions such as identification of transport protocols and flows, to aide understanding of network or

transport performance, or to support network operations or mitigate the effects of specific network segments.

Using network-layer approaches to reveal information has the potential that the same method (and hence same observation and analysis tools) can be consistently used by multiple transport protocols. This approach also could be applied to methods beyond OAM (see Section 5). There can also be less desirable implications from separating the operation of the transport protocol from the measurement framework.

6.1. Use of OAM within a Maintenance Domain

OAM information can be restricted to a maintenance domain, typically owned and operated by a single entity. OAM information can be added at the ingress to the maintenance domain (e.g., an Ethernet protocol header with timestamps and sequence number information using a method such as 802.1lag or in-situ OAM [I-D.ietf-ippm-ioam-data], or as a part of the encapsulation protocol). This additional header information is not delivered to the endpoints and is typically removed at the egress of the maintenance domain.

Although some types of measurements are supported, this approach does not cover the entire range of measurements described in this document. In some cases, it can be difficult to position measurement tools at the appropriate segments/nodes and there can be challenges in correlating the downstream/upstream information when in-band OAM data is inserted by an on-path device.

6.2. Use of OAM across Multiple Maintenance Domains

OAM information can also be added at the network layer by the sender as an IPv6 extension header or an IPv4 option, or in an encapsulation/tunnel header that also includes an extension header or option. This information can be used across multiple network segments, or between the transport endpoints.

One example is the IPv6 Performance and Diagnostic Metrics (PDM) destination option [RFC8250]. This allows a sender to optionally include a destination option that carries header fields that can be used to observe timestamps and packet sequence numbers. This information could be authenticated by a receiving transport endpoint when the information is added at the sender and visible at the receiving endpoint, although methods to do this have not currently been proposed. This needs to be explicitly enabled at the sender.

7. Conclusions

Header encryption and strong integrity checks are being incorporated into new transport protocols and have important benefits. The pace of development of transports using the WebRTC data channel, and the rapid deployment of the QUIC transport protocol, can both be attributed to using the combination of UDP as a substrate while providing confidentiality and authentication of the encapsulated transport headers and payload.

This document has described some current practises, and the implications for some stakeholders, when transport layer header encryption is used. It does not judge whether these practises are necessary, or endorse the use of any specific practise. Rather, the intent is to highlight operational tools and practises to consider when designing and modifying transport protocols, so protocol designers can make informed choices about what transport header fields to encrypt, and whether it might be beneficial to make an explicit choice to expose certain fields to devices on the network path. In making such a decision, it is important to balance:

- o User Privacy: The less transport header information that is exposed to the network, the lower the risk of leaking metadata that might have user privacy implications. Transports that chose to expose some header fields need to make a privacy assessment to understand the privacy cost versus benefit trade-off in making that information available. The design of the QUIC spin bit to the network is an example of such considered analysis.
- o Transport Ossification: Unencrypted transport header fields are likely to ossify rapidly, as network devices come to rely on their presence, making it difficult to change the transport in future. This argues that the choice to expose information to the network is made deliberately and with care, since it is essentially defining a stable interface between the transport and the network. Some protocols will want to make that interface as limited as possible; other protocols might find value in exposing certain information to signal to the network, or in allowing the network to change certain header fields as signals to the transport. The visible wire image of a protocol should be explicitly designed.
- o Network Ossification: While encryption can reduce ossification of the transport protocol, it does not itself prevent ossification of the network service. People seeking to understand network traffic could still come to rely on pattern inferences and other heuristics or machine learning to derive measurement data and as the basis for network forwarding decisions [RFC8546]. This

creates dependencies on the transport protocol, or the patterns of traffic it can generate, resulting in ossification of the service.

- o **Impact on Operational Practice:** The network operations community has long relied on being able to understand Internet traffic patterns, both in aggregate and at the flow level, to support network management, traffic engineering, and troubleshooting. Operational practice has developed based on the information available from unencrypted transport headers. The IETF has supported this practice by developing operations and management specifications, interface specifications, and associated Best Current Practises. Widespread deployment of transport protocols that encrypt their information will impact network operations, unless operators can develop alternative practises that work without access to the transport header.
- o **Pace of Evolution:** Removing obstacles to change can enable an increased pace of evolution. If a protocol changes its transport header format (wire image), or its transport behaviour, this can result in the currently deployed tools and methods becoming no longer relevant. Where this needs to be accompanied by development of appropriate operational support functions and procedures, it can incur a cost in new tooling to catch-up with each change. Protocols that consistently expose observable data do not require such development, but can suffer from ossification and need to consider if the exposed protocol metadata has privacy implications. There is no single deployment context, and therefore designers need to consider the diversity of operational networks (ISPs, enterprises, DDoS mitigation and firewall maintainers, etc.).
- o **Supporting Common Specifications:** Common, open, transport specifications can stimulate engagement by developers, users, researchers, and the broader community. Increased protocol diversity can be beneficial in meeting new requirements, but the ability to innovate without public scrutiny risks point solutions that optimise for specific cases, and that can accidentally disrupt operations of/in different parts of the network. The social contract that maintains the stability of the Internet relies on accepting common transport specifications, and on it being possible to detect violations. The existence of independent measurements, transparency, and public scrutiny of transport protocol behaviour, help the community to enforce the social norm that protocol implementations behave fairly and conform (at least mostly) to the specifications. It is important to find new ways of maintaining that community trust as increased use of transport header encryption limits visibility into transport behaviour (see also Section 5.3).

- o Impact on Benchmarking and Understanding Feature Interactions: An appropriate vantage point for observation, coupled with timing information about traffic flows, provides a valuable tool for benchmarking network devices, endpoint stacks, and/or configurations. This can help understand complex feature interactions. An inability to observe transport header information can make it harder to diagnose and explore interactions between features at different protocol layers, a side-effect of not allowing a choice of vantage point from which this information is observed. New approaches might have to be developed.
- o Impact on Research and Development: Hiding transport header information can impede independent research into new mechanisms, measurement of behaviour, and development initiatives. Experience shows that transport protocols are complicated to design and complex to deploy, and that individual mechanisms have to be evaluated while considering other mechanisms, across a broad range of network topologies and with attention to the impact on traffic sharing the capacity. If increased use of transport header encryption results in reduced availability of open data, it could eliminate the independent checks to the standardisation process that have previously been in place from research and academic contributors (e.g., the role of the IRTF Internet Congestion Control Research Group (ICCRG) and research publications in reviewing new transport mechanisms and assessing the impact of their deployment).

Observable transport header information might be useful to various stakeholders. Other sets of stakeholders have incentives to limit what can be observed. This document does not make recommendations about what information ought to be exposed, to whom it ought to be observable, or how this will be achieved. There are also design choices about where observable fields are placed. For example, one location could be a part of the transport header outside of the encryption envelope, another alternative is to carry the information in a network-layer option or extension header. New transport protocol designs ought to explicitly identify any fields that are intended to be observed, consider if there are alternative ways of providing the information, and reflect on the implications of observable fields being used by on-path network devices, and how this might impact user privacy and protocol evolution when these fields become ossified.

As [RFC7258] notes, "Making networks unmanageable to mitigate (pervasive monitoring) is not an acceptable outcome, but ignoring (pervasive monitoring) would go against the consensus documented here." Providing explicit information can help avoid traffic being

inappropriately classified, impacting application performance. An appropriate balance will emerge over time as real instances of this tension are analysed [RFC7258]. This balance between information exposed and information hidden ought to be carefully considered when specifying new transport protocols.

8. Security Considerations

This document is about design and deployment considerations for transport protocols. Issues relating to security are discussed throughout this document.

Authentication, confidentiality protection, and integrity protection are identified as Transport Features by [RFC8095]. As currently deployed in the Internet, these features are generally provided by a protocol or layer on top of the transport protocol [RFC8922].

Confidentiality and strong integrity checks have properties that can also be incorporated into the design of a transport protocol or to modify an existing transport. Integrity checks can protect an endpoint from undetected modification of protocol fields by on-path network devices, whereas encryption and obfuscation or greasing can further prevent these headers being utilised by network devices [RFC8701]. Preventing observation of headers provides an opportunity for greater freedom to update the protocols and can ease experimentation with new techniques and their final deployment in endpoints. A protocol specification needs to weigh the costs of ossifying common headers, versus the potential benefits of exposing specific information that could be observed along the network path to provide tools to manage new variants of protocols.

Header encryption can provide confidentiality of some or all of the transport header information. This prevents an on-path device from gaining knowledge of the header field. It therefore prevents mechanisms being built that directly rely on the information or seeks to infer semantics of an exposed header field. Reduced visibility into transport metadata can limit the ability to measure and characterise traffic, and conversely can provide privacy benefits.

Extending the transport payload security context to also include the transport protocol header protects both types of information with the same key. A privacy concern would arise if this key was shared with a third party, e.g., providing access to transport header information to debug a performance issue, would also result in exposing the transport payload data to the same third party. Such risks would be mitigated using a layered security design that provides one domain of protection and associated keys for the transport payload and

encrypted transport headers; and a separate domain of protection and associated keys for any observable transport header fields.

Exposed transport headers are sometimes utilised as a part of the information to detect anomalies in network traffic. "While PM is an attack, other forms of monitoring that might fit the definition of PM can be beneficial and not part of any attack, e.g., network management functions monitor packets or flows and anti-spam mechanisms need to see mail message content." [RFC7258]. This can be used as the first line of defence to identify potential threats from DoS or malware and redirect suspect traffic to dedicated nodes responsible for DoS analysis, malware detection, or to perform packet "scrubbing" (the normalisation of packets so that there are no ambiguities in interpretation by the ultimate destination of the packet). These techniques are currently used by some operators to also defend from distributed DoS attacks.

Exposed transport header fields can also form a part of the information used by the receiver of a transport protocol to protect the transport layer from data injection by an attacker. In evaluating this use of exposed header information, it is important to consider whether it introduces a significant DoS threat. For example, an attacker could construct a DoS attack by sending packets with a sequence number that falls within the currently accepted range of sequence numbers at the receiving endpoint. This would then introduce additional work at the receiving endpoint, even though the data in the attacking packet might not finally be delivered by the transport layer. This is sometimes known as a "shadowing attack". An attack can, for example, disrupt receiver processing, trigger loss and retransmission, or make a receiving endpoint perform unproductive decryption of packets that cannot be successfully decrypted (forcing a receiver to commit decryption resources, or to update and then restore protocol state).

One mitigation to off-path attack is to deny knowledge of what header information is accepted by a receiver or obfuscate the accepted header information, e.g., setting a non-predictable initial value for a sequence number during a protocol handshake, as in [RFC3550] and [RFC6056], or a port value that cannot be predicted (see Section 5.1 of [RFC8085]). A receiver could also require additional information to be used as a part of a validation check before accepting packets at the transport layer (e.g., utilising a part of the sequence number space that is encrypted; or by verifying an encrypted token not visible to an attacker). This would also mitigate against on-path attacks. An additional processing cost can be incurred when decryption is attempted before a receiver discards an injected packet.

The existence of open transport protocol standards, and a research and operations community with a history of independent observation and evaluation of performance data, encourages fairness and conformance to those standards. This suggests careful consideration will be made over where, and when, measurement samples are collected. An appropriate balance between encrypting some or all of the transport header information needs to be considered. Open data, and accessibility to tools that can help understand trends in application deployment, network traffic and usage patterns can all contribute to understanding security challenges.

The Security and Privacy Considerations in the Framework for Large-Scale Measurement of Broadband Performance (LMAP) [RFC7594] contain considerations for Active and Passive measurement techniques and supporting material on measurement context.

Addition of observable transport information to the path increases the information available to an observer and may, when this information can be linked to a node or user, reduce the privacy of the user. See the security considerations of [RFC8558].

9. IANA Considerations

This memo includes no request to IANA.

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Appendix A. Revision information

- 00 This is an individual draft for the IETF community.
 - 01 This draft was a result of walking away from the text for a few days and then reorganising the content.
 - 02 This draft fixes textual errors.
 - 03 This draft follows feedback from people reading this draft.
 - 04 This adds an additional contributor and includes significant reworking to ready this for review by the wider IETF community Colin Perkins joined the author list.
- Comments from the community are welcome on the text and recommendations.
- 05 Corrections received and helpful inputs from Mohamed Boucadair.
 - 06 Updated following comments from Stephen Farrell, and feedback via email. Added a draft conclusion section to sketch some strawman scenarios that could emerge.
 - 07 Updated following comments from Al Morton, Chris Seal, and other feedback via email.
 - 08 Updated to address comments sent to the TSVWG mailing list by Kathleen Moriarty (on 08/05/2018 and 17/05/2018), Joe Touch on 11/05/2018, and Spencer Dawkins.
 - 09 Updated security considerations.
 - 10 Updated references, split the Introduction, and added a paragraph giving some examples of why ossification has been an issue.
 - 01 This resolved some reference issues. Updated section on observation by devices on the path.
 - 02 Comments received from Kyle Rose, Spencer Dawkins and Tom Herbert. The network-layer information has also been re-organised after comments at IETF-103.
 - 03 Added a section on header compression and rewriting of sections referring to RTP transport. This version contains author editorial work and removed duplicate section.
 - 04 Revised following SecDir Review

- o Added some text on TLS story (additional input sought on relevant considerations).
- o Section 2, paragraph 8 - changed to be clearer, in particular, added "Encryption with secure key distribution prevents"
- o Flow label description rewritten based on PS/BCP RFCs.
- o Clarify requirements from RFCs concerning the IPv6 flow label and highlight ways it can be used with encryption. (section 3.1.3)
- o Add text on the explicit spin-bit work in the QUIC DT. Added greasing of spin-bit. (Section 6.1)
- o Updated section 6 and added more explanation of impact on operators.
- o Other comments addressed.

-05 Editorial pass and minor corrections noted on TSVWG list.

-06 Updated conclusions and minor corrections. Responded to request to add OAM discussion to Section 6.1.

-07 Addressed feedback from Ruediger and Thomas.

Section 2 deserved some work to make it easier to read and avoid repetition. This edit finally gets to this, and eliminates some duplication. This also moves some of the material from section 2 to reform a clearer conclusion. The scope remains focussed on the usage of transport headers and the implications of encryption - not on proposals for new techniques/specifications to be developed.

-08 Addressed feedback and completed editorial work, including updating the text referring to RFC7872, in preparation for a WGLC.

-09 Updated following WGLC. In particular, thanks to Joe Touch (specific comments and commentary on style and tone); Dimitri Tikonov (editorial); Christian Huitema (various); David Black (various). Amended privacy considerations based on SECDIR review. Emile Stephan (inputs on operations measurement); Various others.

Added summary text and refs to key sections. Note to editors: The section numbers are hard-linked.

-10 Updated following additional feedback from 1st WGLC. Comments from David Black; Tommy Pauly; Ian Swett; Mirja Kuehlewind; Peter

Gutmann; Ekr; and many others via the TSVWG list. Some people thought that "needed" and "need" could

represent requirements in the document, etc. this has been clarified.

-11 Updated following additional feedback from Martin Thomson, and corrections from other reviewers.

-12 Updated following additional feedback from reviewers.

-13 Updated following 2nd WGLC with comments from D.L.Black; T. Herbert; Ekr; and other reviewers.

-14 Update to resolve feedback to rev -13. This moves the general discussion of adding fields to transport packets to section 6, and discusses with reference to material in RFC8558.

-15 Feedback from D.L. Black, T. Herbert, J. Touch, S. Dawkins and M. Duke. Update to add reference to RFC7605. Clarify a focus on immutable transport fields, rather than modifying middleboxes with Tom H. Clarified Header Compression discussion only provides a list of examples of HC methods for transport. Clarified port usage with Tom H/Joe T. Removed some duplicated sentences, and minor edits. Added NULL-ESP. Improved after initial feedback from Martin Duke.

-16 Editorial comments from Mohamed Boucadair. Added DTLS 1.3.

-17 Revised to satisfy ID-NITs and updates REFs to latest rev, updated HC Refs; cited IAB guidance on security and privacy within IETF specs.

-18 Revised based on AD review.

-19 Revised after additional AD review request, and request to restructure.

-20 Revised after directorate reviews and IETF LC comments.

Gen-ART:

- o While section 2 does include a discussion of traffic mis-ordering, it does not include a discussion of ECMP, and the dependence of ECMP on flow identification to avoid significant packet mis-ordering.:: ECMP added as example.
- o Section 5.1 of this document discusses the use of Hop-by-Hop IPv6 options. It seems that it should acknowledge and discuss the applicability of the sentence "New hop-by-hop options are not

recommended..." from section 4.8 of RFC 8200. I think a good argument can be made in this case as to why (based on the rest of the sentence from 8200) the recommendation does not apply to this proposal. The document should make the argument.:: Quoted RFC sentences directly to avoid interpreting them.

- o I found the discussion of header compression slightly confusing. Given that the TCP / UDP header is small even compared to the IP header, it is difficult to see why encrypting it would have a significant impact on header compression efficacy. :: Added a preface that explains that HC methods are most effective for bit-congestive links.
- o The wording in section 6.2 on adding header information to an IP packet has the drawback of seeming to imply that one could add (or remove) such information in the network, without adding an encapsulating header. That is not permitted by RFC 8200 (IPv6). It would be good to clarify the first paragraph. (The example, which talks about the sender putting in the information is, of course, fine.) :: Unintended - added a sentence of preface.

SECDIR:: Previous revisions were updated following Early Review comments.

OPSEC:: No additional changes were requested in the OPSEC review.

IETF LC:: Tom Herbert: Please refer to 8200 on EH :: addressed in response to Joel above. Michael Richardson, Fernando Gont, Tom Herbert: Continuation of discussion on domains where EH might be (or not) useful and the tussle on what information to reveal. Unclear yet what additional text should be changed within this ID.

- 21 Revised after IESG review:

Revision 21 includes revised text after comments from Zahed, Erik Kline, Rob Wilton, Eric Vyncke, Roman Danyliw, and Benjamin Kaduk.

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Transport Options for UDP
draft-ietf-tsvwg-udp-options-18.txt

Abstract

Transport protocols are extended through the use of transport header options. This document extends UDP by indicating the location, syntax, and semantics for UDP transport layer options.

Status of this Memo

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

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1. Introduction

Transport protocols use options as a way to extend their capabilities. TCP [RFC793], SCTP [RFC4960], and DCCP [RFC4340] include space for these options but UDP [RFC768] currently does not. This document defines an extension to UDP that provides space for transport options including their generic syntax and semantics for their use in UDP's stateless, unreliable message protocol.

2. Conventions used in this document

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

In this document, the characters ">>" preceding an indented line(s) indicates a statement using the key words listed above. This convention aids reviewers in quickly identifying or finding the portions of this RFC covered by these key words.

3. Terminology

The following terminology is used in this document:

- o IP datagram [RFC791][RFC8200] - an IP packet, composed of the IP header and an IP payload area
- o User datagram - a UDP packet, composed of a UDP header and UDP payload; as discussed herein, that payload need not extend to the end of the IP datagram
- o UDP packet - the more contemporary term used herein to refer to a user datagram [RFC768]
- o Surplus area - the area of an IP payload that follows a UDP packet; this area is used for UDP options in this document

- o UDP fragment - one or more components of a UDP packet and its UDP options that enables transmission as IP payloads larger than permitted by IP datagram maximum sizes; note that each UDP fragment is itself transmitted as a UDP packet with its own options
- o (UDP) User data - the user data field of a UDP packet [RFC768]
- o UDP Length - the length field of a UDP header [RFC768]
- o Must-support options - UDP options that all implementations are required to support. Their use in individual UDP packets is optional.

4. Background

Many protocols include a default, invariant header and an area for header options that varies from packet to packet. These options enable the protocol to be extended for use in particular environments or in ways unforeseen by the original designers. Examples include TCP's Maximum Segment Size, Window Scale, Timestamp, and Authentication Options [RFC793][RFC5925][RFC7323].

Header options are used both in stateful (connection-oriented, e.g., TCP [RFC793], SCTP [RFC4960], DCCP [RFC4340]) and stateless (connectionless, e.g., IPv4 [RFC791], IPv6 [RFC8200]) protocols. In stateful protocols they can help extend the way in which state is managed. In stateless protocols their effect is often limited to individual packets, but they can have an aggregate effect on a sequence of packets as well.

UDP is one of the most popular protocols that lacks space for header options [RFC768]. The UDP header was intended to be a minimal addition to IP, providing only ports and a checksum for error detection. This document extends UDP to provide a trailer area for such options, located after the UDP user data.

UDP options are possible because UDP includes its own length field, separate from that of the IP header. Other transport protocols infer transport payload length from the IP datagram length (TCP, DCCP, SCTP). There are a number of reasons why Internet historians suggest that UDP includes this field, e.g., to support multiple UDP packets within the same IP datagram or to indicate the length of the UDP user data as distinct from zero padding required for systems that require writes that are not byte-aligned. These suggestions are not consistent with earlier versions of UDP or with concurrent design of multi-segment multiplexing protocols, however, so the real reason

remains unknown. Regardless, this field presents an opportunity to differentiate the UDP user data from the implied transport payload length, which this document leverages to support a trailer options field.

There are other ways to include additional header fields or options in protocols that otherwise are not extensible. In particular, in-band encoding can be used to differentiate transport payload from additional fields, such as was proposed in [Hi15]. This approach can cause complications for interactions with legacy devices, and is thus not considered further in this document.

IPv6 Teredo [RFC6081] uses values of the UDP Length that are larger than the IP payload as an additional type of signal, as noted in Section 20. UDP options uses a value smaller than the IP payload to enable backwards compatibility with existing UDP implementations, i.e., to deliver the UDP Length of UDP user data to the application and silently ignore the additional surplus area data. Using a value larger than the IP payload could either be considered malformed (and ought to be silently dropped by UDP processing) or could cause buffer overruns, and so is not considered silently and safely backward compatible.

5. The UDP Option Area

The UDP transport header includes demultiplexing and service identification (port numbers), an error detection checksum, and a field that indicates the UDP datagram length (including UDP header). The UDP Length field is typically redundant with the size of the maximum space available as a transport protocol payload, as determined by the IP header (see detail in Section 16). The UDP Option area is created when the UDP Length indicates a smaller transport payload than implied by the IP header.

For IPv4, IP Total Length field indicates the total IP datagram length (including IP header) and the size of the IP options is indicated in the IP header (in 4-byte words) as the "Internet Header Length" (IHL), as shown in Figure 1 [RFC791]. As a result, the typical (and largest valid) value for UDP Length is:

$$\text{UDP_Length} = \text{IPv4_Total_Length} - \text{IPv4_IHL} * 4$$

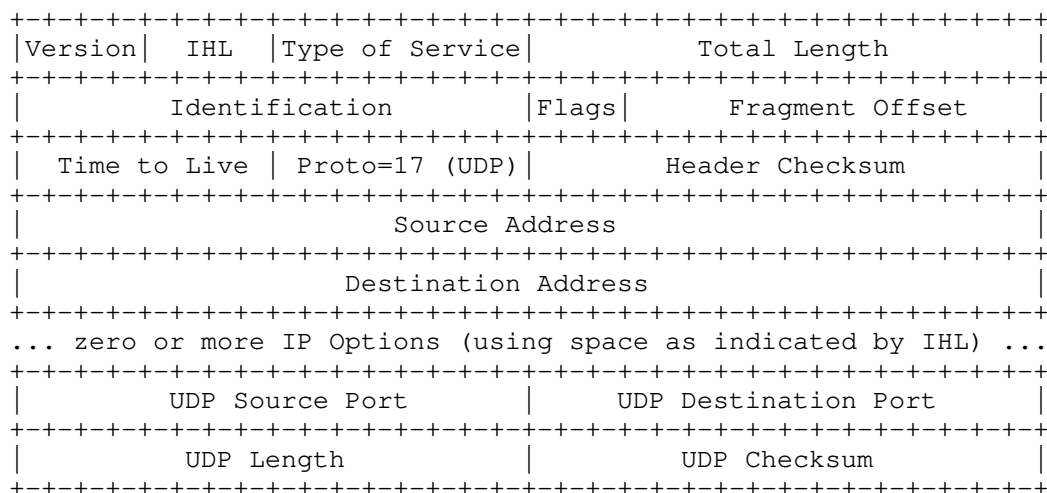


Figure 1 IPv4 datagram with UDP header

For IPv6, the IP Payload Length field indicates the transport payload after the base IPv6 header, which includes the IPv6 extension headers and space available for the transport protocol, as shown in Figure 2 [RFC8200]. Note that the Next HDR field in IPv6 might not indicate UDP (i.e., 17), e.g., when intervening IP extension headers are present. For IPv6, the lengths of any additional IP extensions are indicated within each extension [RFC8200], so the typical (and largest valid) value for UDP Length is:

$$\text{UDP_Length} = \text{IPv6_Payload_Length} - \text{sum}(\text{extension header lengths})$$

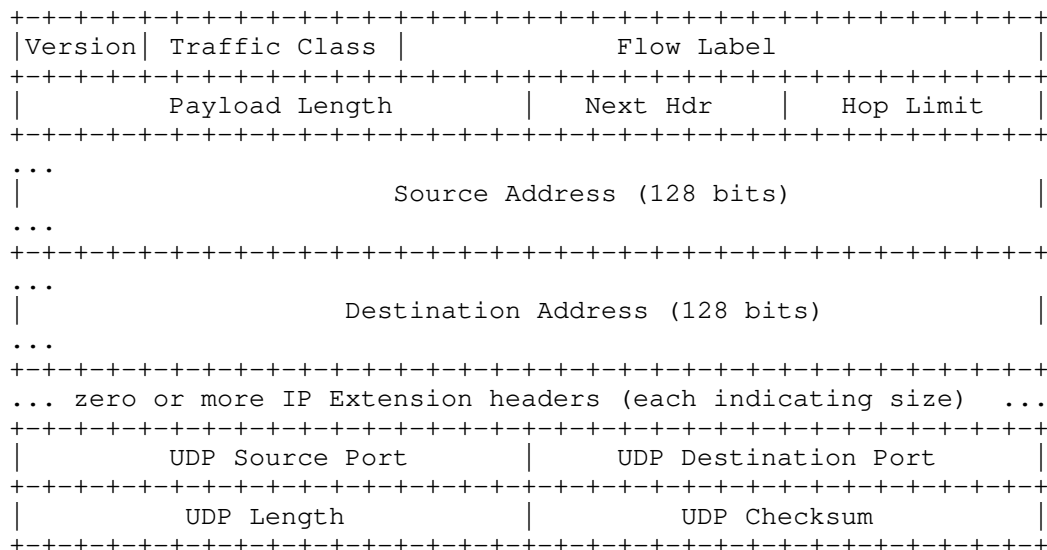


Figure 2 IPv6 datagram with UDP header

In both cases, the space available for the UDP packet is indicated by IP, either directly in the base header (for IPv4) or by adding information in the extensions (for IPv6). In either case, this document will refer to this available space as the "IP transport payload".

As a result of this redundancy, there is an opportunity to use the UDP Length field as a way to break up the IP transport payload into two areas - that intended as UDP user data and an additional "surplus area" (as shown in Figure 3).

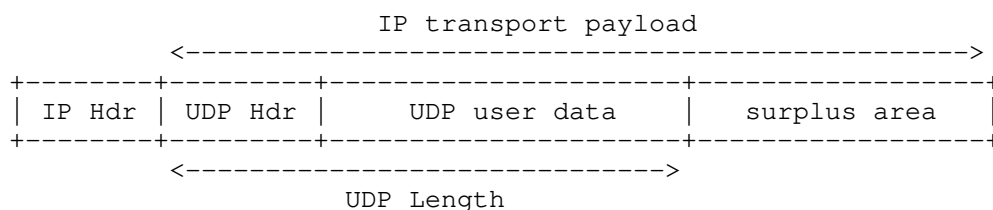


Figure 3 IP transport payload vs. UDP Length

In most cases, the IP transport payload and UDP Length point to the same location, indicating that there is no surplus area. This is not

a requirement of UDP [RFC768] (discussed further in Section 16). This document uses the surplus area for UDP options.

The surplus area can commence at any valid byte offset, i.e., it need not be 16-bit or 32-bit aligned. In effect, this document redefines the UDP "Length" field as a "trailer options offset".

6. The UDP Surplus Area Structure

UDP options use the entire surplus area, i.e., the contents of the IP payload after the last byte of the UDP payload. They commence with a 2-byte Option Checksum (OCS) field aligned to the first 2-byte boundary (relative to the start of the IP datagram) of that area, using zeroes for alignment. The UDP option area can be used with any UDP payload length (including zero), as long as there remains enough space for the aligned OCS and the options used.

>> UDP options MAY begin at any UDP length offset.

>> Option area bytes used for alignment before the OCS MUST be zero.

The OCS contains an optional ones-complement sum that detects errors in the surplus area, which is not otherwise covered by the UDP checksum, as detailed in Section 7.

The remainder of the surplus area consists of options defined using a TLV (type, length, and optional value) syntax similar to that of TCP [RFC793], as detailed in Section 8. These options continue until the end of the surplus area or can end earlier using the EOL (end of list) option, followed by zeroes.

7. The Option Checksum (OCS)

The Option Checksum (OCS) option is conventional Internet checksum [RFC791] that detects errors in the surplus area. The OCS option contains a 16-bit checksum that is aligned to the first 2-byte boundary, preceded by zeroes for padding (if needed), as shown in Figure 4.

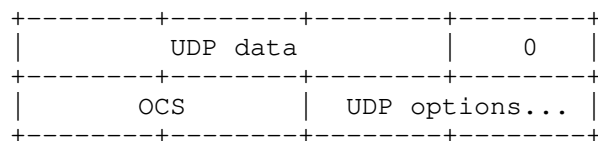


Figure 4 UDP OCS format, here using one zero for alignment

The OCS consists of a 16-bit Internet checksum [RFC1071], computed over the surplus area and including the length of the surplus area as an unsigned 16-bit value. The OCS protects the surplus area from errors in a similar way that the UDP checksum protects the UDP user data (when not zero).

The primary purpose of the OCS is to detect non-standard (i.e., non-option) uses of that area and accidental errors. It is not intended to detect attacks, as discussed further in Section 22.

The design enables traversal of errant middleboxes that incorrectly compute the UDP checksum over the entire IP payload [Fal8], rather than only the UDP header and UDP payload (as indicated by the UDP header length). Because the OCS is computed over the surplus area and its length and then inverted, OCS effectively negates the effect that incorrectly including the surplus has on the UDP checksum. As a result, when OCS is non-zero, the UDP checksum is the same in either case.

>> OCS MUST be non-zero when the UDP checksum is non-zero.

>> When the UDP checksum is zero, the OCS MAY be unused, and is then indicated by a zero OCS value.

Like the UDP checksum, the OCS is optional under certain circumstances and contains zero when not used. UDP checksums can be zero for IPv4 [RFC791] and for IPv6 [RFC8200] when UDP payload already covered by another checksum, as might occur for tunnels [RFC6935]. The same exceptions apply to the OCS when used to detect bit errors; an additional exception occurs for its use in the UDP datagram prior to fragmentation or after reassembly (see Section 9.4).

The OCS covers the surplus area as formatted for transmission and is processed immediately upon reception.

>> If the OCS fails, all options MUST be ignored and the surplus area silently discarded.

>> UDP user data that is validated by a correct UDP checksum MUST be delivered to the application layer, even if the OCS fails, unless the endpoints have negotiated otherwise for this UDP packet's socket pair.

When not used (i.e., containing zero), the OCS is assumed to be "correct" for the purpose of accepting UDP datagrams at a receiver (see Section 12).

8. UDP Options

UDP options are typically a minimum of two bytes in length as shown in Figure 5, excepting only the one byte options "No Operation" (NOP) and "End of Options List" (EOL) described below.

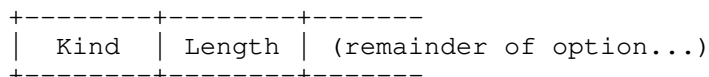


Figure 5 UDP option default format

The Kind field is always one byte. The Length field is one byte for all lengths below 255 (including the Kind and Length bytes). A Length of 255 indicates use of the UDP option extended format shown in Figure 6. The Extended Length field is a 16-bit field in network standard byte order.

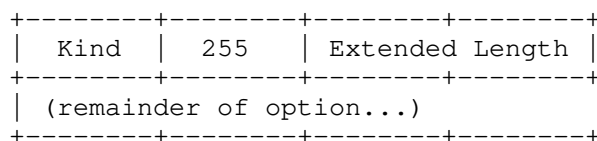


Figure 6 UDP option extended format

>> The UDP length MUST be at least as large as the UDP header (8) and no larger than the IP transport payload. Datagrams with length values outside this range MUST be silently dropped as invalid and logged where rate-limiting permits.

>> Option Lengths (or Extended Lengths, where applicable) smaller than the minimum for the corresponding Kind MUST be treated as an error. Such errors call into question the remainder of the surplus area and thus MUST result in all UDP options being silently discarded.

>> Any UDP option other than EOL and NOP MAY use either the default or extended option formats.

>> Any UDP option whose length is larger than 254 MUST use the UDP option extended format shown in Figure 6.

>> For compactness, UDP options SHOULD use the smallest option format possible.

>> UDP options MUST be interpreted in the order in which they occur in the surplus area.

The following UDP options are currently defined:

Kind	Length	Meaning

0*	-	End of Options List (EOL)
1*	-	No operation (NOP)
2*	6	Alternate payload checksum (APC)
3*	10/12	Fragmentation (FRAG)
4*	4	Maximum datagram size (MDS)
5*	4	Maximum reassembled datagram size (MRDS)
6*	6	Request (REQ)
7*	6	Response (RES)
8	10	Timestamps (TIME)
9	(varies)	Authentication (AUTH)
10-126	(varies)	UNASSIGNED (assignable by IANA)
127	(varies)	RFC 3692-style experiments (EXP)
128-191		RESERVED
192	(varies)	Encryption (UENC)
193-253		UNASSIGNED-UNSAFE (assignable by IANA)
254	(varies)	RFC 3692-style experiments (UEXP)
255		RESERVED

Options indicated by Kind values in the range 0..127 are known as SAFE options because they do not alter the UDP data payload and thus do not interfere with use of that data by legacy endpoints. Options indicated by Kind values in the range 192..254 are known as UNSAFE options because they do alter the UDP data payload and thus would interfere with legacy endpoints. UNSAFE option nicknames are expected to begin with "U", which should be avoided for safe option nicknames (see Section 23). Kind values 128-191 and 255 are RESERVED and not otherwise defined at this time.

>> RESERVED Kind values MUST NOT be assumed to be either SAFE nor UNSAFE until defined.

Although the FRAG option modifies the original UDP payload contents (i.e., is UNSAFE with respect to the original UDP payload), it is used only in subsequent fragments with zero UDP payloads, thus is SAFE in actual use, as discussed further in Section 9.4.

These options are defined in the following subsections. Options 0 and 1 use the same values as for TCP.

>> An endpoint supporting UDP options MUST support those marked with a "*" above: EOL, NOP, APC, FRAG, MDS, MRDS, REQ, and RES. This includes both recognizing and being able to generate these options if configured to do so. These are called "must-support" options.

>> An endpoint supporting UDP options MUST treat unsupported options in the UNSAFE range as terminating all option processing.

>> All other SAFE options (without a "*") MAY be implemented, and their use SHOULD be determined either out-of-band or negotiated, notably if needed to detect when options are silently ignored by legacy receivers.

>> Receivers supporting UDP options MUST silently ignore unknown SAFE options (i.e., in the same way a legacy receiver would). That includes options whose length does not indicate the specified value(s), as long as the length is not inherently invalid (i.e., smaller than 2 for the default and 4 for the extended formats).

>> UNSAFE options are used only in with the FRAG option, in a manner that prevents them from being silently ignored but passing the UDP payload to the user when not supported. This ensures their safe use in environments that might include legacy receivers (See Section 10).

>> Receivers supporting UDP options MUST silently drop all UDP options in a datagram containing an UNSAFE option when any UNSAFE option it contains is unknown. See Section 10 for further discussion of UNSAFE options.

>> Except for NOP, EXP, and UEXP, each option SHOULD NOT occur more than once in a single UDP datagram. If an option other than these occurs more than once, a receiver MUST interpret only the first instance of that option and MUST ignore all others.

>> EXP and UEXP MAY occur more than once, but SHOULD NOT occur more than once using the same ExID (see Sections 9.10 and 10.2).

>> Only the OCS and the AUTH and UENC options depend on the contents of the surplus area. AUTH and UENC are never used together, as UENC would serve both purposes. AUTH and UENC are always computed as if their hash and the OCS are zero; the OCS is always computed as if its contents are zero and after the AUTH or UENC hash has been computed. Future options MUST NOT be defined as having a value dependent on the contents of the surplus area. Otherwise, interactions between those values, the OCS, and the AUTH and UENC options could be unpredictable.

Receivers cannot generally treat unexpected option lengths as invalid, as this would unnecessarily limit future revision of options (e.g., defining a new APC that is defined by having a different length). The exception is only for lengths that imply a physical impossibility, e.g., smaller than two for conventional options and four for extended length options. Impossible lengths should indicate a malformed surplus area and all options silently discarded. Lengths other than those expected should result in safe options being ignored and skipped over, as with any other unknown safe option.

>> Option lengths MUST NOT exceed the IP length of the overall IP datagram. If this occurs, the options MUST be treated as malformed and all options dropped, and the event MAY be logged for diagnostics (logging SHOULD be rate limited).

>> "Must-support" options other than NOP and EOL MUST come before other options.

The requirement that must-support options come before others is intended to allow for endpoints to implement DOS protection, as discussed further in Section 22.

9. Safe UDP Options

Safe UDP options can be silently ignored by legacy receivers without affecting the meaning of the UDP user data. They stand in contrast to Unsafe options, which modify UDP user data in ways that render it unusable by legacy receivers (Section 10). The following subsections describe safe options defined in this document.

9.1. End of Options List (EOL)

The End of Options List (EOL, Kind=0) option indicates that there are no more options. It is used to indicate the end of the list of options without needing to use NOP options (see the following section) as padding to fill all available option space.

```
+-----+
| Kind=0 |
+-----+
```

Figure 7 UDP EOL option format

>> When the UDP options do not consume the entire surplus area, the last non-NOP option MUST be EOL.

>> NOPs SHOULD NOT be used as padding before the EOL option. As a one byte option, it need not be otherwise aligned.

>> All bytes in the surplus area after EOL MUST be set to zero on transmit.

>> Bytes after EOL in the surplus area MAY be checked as being zero on receipt but MUST be treated as zero regardless of their content and are not passed to the user (e.g., as part of the surplus area).

Requiring the post-option surplus area to be zero prevents side-channel uses of this area, requiring instead that all use of the surplus area be UDP options supported by both endpoints. It is useful to allow this area to be used for zero padding to increase the UDP datagram length without affecting the UDP user data length, e.g., for UDP DPLPMTUD (Section 4.1 of [Fa22]).

9.2. No Operation (NOP)

The No Operation (NOP, Kind=1) option is a one-byte placeholder, intended to be used as padding, e.g., to align multi-byte options along 16-bit, 32-bit, or 64-bit boundaries.

```
+-----+  
| Kind=1 |  
+-----+
```

Figure 8 UDP NOP option format

>> UDP packets SHOULD NOT use more than seven consecutive NOPs, i.e., to support alignment up to 8-byte boundaries. UDP packets SHOULD NOT use NOPs at the end of the options area as a substitute for EOL followed by zero-fill. NOPs are intended to assist with alignment, not as other padding or fill.

This issue is discussed further in Section 22.

9.3. Alternate Payload Checksum (APC)

The Alternate Payload Checksum (APC, Kind=2) option provides a stronger alternative to the checksum in the UDP header, using a 32-bit CRC of the conventional UDP user data payload only (excluding the IP pseudoheader, UDP header, and surplus area). It is an "alternate" to the UDP checksum that covers the user data - not to the OCS (the latter covers the surplus area only). Unlike the UDP checksum, APC does not include the IP pseudoheader or UDP header, thus it does not need to be updated by NATs when IP addresses or UDP

ports are rewritten. Its purpose is to detect user data errors that the UDP checksum, when used, might not detect.

A CRC32c has been chosen because of its ubiquity and use in other Internet protocols, including iSCSI and SCTP. The option contains the CRC32c in network standard byte order, as described in [RFC3385].

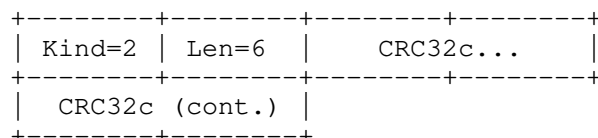


Figure 9 UDP APC option format

When present, the APC always contains a valid CRC checksum. There are no reserved values, including the value of zero. If the CRC is zero, this must indicate a valid checksum (i.e., it does not indicate that the APC is not used; instead, the option would simply not be included if that were the desired effect).

APC does not protect the UDP pseudoheader; only the current UDP checksum provides that protection (when used). APC cannot provide that protection because it would need to be updated whenever the UDP pseudoheader changed, e.g., during NAT address and port translation; because this is not the case, APC does not cover the pseudoheader.

>> UDP packets with incorrect APC checksums MUST be passed to the application by default, e.g., with a flag indicating APC failure.

Like all safe UDP options, APC needs to be silently ignored when failing by default, unless the receiver has been configured to do otherwise. Although all UDP option-aware endpoints support APC (being in the required set), this silently-ignored behavior ensures that option-aware receivers operate the same as legacy receivers unless overridden.

>> UDP packets with unrecognized APC lengths MUST be receive the same treatment as UDP packets with incorrect APC checksums.

Ensuring that unrecognized APC lengths are treated as incorrect checksums enables future variants of APC to be treated as APC-like.

9.4. Fragmentation (FRAG)

The Fragmentation (FRAG, Kind=3) option supports UDP fragmentation and reassembly, which can be used to transfer UDP messages larger than limited by the IP receive MTU (EMTU_R [RFC1122]). FRAG includes a copy of the same UDP transport ports in each fragment, enabling them to traverse Network Address (and port) Translation (NAT) devices, in contrast to the behavior of IP fragments. FRAG is typically used with the UDP MDS and MRDS options to enable more efficient use of large messages, both at the UDP and IP layers. FRAG is designed similar to the IPv6 Fragmentation Header [RFC8200], except that the UDP variant uses a 16-bit Offset measured in bytes, rather than IPv6's 13-bit Fragment Offset measured in 8-byte units. This UDP variant avoids creating reserved fields.

>> When FRAG is present, it SHOULD come as early as possible in the UDP options list.

>> When FRAG is present, the UDP user data MUST be empty. If the user data is not empty, all UDP options MUST be silently ignored and the user data received sent to the user.

Legacy receivers interpret FRAG messages as zero-length user data UDP packets (i.e., UDP Length field is 8, the length of just the UDP header), which would not affect the receiver unless the presence of the UDP packet itself were a signal (see Section 5 of [RFC8085]). In this manner, the FRAG option also helps hide UNSAFE options so they can be used more safely in the presence of legacy receivers.

The FRAG option has two formats; non-terminal fragments use the shorter variant (Figure 10) and terminal fragments use the longer (Figure 11). The latter includes stand-alone fragments, i.e., when data is contained in the FRAG option but reassembly is not required.

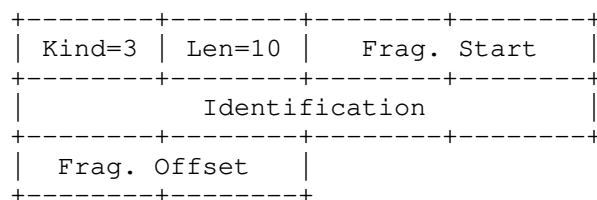


Figure 10 UDP non-terminal FRAG option format

In the non-terminal FRAG option format, Frag. Start indicates the location of the beginning of the fragment data, measured from the beginning of the UDP header of the fragment. The fragment data

follows the remainder of the UDP options and continues to the end of the IP datagram (i.e., the end of the surplus area). Those options are applied to this UDP fragment. Non-terminal fragments never have options after the fragment.

The Frag. Offset field indicates the location of this fragment relative to the original UDP datagram (prior to fragmentation), measured from the start of the original UDP datagram's UDP header.

The FRAG option does not need a "more fragments" bit because it provides the same indication by using the longer, 12-byte variant, as shown in Figure 11.

>> The FRAG option MAY be used on a single fragment, in which case the Frag. Offset would be zero and the option would have the 12-byte format.

>> Endpoints supporting UDP options MUST be capable of fragmenting and reassembling at least 2 fragments, for a total of at least 3,000 bytes (see MRDS in Section 9.6).

Use of the single fragment variant can be helpful in supporting use of UNSAFE options without undesirable impact to receivers that do not support either UDP options or the specific UNSAFE options.

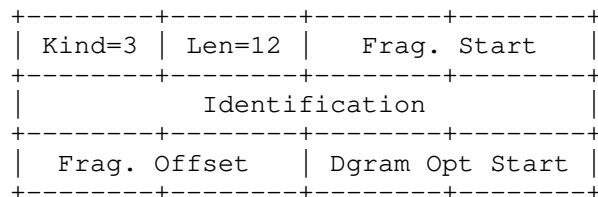


Figure 11 UDP terminal FRAG option format

The terminal FRAG option format adds a Datagram Option Start pointer, measured from the start of the original UDP datagram header, indicating the end of the reassembled data and the start of the surplus area after the original UDP datagram. In this variant, UDP options that apply to the reassembled datagram may occur after the terminal fragment data. UDP options that occur before the FRAG data are processed on the fragment; UDP options after the FRAG data are processed after reassembly, such that the reassembled data represents the original UDP user data. This allows either pre-reassembly or post-reassembly UDP option effects, such as using UENC on each fragment while also using TIME on the reassembled datagram for round-trip latency measurements.

>> During fragmentation, the UDP header checksum of each fragment remains constant and does not depend on the fragment data (which appears in the surplus area), because all fragments have a zero-length user data field.

The Fragment Offset is 16 bits and indicates the location of the UDP payload fragment in bytes from the beginning of the original unfragmented payload. The option Len field indicates whether there are more fragments (Len=10) or no more fragments (Len=12).

>> The Identification field is a 32-bit value that MUST be unique over the expected fragment reassembly timeout.

>> The Identification field SHOULD be generated in a manner similar to that of the IPv6 Fragment ID [RFC8200].

>> UDP fragments MUST NOT overlap.

Similar to IPv6 reassembly [RFC8200], if any of the fragments being reassembled overlap with any other fragments being reassembled for the same UDP packet, reassembly of that UDP packet must be abandoned and all the fragments that have been received for that UDP packet must be discarded, and no ICMP error messages should be sent.

It should be noted that fragments may be duplicated in the network. Instead of treating these exact duplicate fragments as overlapping fragments, an implementation may choose to detect this case and drop exact duplicate fragments while keeping the other fragments belonging to the same UDP packet.

UDP fragmentation relies on a fragment expiration timer, which can be preset or could use a value computed using the UDP Timestamp option.

>> The default UDP reassembly SHOULD be no more than 2 minutes.

>> UDP reassembly space SHOULD be limited to reduce the impact of DOS attacks on resource use.

>> UDP reassembly space limits SHOULD NOT be computed as a shared resource across multiple sockets, to avoid cross-socketpair DOS attacks.

>> Individual UDP fragments MUST NOT be forwarded to the user. The reassembled datagram is received only after complete reassembly, checksum validation, and continued processing of the remaining UDP options.

Any per-datagram UDP options, if used, follow the FRAG option in the final fragment and would be included in the reassembled UDP packet. Processing of those options would commence after reassembly. This is especially important for UNSAFE options, which are interpreted only after FRAG.

In general, UDP packets are fragmented as follows:

1. Create a UDP packet with data and UDP options, which we will call "D". Note that the UDP options treat the data area as UDP user data and thus must follow that data.

Process these UDP options before the rest of the fragmentation steps below. Note that the OCS value of the original packet SHOULD be zero if each fragment will have a non-zero OCS value (as will be the case if the UDP checksum is non-zero).

2. Identify the desired fragment size, which we will call "S". This value should take into account the path MTU (if known) and allow space for per-fragment options.
3. Fragment "D" into chunks of size no larger than "S"-10 each, with one final chunk no larger than "S"-12. Note that all the non-FRAG options in step #1 need not be limited to the terminal fragment, i.e., the Dgram Opt. Start pointer can indicate the start of the original surplus area anywhere in the reassembled data.
4. For each chunk of "D" in step #3, create a zero-data UDP packet followed by the word-aligned OCS, the FRAG option, and any additional UDP options, followed by the FRAG data chunk.

The last chunk includes the non-FRAG options noted in step #1 after the end of the FRAG data. These UDP options apply to the reassembled user data as a whole when received.

5. Process the pre-reassembly UDP options of each fragment.

Receivers reverse the above sequence. They process all received options in each fragment. When the FRAG option is encountered, the FRAG data is used in reassembly. After all fragments are received, the entire UDP packet is processed with any trailing UDP options applying to the reassembled user data.

9.5. Maximum Datagram Size (MDS)

The Maximum Datagram Size (MDS, Kind=4) option is a 16-bit hint of the largest unfragmented UDP packet that an endpoint believes can be

received. As with the TCP Maximum Segment Size (MSS) option [RFC793], the size indicated is the IP layer MTU decreased by the fixed IP and UDP headers only [RFC6691]. The space needed for IP and UDP options need to be adjusted by the sender when using the value indicated. The value transmitted is based on EMTU_R, the largest IP datagram that can be received (i.e., reassembled at the receiver) [RFC1122]. However, as with TCP, this value is only a hint at what the receiver believes; it does not indicate a known path MTU and thus MUST NOT be used to limit transmissions.

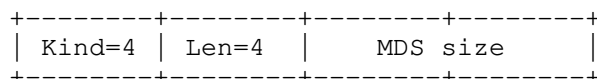


Figure 12 UDP MDS option format

The UDP MDS option MAY be used as a hint for path MTU discovery [RFC1191][RFC8201], but this may be difficult because of known issues with ICMP blocking [RFC2923] as well as UDP lacking automatic retransmission. It is more likely to be useful when coupled with IP source fragmentation or UDP fragmentation to limit the largest reassembled UDP message as indicated by MRDS (see Section 9.6), e.g., when EMTU_R is larger than the required minimums (576 for IPv4 [RFC791] and 1500 for IPv6 [RFC8200]). It can also be used with DPLPMTUD [RFC8899] to provide a hint to maximum DPLPMTU, though it MUST NOT prohibit transmission of larger UDP packets (or fragments) used as DPLPMTU probes.

9.6. Maximum Reassembled Datagram Size (MRDS)

The Maximum Reassembled Segment Size (MRDS, Kind=5) option is a 16-bit indicator of the largest reassembled UDP segment that can be received. MRDS is the UDP equivalent of IP's EMTU_R but the two are not related [RFC1122]. Using the FRAG option (Section 9.4), UDP packets can be transmitted as transport fragments, each in their own (presumably not fragmented) IP datagram and be reassembled at the UDP layer.

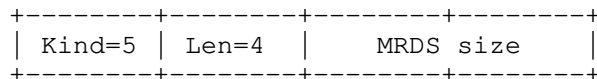


Figure 13 UDP MRDS option format

>> Endpoints supporting UDP options MUST support a local MRDS of at least 3,000 bytes.

9.7. Echo request (REQ) and echo response (RES)

The echo request (REQ, Kind=6) and echo response (RES, Kind=7) options provide a means for UDP options to be used to provide UDP packet-level acknowledgements. One such use is described as part of the UDP options variant of packetization layer path MTU discovery (PLPMTUD) [Fa22]. The options both have the format indicated in Figure 14, in which the token has no internal structure or meaning.

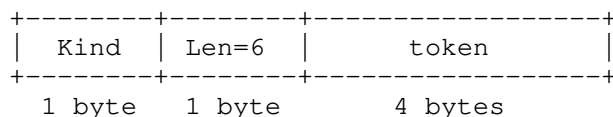


Figure 14 UDP REQ and RES options format

Each of these option kinds appears at most once in each UDP packet, as with other options. Note also that the FRAG option is not used when sending DPLPMTUD probes to determine a PLPMTU [Fa22].

9.8. Timestamps (TIME)

The Timestamp (TIME, Kind=8) option exchanges two four-byte unsigned timestamp fields. It serves a similar purpose to TCP's TS option [RFC7323], enabling UDP to estimate the round trip time (RTT) between hosts. For UDP, this RTT can be useful for establishing UDP fragment reassembly timeouts or transport-layer rate-limiting [RFC8085].

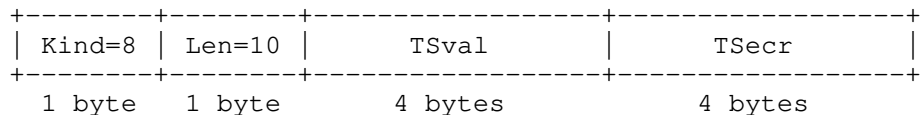


Figure 15 UDP TIME option format

TS Value (TSval) and TS Echo Reply (TSecr) are used in a similar manner to the TCP TS option [RFC7323]. On transmitted UDP packets using the option, TS Value is always set based on the local "time" value. Received TSval and TSecr values are provided to the application, which can pass the TSval value to be used as TSecr on UDP messages sent in response (i.e., to echo the received TSval). A received TSecr of zero indicates that the TSval was not echoed by the transmitter, i.e., from a previously received UDP packet.

>> TIME MAY use an RTT estimate based on nonzero Timestamp values as a hint for fragmentation reassembly, rate limiting, or other mechanisms that benefit from such an estimate.

>> an application MAY use TIME to compute this RTT estimate for further use by the user.

UDP timestamps are modeled after TCP timestamps and have similar expectations. In particular, they are expected to be:

- o Values are monotonic and non-decreasing except for anticipated number-space rollover events
- o Values should "increase" (allowing for rollover) according to a typical 'tick' time
- o A request is defined as TSval being non-zero and a reply is defined as TSecr being non-zero.
- o A receiver should always respond to a request with the highest TSval received (allowing for rollover), which is not necessarily the most recently received.

Rollover can be handled as a special case or more completely using sequence number extension [RFC9187], however zero values need to be avoided explicitly.

>> TIME values MUST NOT use zeros as valid time values, because they are used as indicators of requests and responses.

9.9. Authentication (AUTH)

The Authentication (AUTH, Kind=9) option is intended to allow UDP to provide a similar type of authentication as the TCP Authentication Option (TCP-AO) [RFC5925]. AUTH covers the UDP user data. AUTH supports NAT traversal in a similar manner as TCP-AO [RFC6978]. Figure 16 shows the UDP AUTH format, whose contents are identical to that of the TCP-AO option.

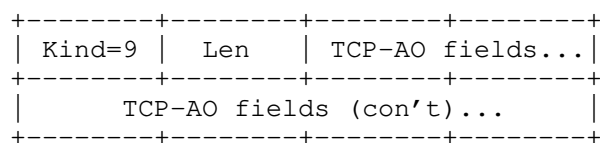


Figure 16 UDP AUTH option format

Like TCP-AO, AUTH is not negotiated in-band. Its use assumes both endpoints have populated Master Key Tuples (MKTs), used to exclude non-protected traffic.

TCP-AO generates unique traffic keys from a hash of TCP connection parameters. UDP lacks a three-way handshake to coordinate connection-specific values, such as TCP's Initial Sequence Numbers (ISNs) [RFC793], thus AUTH's Key Derivation Function (KDF) uses zeroes as the value for both ISNs. This means that the AUTH reuses keys when socket pairs are reused, unlike TCP-AO.

>> UDP packets with incorrect AUTH HMACs MUST be passed to the application by default, e.g., with a flag indicating AUTH failure.

Like all non-UNSAFE UDP options, AUTH needs to be silently ignored when failing. This silently-ignored behavior ensures that option-aware receivers operate the same as legacy receivers unless overridden.

In addition to the UDP user data (which is always included), AUTH can be configured to either include or exclude the surplus area, in a similar way as can TCP-AO can optionally exclude TCP options. When UDP options are covered, the OCS value and AUTH (and later, UENC) hash areas are zeroed before computing the AUTH hash. It is important to consider that options not yet defined might yield unpredictable results if not confirmed as supported, e.g., if they were to contain other hashes or checksums that depend on the surplus area contents. This is why such dependencies are not permitted except as defined for the OCS and the AUTH (and later, UENC) option.

Similar to TCP-AO-NAT, AUTH (and later, UENC) can be configured to support NAT traversal, excluding (by zeroing out) one or both of the UDP ports and corresponding IP addresses [RFC6978].

9.10. Experimental (EXP)

The Experimental option (EXP, Kind=127) is reserved for experiments [RFC3692]. Only one such value is reserved because experiments are expected to use an Experimental ID (ExIDs) to differentiate concurrent use for different purposes, using UDP ExIDs registered with IANA according to the approach developed for TCP experimental options [RFC6994].

Kind=127	Len	UDP ExID
(option contents, as defined)...		

Figure 17 UDP EXP option format

>> The length of the experimental option MUST be at least 4 to account for the Kind, Length, and the minimum 16-bit UDP ExID identifier (similar to TCP ExIDs [RFC6994]).

The UDP EXP option also includes an extended length format, where the option LEN is 255 followed by two bytes of extended length.

Kind=127	255	Extended Length
UDP ExID.	(option contents...)	

Figure 18 UDP EXP option format

Assigned UDP experimental IDs (ExIDs) assigned from a single registry managed by IANA (see Section 23). Assigned ExIDs can be used in either the EXP or UEXP options (see Section 10.2 for the latter).

10. UNSAFE Options

UNSAFE options are not safe to ignore and can be used unidirectionally or without soft-state confirmation of UDP option capability. They are always used only when the user data occurs inside a reassembled set of one or more UDP fragments, such that if UDP fragmentation is not supported, the enclosed UDP user data would be silently dropped anyway.

>> Applications using UNSAFE options SHOULD NOT also use zero-length UDP packets as signals, because they will arrive when UNSAFE options fail. Those that choose to allow such packets MUST account for such events.

>> UNSAFE options MUST be used only as part of UDP fragments, used either per-fragment or after reassembly.

>> Receivers supporting UDP options MUST silently drop the UDP user data of the reassembled datagram if any fragment or the entire

datagram includes an UNSAFE option whose UKind is not supported. Note that this still results in the receipt of a zero-length UDP datagram.

10.1. UNSAFE Encryption (UENC)

UNSAFE encryption (UENC, Kind=192) has the same format as AUTH (Section 9.9), except that it encrypts (modifies) the user data. It provides a similar encryption capability as TCP-AO-ENC, in a similar manner [Tol8]. Its fields, coverage, and processing are the same as for AUTH, except that UENC encrypts only the user data, although it can (optionally) depend on the surplus area (with certain fields zeroed, as per AUTH, e.g., providing authentication over the surplus area). Like AUTH, UENC can be configured to be compatible with NAT traversal.

10.2. UNSAFE Experimental (UEXP)

The UNSAFE Experimental option (UEXP, Kind=254) is reserved for experiments [RFC3692]. As with EXP, only one such UEXP value is reserved because experiments are expected to use an Experimental ID (ExIDs) to differentiate concurrent use for different purposes, using UDP ExIDs registered with IANA according to the approach developed for TCP experimental options [RFC6994].

Assigned ExIDs can be used with either the UEXP or EXP options.

11. Rules for designing new options

The UDP option Kind space allows for the definition of new options, however the currently defined options do not allow for arbitrary new options. The following is a summary of rules for new options and their rationales:

>> New options MUST NOT modify other option content.

>> New options MUST NOT depend on the content of other options.

>> UNSAFE options can both depend on and vary user data content because they are contained only inside UDP fragments and thus are processed only by UDP option capable receivers.

>> New options MUST NOT declare their order relative to other options, whether new or old.

>> At the sender, new options MUST NOT modify UDP packet content anywhere except within their option field, excepting only those

contained within the UNSAFE option; areas that need to remain unmodified include the IP header, IP options, the UDP user data, and the surplus area (i.e., other options).

>> Options MUST NOT be modified in transit. This includes those already defined as well as new options.

>> New options MUST NOT require or intend optionally for modification of any UDP options, including their new areas, in transit.

Note that only certain of the initially defined options violate these rules:

- o >> Only FRAG and UNSAFE options are permitted to modify the UDP body.

The following recommendation helps enable efficient zero-copy processing:

- o >> FRAG SHOULD be the first option, when present.

12. Option inclusion and processing

The following rules apply to option inclusion by senders and processing by receivers.

>> Senders MAY add any option, as configured by the API.

>> All "must-support" options MUST be processed by receivers, if present (presuming UDP options are supported at that receiver).

>> Non-"must-support" options MAY be ignored by receivers, if present, e.g., based on API settings.

>> All options MUST be processed by receivers in the order encountered in the options area.

>> All options except UNSAFE options MUST result in the UDP user data being passed to the application layer, regardless of whether all options are processed, supported, or succeed.

The basic premise is that, for options-aware endpoints, the sender decides what options to add and the receiver decides what options to handle. Simply adding an option does not force work upon a receiver, with the exception of the "must-support" options.

Upon receipt, the receiver checks various properties of the UDP packet and its options to decide whether to accept or drop the UDP packet and whether to accept or ignore some its options as follows (in order):

```
if the UDP checksum fails then
    silently drop the entire UDP packet (per RFC1122)
if the UDP checksum passes then
    if OCS != 0 and fails or is zero when UDP CS != 0 then
        deliver the UDP user data but ignore other options
        (this is required to emulate legacy behavior)
    if OCS is nonzero and passes or is zero then
        deliver the UDP user data after parsing
        and processing the rest of the options,
        regardless of whether each is supported or succeeds
        (again, this is required to emulate legacy behavior)
```

The design of the UNSAFE options as used only inside the FRAG area ensures that the resulting UDP data will be silently dropped in both legacy and options-aware receivers. Again, note that this still results in the delivery of a zero-length UDP packet.

Options-aware receivers can drop UDP packets with option processing errors via either an override of the default UDP processing or at the application layer.

I.e., all options are treated the same, in that the transmitter can add it as desired and the receiver has the option to require it or not. Only if it is required (e.g., by API configuration) should the receiver require it being present and correct.

I.e., for all options:

- o if the option is not required by the receiver, then UDP packets missing the option are accepted.
- o if the option is required (e.g., by override of the default behavior at the receiver) and missing or incorrectly formed, silently drop the UDP packet.
- o if the UDP packet is accepted (either because the option is not required or because it was required and correct), then pass the option with the UDP packet via the API.

Any options whose length exceeds that of the UDP packet (i.e., intending to use data that would have been beyond the surplus area) should be silently ignored (again to model legacy behavior).

13. UDP API Extensions

UDP currently specifies an application programmer interface (API), summarized as follows (with Unix-style command as an example) [RFC768]:

- o Method to create new receive ports
 - o E.g., `bind(handle, recvaddr(optional), recvport)`
- o Receive, which returns data octets, source port, and source address
 - o E.g., `recvfrom(handle, srcaddr, srcport, data)`
- o Send, which specifies data, source and destination addresses, and source and destination ports
 - o E.g., `sendto(handle, destaddr, destport, data)`

This API is extended to support options as follows:

- o Extend the method to create receive ports to include per-packet and per-fragment receive options that are required as indicated by the application. Datagrams not containing these required options MUST be silently dropped and MAY be logged. This includes a minimum datagram length, such that the options list ends in EOL and additional space is zero-filled as needed.
- o WG QUESTION: DO WE ALSO WANT A MIN FRAG SIZE? OR MAX?
- o Extend the receive function to indicate the per-packet options and their parameters as received with the corresponding received datagram. Note that per-fragment options are handled within the processing of each fragment.
- o WG QUESTION: SHOULD WE ACCUMULATE THOSE OPTIONS? OR DISCARD THEM?
- o Extend the send function to indicate the options to be added to the corresponding sent datagram. This includes indicating which options apply to individual fragments vs. which apply to the UDP packet prior to fragmentation, if fragmentation is enabled.

Examples of API instances for Linux and FreeBSD are provided in Appendix A, to encourage uniform cross-platform implementations.

14. UDP Options are for Transport, Not Transit

UDP options are indicated in the surplus area of the IP payload that is not used by UDP. That area is really part of the IP payload, not the UDP payload, and as such, it might be tempting to consider whether this is a generally useful approach to extending IP.

Unfortunately, the surplus area exists only for transports that include their own transport layer payload length indicator. TCP and SCTP include header length fields that already provide space for transport options by indicating the total length of the header area, such that the entire remaining area indicated in the network layer (IP) is transport payload. UDP-Lite already uses the UDP Length field to indicate the boundary between data covered by the transport checksum and data not covered, and so there is no remaining area where the length of the UDP-Lite payload as a whole can be indicated [RFC3828].

UDP options are intended for use only by the transport endpoints. They are no more (or less) appropriate to be modified in-transit than any other portion of the transport datagram.

UDP options are transport options. Generally, transport headers, options, and data are not intended to be modified in-transit. UDP options are no exception and here are specified as "MUST NOT" be altered in transit. However, the UDP option mechanism provides no specific protection against in-transit modification of the UDP header, UDP payload, or surplus area, except as provided by the OCS or the options selected (e.g., AUTH, or UENC).

15. UDP options vs. UDP-Lite

UDP-Lite provides partial checksum coverage, so that UDP packets with errors in some locations can be delivered to the user [RFC3828]. It uses a different transport protocol number (136) than UDP (17) to interpret the UDP Length field as the prefix covered by the UDP checksum.

UDP (protocol 17) already defines the UDP Length field as the limit of the UDP checksum, but by default also limits the data provided to the application as that which precedes the UDP Length. A goal of UDP-Lite is to deliver data beyond UDP Length as a default, which is why a separate transport protocol number was required.

UDP options do not use or need a separate transport protocol number because the data beyond the UDP Length offset (surplus data) is not provided to the application by default. That data is interpreted exclusively within the UDP transport layer.

UDP-Lite cannot support UDP options, either as proposed here or in any other form, because the entire payload of the UDP packet is already defined as user data and there is no additional field in which to indicate a surplus area for options. The UDP Length field in UDP-Lite is already used to indicate the boundary between user data covered by the checksum and user data not covered.

16. Interactions with Legacy Devices

It has always been permissible for the UDP Length to be inconsistent with the IP transport payload length [RFC768]. Such inconsistency has been utilized in UDP-Lite using a different transport number. There are no known systems that use this inconsistency for UDP [RFC3828]. It is possible that such use might interact with UDP options, i.e., where legacy systems might generate UDP datagrams that appear to have UDP options. The OCS provides protection against such events and is stronger than a static "magic number".

UDP options have been tested as interoperable with Linux, macOS, and Windows Cygwin, and worked through NAT devices. These systems successfully delivered only the user data indicated by the UDP Length field and silently discarded the surplus area.

One reported embedded device passes the entire IP datagram to the UDP application layer. Although this feature could enable application-layer UDP option processing, it would require that conventional UDP user applications examine only the UDP user data. This feature is also inconsistent with the UDP application interface [RFC768] [RFC1122].

It has been reported that Alcatel-Lucent's "Brick" Intrusion Detection System has a default configuration that interprets inconsistencies between UDP Length and IP Length as an attack to be reported. Note that other firewall systems, e.g., CheckPoint, use a default "relaxed UDP length verification" to avoid falsely interpreting this inconsistency as an attack.

17. Options in a Stateless, Unreliable Transport Protocol

There are two ways to interpret options for a stateless, unreliable protocol -- an option is either local to the message or intended to

affect a stream of messages in a soft-state manner. Either interpretation is valid for defined UDP options.

It is impossible to know in advance whether an endpoint supports a UDP option.

>> All UDP options other than UNSAFE ones MUST be ignored if not supported or upon failure (e.g., APC).

>> All UDP options that fail MUST result in the UDP data still being sent to the application layer by default, to ensure equivalence with legacy devices.

>> UDP options that rely on soft-state exchange MUST allow for message reordering and loss.

The above requirements prevent using any option that cannot be safely ignored unless it is hidden inside the FRAG area (i.e., UNSAFE options). Legacy systems also always need to be able to interpret the transport fragments as individual UDP packets.

18. UDP Option State Caching

Some TCP connection parameters, stored in the TCP Control Block, can be usefully shared either among concurrent connections or between connections in sequence, known as TCP Sharing [RFC9040]. Although UDP is stateless, some of the options proposed herein may have similar benefit in being shared or cached. We call this UCB Sharing, or UDP Control Block Sharing, by analogy. Just as TCB sharing is not a standard because it is consistent with existing TCP specifications, UCB sharing would be consistent with existing UDP specifications, including this one. Both are implementation issues that are outside the scope of their respective specifications, and so UCB sharing is outside the scope of this document.

19. Updates to RFC 768

This document updates RFC 768 as follows:

- o This document defines the meaning of the IP payload area beyond the UDP length but within the IP length as the surplus area used herein for UDP options.
- o This document extends the UDP API to support the use of UDP options.

20. Interactions with other RFCs (and drafts)

This document clarifies the interaction between UDP Length and IP length that is not explicitly constrained in either UDP or the host requirements [RFC768] [RFC1122].

Teredo extensions (TE) define use of a similar difference between these lengths for trailers [RFC6081]. TE defines the UDP length pointing beyond (larger) than the location indicated by the IP length rather than shorter (as used herein):

"..the IPv6 packet length (i.e., the Payload Length value in the IPv6 header plus the IPv6 header size) is less than or equal to the UDP payload length (i.e., the Length value in the UDP header minus the UDP header size)"

As a result, UDP options are not compatible with TE, but that is also why this document does not update TE. Additionally, it is not at all clear how TE operates, as it requires network processing of the UDP length field to understand the total message including TE trailers.

TE updates Teredo NAT traversal [RFC4380]. The NAT traversal document defined "consistency" of UDP length and IP length as:

"An IPv6 packet is deemed valid if it conforms to [RFC2460]: the protocol identifier should indicate an IPv6 packet and the payload length should be consistent with the length of the UDP datagram in which the packet is encapsulated."

IPv6 is clear on the meaning of this consistency, in which the pseudoheader used for UDP checksums is based on the UDP length, not inferred from the IP length, using the same text in the current specification [RFC8200]:

"The Upper-Layer Packet Length in the pseudo-header is the length of the upper-layer header and data (e.g., TCP header plus TCP data). Some upper-layer protocols carry their own length information (e.g., the Length field in the UDP header); for such protocols, that is the length used in the pseudo-header."

This document is consistent the UDP profile for Robust Header Compression (ROHC) [RFC3095], noted here:

"The Length field of the UDP header MUST match the Length field(s) of the preceding subheaders, i.e., there must not

be any padding after the UDP payload that is covered by the IP Length."

ROHC compresses UDP headers only when this match succeeds. It does not prohibit UDP headers where the match fails; in those cases, ROHC default rules (Section 5.10) would cause the UDP header to remain uncompressed. Upon receipt of a compressed UDP header, Section A.1.3 of that document indicates that the UDP length is "INFERRED"; in uncompressed packets, it would simply be explicitly provided.

This issue of handling UDP header compression is more explicitly described in more recent specifications, e.g., Sec. 10.10 of Static Context Header Compression [RFC8724].

21. Multicast Considerations

UDP options are primarily intended for unicast use. Using these options over multicast IP requires careful consideration, e.g., to ensure that the options used are safe for different endpoints to interpret differently (e.g., either to support or silently ignore) or to ensure that all receivers of a multicast group confirm support for the options in use.

22. Security Considerations

There are a number of security issues raised by the introduction of options to UDP. Some are specific to this variant, but others are associated with any packet processing mechanism; all are discussed in this section further.

The use of UDP packets with inconsistent IP and UDP Length fields has the potential to trigger a buffer overflow error if not properly handled, e.g., if space is allocated based on the smaller field and copying is based on the larger. However, there have been no reports of such vulnerability and it would rely on inconsistent use of the two fields for memory allocation and copying.

UDP options are not covered by DTLS (datagram transport-layer security). Despite the name, neither TLS [RFC8446] (transport layer security, for TCP) nor DTLS [RFC6347] (TLS for UDP) protect the transport layer. Both operate as a shim layer solely on the user data of transport packets, protecting only their contents. Just as TLS does not protect the TCP header or its options, DTLS does not protect the UDP header or the new options introduced by this document. Transport security is provided in TCP by the TCP Authentication Option (TCP-AO [RFC5925]) or in UDP by the Authentication (AUTH) option (Section 9.9) and UNSAFE Encryption

(UENC) option (Section 10). Transport headers are also protected as payload when using IP security (IPsec) [RFC4301].

UDP options use the TLV syntax similar to that of TCP. This syntax is known to require serial processing and may pose a DOS risk, e.g., if an attacker adds large numbers of unknown options that must be parsed in their entirety, as is the case for IPv6 [RFC8504].

>> Implementations concerned with the potential for this vulnerability MAY implement only the required UDP options and MAY also limit processing of TLVs, either in number of non-padding options or total length, or both. The number of non-zero TLVs allowed in such cases MUST be at least 8.

Because required options come first and at most once each (with the exception of NOPs, which should never need to come in sequences of more than seven in a row), this limits their DOS impact. Note that TLV formats for options does require serial processing, but any format that allows future options, whether ignored or not, could introduce a similar DOS vulnerability.

UDP security should never rely solely on transport layer processing of options. UNSAFE options are the only type that share fate with the UDP data, because of the way that data is hidden in the surplus area until after those options are processed. All other options default to being silently ignored at the transport layer but may be dropped either if that default is overridden (e.g., by configuration) or discarded at the application layer (e.g., using information about the options processed that are passed along with the UDP packet).

UDP fragmentation introduces its own set of security concerns, which can be handled in a manner similar to IP reassembly or TCP segment reordering [CERT18]. In particular, the number of UDP packets pending reassembly and effort used for reassembly is typically limited. In addition, it may be useful to assume a reasonable minimum fragment size, e.g., that non-terminal fragments should never be smaller than 500 bytes.

23. IANA Considerations

Upon publication, IANA is hereby requested to create a new registry for UDP Option Kind numbers, similar to that for TCP Option Kinds. Initial values of this registry are as listed in Section 8. Additional values in this registry are to be assigned from the UNASSIGNED values in Section 8 by IESG Approval or Standards Action

[RFC8126]. Those assignments are subject to the conditions set forth in this document, particularly (but not limited to) those in Section 11.

Although option nicknames are not used in-band, IANA should require UNSAFE safe option values to commence with the letter "U" and avoid that letter as commencing safe options.

Upon publication, IANA is hereby requested to create a new registry for UDP Experimental Option Experiment Identifiers (UDP ExIDs) for use in a similar manner as TCP ExIDs [RFC6994]. UDP ExIDs can be used in either (or both) the EXP or UEXP options. This registry is initially empty. Values in this registry are to be assigned by IANA using first-come, first-served (FCFS) rules [RFC8126]. Options using these ExIDs are subject to the same conditions as new options, i.e., they too are subject to the conditions set forth in this document, particularly (but not limited to) those in Section 11.

24. References

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Appendix A. Implementation Information

The following information is provided to encourage interoperable API implementations.

System-level variables (sysctl):

Name	default	meaning
net.ipv4.udp_opt	0	UDP options available
net.ipv4.udp_opt_ocs	1	Default use OCS
net.ipv4.udp_opt_apc	0	Default include APC
net.ipv4.udp_opt_frag	0	Default fragment
net.ipv4.udp_opt_mds	0	Default include MDS
net.ipv4.udp_opt_mrds	0	Default include MRDS
net.ipv4.udp_opt_req	0	Default include REQ
net.ipv4.udp_opt_resp	0	Default include RES
net.ipv4.udp_opt_time	0	Default include TIME
net.ipv4.udp_opt_auth	0	Default include AUTH
net.ipv4.udp_opt_exp	0	Default include EXP
net.ipv4.udp_opt_uenc	0	Default include UENC
net.ipv4.udp_opt_uexp	0	Default include UEXP

Socket options (sockopt), cached for outgoing datagrams:

Name	meaning
UDP_OPT	Enable UDP options (at all)
UDP_OPT_OCS	Use UDP OCS
UDP_OPT_APC	Enable UDP APC option
UDP_OPT_FRAG	Enable UDP fragmentation
UDP_OPT_MDS	Enable UDP MDS option
UDP_OPT_MRDS	Enable UDP MRDS option
UDP_OPT_REQ	Enable UDP REQ option
UDP_OPT_RES	Enable UDP RES option
UDP_OPT_TIME	Enable UDP TIME option
UDP_OPT_AUTH	Enable UDP AUTH option
UDP_OPT_EXP	Enable UDP EXP option
UDP_OPT_UENC	Enable UDP UENC option
UDP_OPT_UEXP	Enable UDP UEXP option

Send/sendto parameters:

Connection parameters (per-socketpair cached state, part UCB):

Name	Initial value

opts_enabled	net.ipv4.udp_opt
ocs_enabled	net.ipv4.udp_opt_ocs

The following option is included for debugging purposes, and MUST NOT be enabled otherwise.

System variables

net.ipv4.udp_opt_junk 0

System-level variables (sysctl):

Name	default	meaning

net.ipv4.udp_opt_junk	0	Default use of junk

Socket options (sockopt):

Name	params	meaning

UDP_JUNK	-	Enable UDP junk option
UDP_JUNK_VAL	fillval	Value to use as junk fill
UDP_JUNK_LEN	length	Length of junk payload in bytes

Connection parameters (per-socketpair cached state, part UCB):

Name	Initial value

junk_enabled	net.ipv4.udp_opt_junk
junk_value	0xABCD
junk_len	4

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The Some Congestion Experienced ECN Codepoint
draft-morton-tsvwg-sce-03

Abstract

This memo reclassifies ECT(1) to be an early notification of congestion on ECT(0) marked packets, which can be used by AQM algorithms and transports as an earlier signal of congestion than CE. It is a simple, transparent, and backward compatible upgrade to existing IETF-approved AQMs, RFC3168, and nearly all congestion control algorithms.

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1. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119] and [RFC8174] when, and only when, they appear in all capitals, as shown here.

2. Introduction

Traditional TCP congestion control exhibits a "sawtooth" pattern which, in the most favourable cases, oscillates around the optimum operating point of maximum throughput and minimum delay, which exists at the point where the congestion window equals path BDP. The term "sawtooth" brings to mind the straight-edged graphs of TCP Reno, but the equally common TCP CUBIC is essentially similar in character, as are other AIMD-derived algorithms.

A number of proposals have sought to improve this, but introduce various other tradeoffs in return. TCP Vegas is consistently outcompeted by standard TCPs, DCTCP proved to be too aggressive for deployment in the public Internet, and while BBR appears to have avoided both of these problems, its complexity makes it difficult to implement correctly. Each of these proposals is characterised by primarily changing only the endpoints, not the network nodes on the path between them; though DCTCP is intended for use with a specific style of AQM, it can work with standard AQMs as long as there is no competing non-DCTCP traffic.

Some other proposals have attempted to convey information about the network path explicitly, by having network nodes inject data about link capacity and/or utilisation into passing traffic. These proposals have generally been unsuccessful due to the complex slow-path processing required in network nodes, and are not widely deployed. The only successful proposal of this type is Explicit Congestion Notification [RFC3168] which allows an AQM to signal congestion by marking packets with (essentially) a one-bit signal in preference to dropping them.

ECN defines a two-bit field supporting four codepoints, of which three are in active use and the fourth is a semantic duplicate. It was explicitly suggested during ECN's development that new meaning could be given to this spare codepoint, including as a lesser indication of congestion in [RFC3168] (section 20.2). With an alternative use of this codepoint having fallen out of favour, the time is right to revisit this suggestion and propose a workable method of applying it.

In so doing, care must be taken that backwards compatibility is maintained with existing traffic, endpoints and network nodes that are known or suspected to have been deployed. Keeping the changes to on-wire protocols minimal, and the complexity of implementation low, are also highly desirable.

This memo reclassifies ECT(1) to be an early notification of congestion on ECT(0) marked packets, which can be used by AQM algorithms and transports as an earlier signal of congestion than CE ("Congestion Experienced").

This memo also briefly discusses how transports should respond to ECT(1) marked packets. Detailed specifications of this behaviour are left to transport-specific memos.

3. Background

[RFC3168] defines the lower two bits of the (former) TOS byte in the IPv4/6 header as the ECN field. This may take four values: Not-ECT, ECT(0), ECT(1) or CE.

Binary	Keyword	References
00	Not-ECT (Not ECN-Capable Transport)	[RFC3168]
01	ECT(1) (ECN-Capable Transport (1))	[RFC3168]
10	ECT(0) (ECN-Capable Transport (0))	[RFC3168]
11	CE (Congestion Experienced)	[RFC3168]

Table 1

Research has shown that the ECT(1) codepoint goes essentially unused, with the "Nonce Sum" extension to ECN having not been implemented in practice and thus subsequently obsoleted by [RFC8311] (section 3). Additionally, known [RFC3168] compliant senders do not emit ECT(1), and compliant middleboxes do not alter the field to ECT(1), while compliant receivers all interpret ECT(1) identically to ECT(0). These are useful properties which represent an opportunity for improvement.

Experience gained with 7 years of [RFC8290] deployment in the field suggests that it remains difficult to maintain the desired 100% link utilisation, whilst simultaneously strictly minimising induced delay due to excess queue depth - irrespective of whether ECN is in use.

This leads to a reluctance amongst hardware vendors to implement the most effective AQM schemes because their headline benchmarks are throughput-based.

The underlying cause is the very sharp "multiplicative decrease" reaction required of transport protocols to congestion signalling (whether that be packet loss or CE marks), which tends to leave the congestion window significantly smaller than the ideal BDP when triggered at only slightly above the ideal value. The availability of this sharp response is required to assure network stability (AIMD principle), but there is presently no standardised and backwards-compatible means of providing a less drastic signal.

4. Some Congestion Experienced

As consensus has arisen that some form of ECN signaling should be an earlier signal than drop, this memo changes the meaning of ECT(1) to SCE, meaning "Some Congestion Experienced". Since there is no longer ambiguity between two ECT codepoints, ECT(0) is referred to as ECT. The ECN-field codepoint table then becomes:

Binary	Keyword	References
00	Not-ECT (Not ECN-Capable Transport)	[RFC3168]
01	SCE (Some Congestion Experienced)	[This draft]
10	ECT (ECN-Capable Transport)	[RFC3168]
11	CE (Congestion Experienced)	[RFC3168]

Table 2

This permits middleboxes implementing AQM to signal incipient congestion, below the threshold required to justify setting CE, by converting some proportion of ECT codepoints to SCE ("SCE marking"). Existing [RFC3168] compliant receivers MUST transparently ignore this new signal with respect to congestion control, and both existing and SCE-aware middleboxes SHOULD convert SCE to CE in the same circumstances as for ECT, thus ensuring backwards compatibility with [RFC3168] ECN endpoints.

The permitted ECN codepoint transitions by middleboxes are:

From	To
Not-ECT	Not-ECT
ECT	ECT or SCE or CE
SCE	SCE or CE
CE	CE

Table 3

Note that dropping a packet is an allowed action for any ECN codepoint. While that is the only way of indicating congestion with Not-ECT, it may also be used to both indicate and reduce congestion in any state.

To re-state the allowed transitions another way: for ECN-aware flows, the ECN marking of an individual packet MAY be increased by a middlebox to signal congestion, but MUST NOT be decreased, and packets SHALL NOT be altered to appear to be ECN-aware if they were not originally, nor vice versa. Note however that SCE is numerically less than ECT, but semantically greater, and the latter definition applies for this rule.

Receivers and transport protocols conforming to this specification SHALL continue to apply the [RFC3168] interpretation of the CE codepoint, that is, to signal the sender to back off send rate to the same extent as if a packet loss were detected. This maintains compatibility with existing middleboxes, senders and receivers.

New SCE-aware receivers and transport protocols SHOULD interpret the SCE codepoint as an indication of mild congestion, and respond accordingly by applying send rates intermediate between those resulting from a continuous sequence of ECT codepoints, and those resulting from a CE codepoint. The ratio of ECT and SCE codepoints received indicates the relative severity of such congestion, with a higher proportion of SCE codepoints indicating more congestion.

The intent of SCE marking is a "cruise control" signal which permits middleboxes to request relatively small reductions in send rate, or merely a slowing of send rate growth. Accordingly, SCE marks SHOULD progressively trigger exit from exponential slow-start growth, then reduction to Reno-linear growth (for congestion control algorithms which support higher growth rates in congestion-avoidance phase), then a halt to send rate growth, then a gradual reduction of send

rate. For immediate large reductions of send rate, the CE mark MUST retain its original Multiplicative Decrease power as per [RFC8511], and compliant AQMs SHOULD retain the ability to employ it where appropriate.

Details of how to implement SCE awareness at the transport layer are left to additional Internet Drafts. To ensure RTT-fair convergence with single-queue SCE AQMs, transports SHOULD stabilise at lower SCE-mark ratios for higher BDPs, and MAY reduce their response to CE marks IFF they are responding to SCE signals received at around the same time (eg. within 1-2 RTTs) in the same flow.

To maximise the benefit of SCE, middleboxes SHOULD begin to produce SCE marks at lower congestion levels than they begin to produce CE marks. This will usually ensure that SCE-aware flows avoid receiving CE marks. When a single-queue AQM is upgraded to SCE awareness, this will tend to cause SCE flows to give way to non-SCE flows; to avoid this behaviour, single-queue AQMs MAY be left as [RFC3168] compliant without SCE support.

For the avoidance of doubt, a decision to mark CE or to drop a packet always takes precedence over SCE marking.

5. Design Rationale

The SCE design sees ECN as a "network feature". The risks with ECN signaling (Section 5.1), the need to handle unresponsive flows (Section 5.2), the utility of fairness (Section 5.3), and the availability of only one ECN codepoint all influenced the SCE signaling design. This section discusses these related concerns, along with what is needed from middleboxes to address them, and how that ultimately led to the selection of ECT(1) as an additional signal of lesser congestion (Section 5.4).

5.1. Risks with ECN Signaling

The safety and effectiveness of ECN signaling depends upon the unaltered transmission of the ECN bits, both for the indication of ECN support, and for ECN signaling. Unlike a drop, which is reliably and irrevocably signaled, ECN signals may be erased or manipulated. Specifically, any of the following results in the lack of a congestion response, which is likely to lead to the near starvation of competing flows:

- * if transports indicate ECT(0) but do not respond to CE
- * if packets are erroneously changed from Not-ECT to ECT(0) in the network

- * if CE marks are erased after a bottleneck
- * if ECE marks are erased post-negotiation

Although the lack of a congestion response is similar to when transports do not respond appropriately to drop, the difference is that with ECN, the behavior can be brought about in the network, without changes to the endpoint. This may happen by accident, for example due to a broken network configuration or endpoint implementation, or on purpose, e.g. using a simple firewall rule.

Unresponsive flow mitigation, discussed in the next section, deals with flows that are not responding to congestion signals, including for the reasons listed above.

5.2. Unresponsive Flows

A single unresponsive flow has the potential to nearly starve all other competing flows in a congested bottleneck, resulting in unacceptable network delays and collapses in throughput. The need to handle unresponsive flows is corroborated in [RFC7567] (section 4), stating:

"Research, engineering, and measurement efforts are needed regarding the design of mechanisms to deal with flows that are unresponsive to congestion notification or are responsive, but are more aggressive than present TCP."

The source language from [RFC2309] (section 5) is more direct:

"It is urgent to begin or continue research, engineering, and measurement efforts contributing to the design of mechanisms to deal with flows that are unresponsive to congestion notification or are responsive but more aggressive than TCP."

The [COBALT] AQM algorithm is one example of how unresponsive flows can be dealt with, using the [BLUE] algorithm to detect overload and trigger drops.

Regardless of how it's done exactly, unresponsive flow mitigation is most effectively implemented with some level of flow awareness, so that drops may be directed to the offending flow/s. Once flow awareness is available, fairness steering becomes possible, discussed further in the following section.

5.3. Fairness

In order for SCE flows to compete fairly with non-SCE flows, at least one of the following is required: some form of fairness steering, or some way of separating SCE and non-SCE flows. Following is a non-exhaustive list of options:

- * FQ (fair queueing), to isolate and schedule flows fairly from separate queues
- * AF (approximate fairness), so that SCE and non-SCE flows can share the same queue, e.g. [AFD], [I-D.morton-tsvwg-codel-approx-fair], [I-D.morton-tsvwg-lightweight-fair-queueing]
- * DSCP [RFC2474], to explicitly separate SCE and non-SCE flows (see Section 6)

When available, fairness is viewed as an advantage, in that it:

- * controls aggressive flows
- * prevents network bias
- * promotes the fair interoperation between the ever-expanding matrix of new congestion control mechanisms

The abundance of new and proposed congestion controls is making their fair competition across bandwidths, RTTs and network conditions more difficult if not impossible to ensure in the endpoint alone [CC-REVOLUTION] [CC-COMPAT]. Congestion control implementations may dominate one another under different conditions, e.g. [BBR-CUBIC], while the widespread deployment of potentially beneficial congestion controls that seek to minimize delay is discouraged by the fact that they are often out-competed in bottlenecks by standard TCP. Fairness in the network both improves these conditions and assists transports responding to SCE.

5.4. ECT(1) as SCE

With only a single ECN codepoint remaining, options are limited for how to signal congestion with high fidelity. Meanwhile, the recent rise in ECN signaling makes backwards compatibility with [RFC3168] a practical requirement.

Fortunately, the same network technologies that mitigate the well recognized risks listed in Section 5 above, also make the use of ECT(1) as defined by SCE possible, without a separate traffic identifier. Where those technologies cannot be deployed, Diffserv

may be used to identify SCE traffic (see Section 6), a purpose for which it was expressly designed. Where that is impossible, SCE allows a graceful fallback to [RFC3168] ECN. SCE's usage of ECT(1) provides a safe and solid foundation on which future innovations in the network can improve the availability and performance of high-fidelity congestion signaling.

6. Diffserv Usage

SCE is not dependent on Diffserv [RFC2474] for its signaling, but makes use of it in the following ways:

- * to mark SCE traffic for experimental or private use
- * to assist middleboxes in their operation
- * to request special SCE treatment, such as low delay or low cost

6.1. SCE Diffserv Codepoints (DSCPs)

All SCE DSCPs indicate SCE support in the originating endpoint. This MAY assist SCE marking middleboxes in their operation, but MUST NOT be depended upon for effective congestion control. See Section 7.3 for an example of such a usage.

SCE middleboxes MUST retain any SCE DSCPs that arrive on incoming packets, and MUST NOT set them on packets that do not already have them.

The SCE DSCPs MAY be set on TCP ACK and control packets which have the Not-ECT codepoint set in the ECN field, IFF the TCP connection as a whole is SCE capable (or in the process of being negotiated as such). This allows all packets relating to that connection to be treated equally by middleboxes which distinguish them. Should ECN negotiation fail, the DSCP should be changed to some non-SCE value for subsequent traffic on that connection.

6.1.1. SCE-CAPABLE

The SCE-CAPABLE DSCP indicates SCE support, with standard, best-effort service implied. This is the appropriate service for capacity-seeking traffic, for which latency is a secondary consideration.

6.1.2. SCE-LOWDELAY

The SCE-LOWDELAY DSCP is used to both indicate SCE support and request low-delay service. This MAY be used by AQMs to select a low delay queue with tighter marking parameters that reduce delay, at the possible expense of throughput.

6.1.3. SCE-LOWCOST

The SCE-LOWCOST DSCP is used to both indicate SCE support and request altruistic low-cost service. This MAY be used by AQMs to deprioritise this traffic in favour of low-delay and best-effort traffic, similar to the LE PHB [RFC8622].

6.2. Diffserv Codepoints for Experimental and Private Use

Prior to approval for public experiment, the SCE DSCPs are defined in the experimental pool xxxx11, and the following rules MUST be observed to contain SCE traffic within the experimental network:

- * SCE senders SHOULD set one of the SCE DSCPs when participating in an SCE experimental network.
- * SCE middleboxes MUST NOT mark SCE on packets lacking an SCE DSCP, or packets that may leave the experimental network.
- * SCE receivers MUST check that one of the SCE DSCPs is present before returning SCE feedback.
- * All SCE DSCPs MUST be bleached at the experimental network boundaries.

The following values are proposed for guidance only. Because they are in the experimental pool, they may be changed to suit the environment:

Name	Value (Binary)	Value (Decimal)
SCE-CAPABLE	000111	7
SCE-LOWDELAY	001011	11
SCE-LOWCOST	000011	3

Table 4

6.3. Diffserv Codepoints for Public Use

In the event that SCE is approved for public experiment, the DSCPs will be allocated in an appropriate standards action pool, using a value that is intended to be treated as best-effort traffic by existing deployed devices.

One of the SCE DSCPs SHOULD be set by sending endpoints on all SCE capable traffic. However, they neither need to be checked by middleboxes that do not require them before marking SCE, nor by receiving endpoints before returning SCE feedback. That way, they can serve as hints for middleboxes, but the SCE signaling mechanism is not dependent on end-to-end DSCP traversal.

Unless and until a public experiment is approved, the guidance in Section 6.2 MUST be followed.

7. Examples of use

7.1. Codel-type AQMs

A simple and natural way to implement SCE in a Codel-type AQM is to mark all ECT packets as SCE if they are over half the Codel target sojourn time, and not marked CE by Codel itself. This threshold function does not necessarily produce the best performance, but is very easy to implement and provides useful information to SCE-aware flows, often sufficient to avoid receiving CE marks whilst still efficiently using available capacity.

For a more sophisticated approach avoiding even small-scale oscillation, a stochastic ramp function may be implemented with 100% marking at the Codel target, falling to 0% marking at or above zero sojourn time. The lower point of the ramp should be chosen so that SCE is not accidentally signalled due to CPU scheduling latencies or serialisation delays of single packets. Absent rigorous analysis of these factors, setting the lower limit at half the Codel target should be safe in many cases.

The default configuration of Codel is 100ms interval, 5ms target. A typical ramp function for these parameters might cease marking below 2.5ms sojourn time, increase marking probability linearly to 100% at 5ms, and mark at 100% for sojourn times above 5ms (in which CE marking is also possible).

In single-queue AQMs, the above strategy will result in SCE flows yielding to pressure from non-SCE flows, since CE marks do not occur until SCE marking has reached 100%. A balance between smooth SCE behaviour and fairness versus non-SCE traffic can be found by having

the marking ramp cross the Codel target at some lower SCE marking rate, perhaps even 0%. A two-part ramp, reaching $1/\sqrt{X}$ at the Codel target (for some chosen X , a $cwnd$ at which the crossover between smoothness and fairness occurs) and ramping up more steeply thereafter, has been implemented successfully for experimentation.

The CNQ algorithm [I-D.morton-tsvwg-cheap-nasty-queueing] offers a relatively simple way to limit this yielding behaviour and ensure that, even in competition with non-SCE flows, SCE flows maintain a reasonable minimum throughput capability. This may be sufficient to avoid the need for the two-part ramp described above.

Flow-isolating AQMs, including especially CNQ and DRR++ based algorithms, should avoid signalling SCE to flows classified as "sparse", in order to encourage the fastest possible convergence to the fair share.

7.2. RED-type AQMs (including PIE)

There are several reasonable methods of producing SCE signals in a RED-type AQM.

The simplest would be a threshold function, giving a hard boundary in queue depth between 0% and 100% SCE marking. This could be a sensible option for limited hardware implementations. The threshold should be set below the point at which a growing queue might trigger CE marking or packet drops.

Another option would be to implement a second marking probability function, occupying a queue-depth space just below that occupied by the main marking probability function. This should be arranged so that high marking rates (ideally 100%) are achieved at or before the point at which CE marking or packet drops begin.

For PIE specifically, a second marking probability function could be added with the same parameters as the main marking probability function, except for a lower `QDELAY_REF` value. This would result in the SCE marking probability remaining strictly higher than the CE marking probability for ECT flows.

7.3. Simple Two-Queue Middleboxes

In high-capacity or resource constrained SCE marking middleboxes, DSCP may be used to select one of two queues, in lieu of implementing fairness steering. Packets marked with an SCE DSCP are placed in an SCE queue, where an AQM instance may mark congestion with either SCE or CE. Packets not marked with an SCE DSCP are placed in a second [RFC3168] queue, whose AQM instance may only mark congestion with CE.

For approximate flow fairness, the queues may be scheduled in proportion to the number of flows they contain.

Note that as long as the SCE DSCP remains intact from the sending endpoint to the marking queue, the SCE queue may be used. If it has been erased or altered to a non-SCE DSCP, the packet will be placed in the [RFC3168] queue, and may still benefit from standard ECN.

If this middlebox is to be used in public environments, some form of unresponsive flow mitigation is warranted to ensure that flows haven't indicated their support for either SCE or [RFC3168] ECN incorrectly. If flows do not respond to the signals they advertise support for, they will dominate competing traffic in the same queue.

7.4. TCP

The proposed mechanism for TCP to feed back SCE signals to the sender is outlined in [I-D.grimes-tcpm-tcpsce]. Use is made of the redundant NS bit in the TCP header, which was formerly associated with ECT(1) in the Nonce Sum specification.

The recommended response to each single segment marked with SCE is to reduce cwnd by an amortised $1/\sqrt{\text{cwnd}}$ segments. Other responses, such as the $1/\text{cwnd}$ from DCTCP, are also acceptable but may perform less well.

7.5. Other

New transports under development, such as QUIC, may implement a fine-grained signal back to the sender based on SCE. QUIC itself appears to have this sort of feedback already (counting ECT(0), ECT(1) and CE packets received), and the data should be made available for congestion control.

8. Compatibility

8.1. Existing ECN & AQM Deployments

SCE explicitly retains [RFC8511] compliant Multiplicative Decrease responses to CE marks, and conventional Multiplicative Decrease responses to packet loss. SCE senders' behaviour is thus naturally compliant with existing specifications when running over existing networks.

Existing endpoints, supporting Not-ECT or [RFC3168] compliant congestion control, are required to treat SCE marks (that is, ECT(1)) as identical to ECT(0), and will thus transparently ignore SCE marks. This is allowed for in SCE's design, and allows SCE middleboxes to be deployed into a heterogeneous network.

Hence the incremental deployability of SCE endpoints and middleboxes is good.

8.2. L4S

L4S [I-D.ietf-tsvwg-l4s-arch] also claims the ECT(1) codepoint, with significantly different semantic meaning than SCE, so a discussion around the potential for L4S and SCE compatibility is warranted. In the L4S system, ECT(1) is used to identify L4S flows, to distinguish them from [RFC3168] flows - necessary since in L4S, the semantic meaning of CE marks is also changed.

Since L4S connections are explicitly negotiated through support of AccECN, and AccECN doesn't support SCE, there is no ambiguity regarding the mode of the connection as far as endpoints are concerned.

SCE middleboxes will treat L4S flows in the same way as [RFC3168] does. However, because SCE middleboxes are likely to upgrade ECT(1) marked packets to CE at a higher threshold than L4S middleboxes would, L4S flows will outcompete non-L4S flows in a single SCE-aware queue. This is the same known safety concern with L4S deployment in regards to existing [RFC3168] queues, resulting from the redefinition of CE in L4S. Fairness steering in SCE middleboxes could mitigate this.

L4S middleboxes may interpret ECT packets which have received SCE markings at some other SCE-aware middlebox as though they were L4S traffic. This may result in a higher CE marking rate and/or different queuing behaviour. It may also result in the reordering of packets for both SCE and non-SCE aware flows through L4S middleboxes, as packets marked ECT(1) will on average traverse the bottleneck with lower delay than packets not marked ECT(1). Although this could be mitigated by [I-D.ietf-tcpm-rack], it may lead to reduced throughput and head-of-line blocking for flows that traverse both SCE and L4S bottlenecks.

There are at least two secondary concerns brought about by the L4S use of ECT(1) as a traffic identifier:

- * If it is found necessary to firewall L4S traffic off from the general Internet, then SCE-marked packets are also likely to be dropped at this boundary. This could have a significantly detrimental effect on ECT traffic traversing both an SCE and an L4S enabled network, even if the endpoints are not explicitly SCE aware.
- * If it is found necessary to bleach ECT(1) in order to disable L4S in a network, this would erase SCE signals sent to endpoints. Although not ideal, SCE transports would still safely fall back to relying on CE for congestion notification.

Lastly, an ambiguous definition of ECT(1) complicates network debugging with packet captures, since it would be unclear whether a packet was marked ECT(1) due to congestion at an SCE bottleneck, or because it is an L4S flow. Although examination of other packets in the flow could reduce this ambiguity, the necessity of observing flow state is generally discouraged for debugging purposes.

Thus far, the working group is operating under the assumption that coexistence of SCE and L4S is not an option.

9. Ongoing Research and Development

The SCE proposal is a work in progress, with ongoing or planned work in at least the following areas:

- * AQM strategies for a small number of FIFO queues
- * Tunnel traversal, with possible updates to [RFC3168] and [RFC6040]
- * Research ways of reducing RTT dependence (Prague requirement #5)
- * Performance in environments with jitter and burstiness
- * New testing tools that cover many short flows, and VBR UDP flows
- * Testing, with guidance from [RFC2914], [RFC7141] and [RFC5033]

10. Related Work

[RFC8087] [RFC7567] [RFC7928] [RFC8290] [RFC8289] [RFC8033] [RFC8034]
[I-D.morton-tsvwg-interflow-intraflow-delays]

11. IANA Considerations

There are no IANA considerations.

12. Security Considerations

An adversary could inappropriately set SCE marks at middleboxes he controls to slow down SCE-aware flows, eventually reaching a minimum congestion window. However, the same threat already exists with respect to inappropriately setting CE marks on normal ECN flows, and this would have a greater impact per mark. Therefore no new threat is exposed by SCE in practice.

An adversary could also simply ignore SCE marks at the receiver, or ignore SCE information fed back from the receiver to the sender, in an attempt to gain some advantage in throughput. Again, the same could be said about ignoring CE marks, so no truly new threat is exposed. Additionally, correctly implemented SCE detection may actually improve long-term goodput compared to ignoring SCE.

An adversary could erase congestion information by converting SCE marks to ECT or Not-ECT codepoints, thus hiding it from the receiver. This has equivalent effects to ignoring SCE signals at the receiver. An identical threat already exists for erasing congestion information from CE marked packets, and may be mitigated by AQMs switching to dropping packets from flows observed to be non-responsive to CE.

An adversary could drop SCE-marked packets, believing them to be bogons (see also L4S Compatibility, above). Endpoints should be able to recover from this through retransmission and a reduction of cwnd. However, it is possible for this to lead to a significant denial of service. A workaround is to disable ECN for connections over the affected path.

13. Acknowledgements

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Authenticated Chunks for the Stream Control Transmission Protocol (SCTP)
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Abstract

This document obsoletes RFC4895 if approved. This document describes a new chunk type, several parameters, and procedures for the Stream Control Transmission Protocol (SCTP). This new chunk type can be used to authenticate SCTP chunks by using shared keys between the sender and receiver. The new parameters are used to establish the shared keys.

This document describes the limitations with the current SCTP AUTH RFC4895 and thus enhances the document to resolve such ambiguities and thus strengthen the overall AUTH procedure.

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1. Introduction

SCTP uses 32-bit verification tags to protect itself against blind attackers. These values are not changed during the lifetime of an SCTP association.

Looking at new SCTP extensions, there is the need to have a method of proving that an SCTP chunk(s) was really sent by the original peer that started the association and not by a malicious attacker.

Using Transport Layer Security (TLS), as defined in RFC 3436 [RFC3436], does not help because it only secures SCTP user data.

Therefore, an SCTP extension that provides a mechanism for deriving shared keys for each association is presented. These association shared keys are derived from endpoint pair shared keys, which are configured and might be empty, and data that is exchanged during the SCTP association setup.

The extension presented in this document allows an SCTP sender to authenticate chunks using shared keys between the sender and receiver. The receiver can then verify that the chunks are sent from the sender and not from a malicious attacker (as long as the attacker does not know an association shared key).

The extension described in this document places the result of a Hashed Message Authentication Code (HMAC) computation before the data covered by that computation. Placing it at the end of the packet would have required placing a control chunk after DATA chunks in case of authenticating DATA chunks. This would break the rule that control chunks occur before DATA chunks in SCTP packets. It should also be noted that putting the result of the HMAC computation after the data being covered would not allow sending the packet during the computation of the HMAC because the result of the HMAC computation is needed to compute the CRC32C checksum of the SCTP packet, which is placed in the common header of the SCTP packet.

The SCTP extension for Dynamic Address Reconfiguration (ADD-IP) requires the usage of the extension described in this document. The SCTP Partial Reliability Extension (PR-SCTP) can be used in conjunction with the extension described in this document.

2. Conventions

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

3. New Parameter Types

This section defines the new parameter types that will be used to negotiate the authentication during association setup. Table 1 illustrates the new parameter types.

Parameter Type	Parameter Name
0x8002	Random Parameter (RANDOM)
0x8003	Chunk List Parameter (CHUNKS)
0x8004	Requested HMAC Algorithm Parameter (HMAC-ALGO)

Note that the parameter format requires the receiver to ignore the parameter and continue processing if the parameter is not understood. This is accomplished (as described in RFC 4960 [RFC4960], Section 3.2.1.) by the use of the upper bits of the parameter type.

3.1. Random Parameter (RANDOM)

This parameter is used to carry a random number of an arbitrary length.

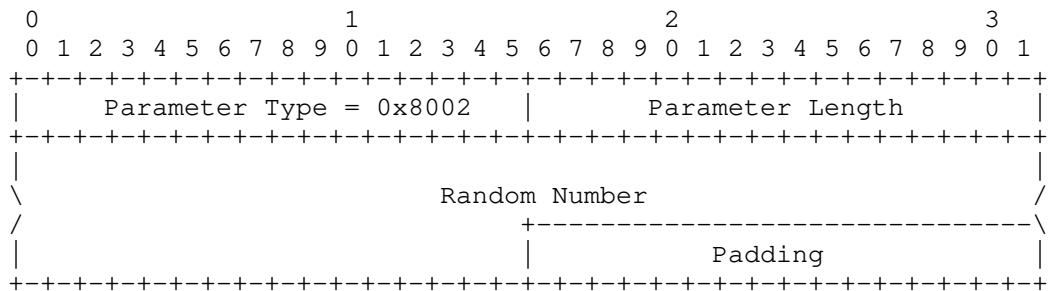


Figure 1: RANDOM Parameter

Parameter Type: 2 bytes (unsigned integer)
This value MUST be set to a value of 0x8002.

Parameter Length: 2 bytes (unsigned integer)
This value is the length of the Random Number in bytes plus 4.

Random Number: n bytes (unsigned integer)
This value represents an arbitrary Random Number in network byte order.

Padding: 0, 1, 2, or 3 bytes (unsigned integer)

If the length of the Random Number is not a multiple of 4 bytes, the sender MUST pad the parameter with all zero bytes to make the parameter 32-bit aligned. The Padding MUST NOT be longer than 3 bytes and it MUST be ignored by the receiver.

The RANDOM parameter MUST be included once in the INIT or INIT-ACK chunk, if the sender wants to send or receive authenticated chunks, to provide a 32-byte Random Number. For 32-byte Random Numbers, the Padding is empty.

3.2. Chunk List Parameter (CHUNKS)

This parameter is used to specify which chunk types are required to be authenticated before being sent by the peer.

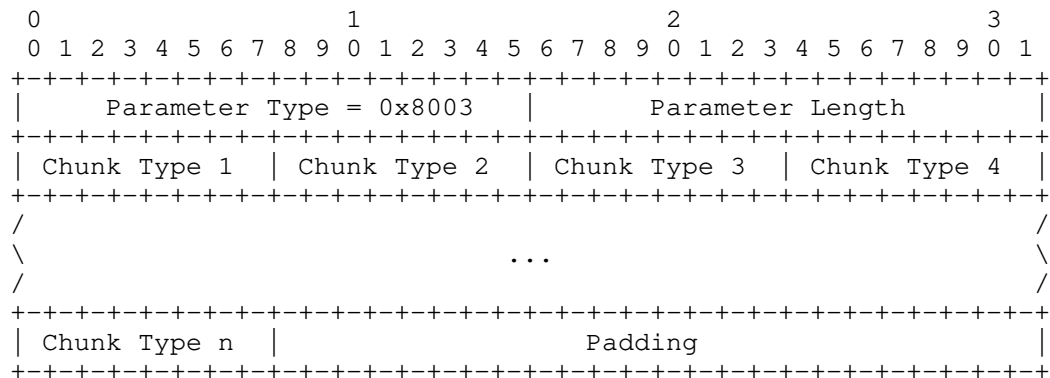


Figure 2: Chunks Parameter

Parameter Type: 2 bytes (unsigned integer)

This value MUST be set to 0x8003.

Parameter Length: 2 bytes (unsigned integer)

This value is the number of listed Chunk Types plus 4.

Chunk Type n: 1 byte (unsigned integer)

Each Chunk Type listed is required to be authenticated when sent by the peer.

Padding: 0, 1, 2, or 3 bytes (unsigned integer)

If the number of Chunk Types is not a multiple of 4, the sender MUST pad the parameter with all zero bytes to make the parameter 32-bit aligned. The Padding MUST NOT be longer than 3 bytes and it MUST be ignored by the receiver.

The CHUNKS parameter MUST be included once in the INIT or INIT-ACK chunk if the sender wants to receive authenticated chunks. Its maximum length is 260 bytes.

The chunk types for INIT, INIT-ACK, SHUTDOWN-COMPLETE, and AUTH chunks MUST NOT be listed in the CHUNKS parameter. However, if a CHUNKS parameter is received then the types for INIT, INIT-ACK, SHUTDOWN-COMPLETE, and AUTH chunks MUST be ignored.

If the receiver of the CHUNKS parameters finds that a particular chunk type is unknown or unrecognized, then the receiver SHOULD ignore this chunk type and continue parsing and add only the valid and recognized chunks to its AUTH chunk list which it needs to send in the authenticated way.

3.3. Requested HMAC Algorithm Parameter (HMAC-ALGO)

This parameter is used to list the HMAC Identifiers the peer MUST use.

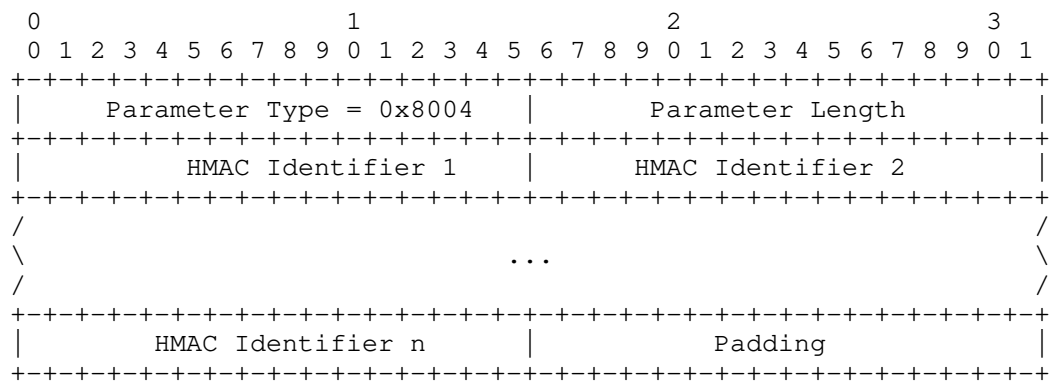


Figure 3: HMAC-ALGO

Parameter Type: 2 bytes (unsigned integer)
This value MUST be set to 0x8004.

Parameter Length: 2 bytes (unsigned integer)
This value is the number of HMAC Identifiers multiplied by 2, plus 4.

HMAC Identifier n: 2 bytes (unsigned integer)
The values expressed are a list of HMAC Identifiers that may be used by the peer. The values are listed by preference, with respect to the sender, where the first HMAC Identifier listed is the one most preferable to the sender.

Padding: 0 or 2 bytes (unsigned integer)

If the number of HMAC Identifiers is not even, the sender MUST pad the parameter with all zero bytes to make the parameter 32-bit aligned. The Padding MUST be 0 or 2 bytes long and it MUST be ignored by the receiver.

The HMAC-ALGO parameter MUST be included once in the INIT or INIT-ACK chunk if the sender wants to send or receive authenticated chunks.

Table 2 shows the currently defined values for HMAC Identifiers.

HMAC Identifier	Message Digest Algorithm
0	Reserved
1	SHA-1 defined in [FIPS180-2]
2	Reserved
3	SHA-256 defined in [FIPS180-2]

Every endpoint supporting SCTP chunk authentication MUST support the HMAC based on the SHA-1 algorithm.

3.4. Supported Extensions Parameter

The supported Extensions parameter as defined in RFC 5061 [RFC5061] section 4.2.7 MUST be exchanged by including AUTH chunk as defined in Section 5.1 below to support this SCTP protocol extension. This is done keeping in mind to make this extension consistent with all the SCTP extensions which mandates sending this extension parameter in INIT/INIT-ACK chunk whenever a new chunk type is added.

4. New Error Cause

This section defines a new error cause that will be sent if an AUTH chunk is received with an unsupported HMAC Identifier. Table 3 illustrates the new error cause.

Cause Code	Error Cause Name
0x0105	Unsupported HMAC Identifier

4.1. Unsupported HMAC Identifier Error Cause

This error cause is used to indicate that an AUTH chunk has been received with an unsupported HMAC Identifier.

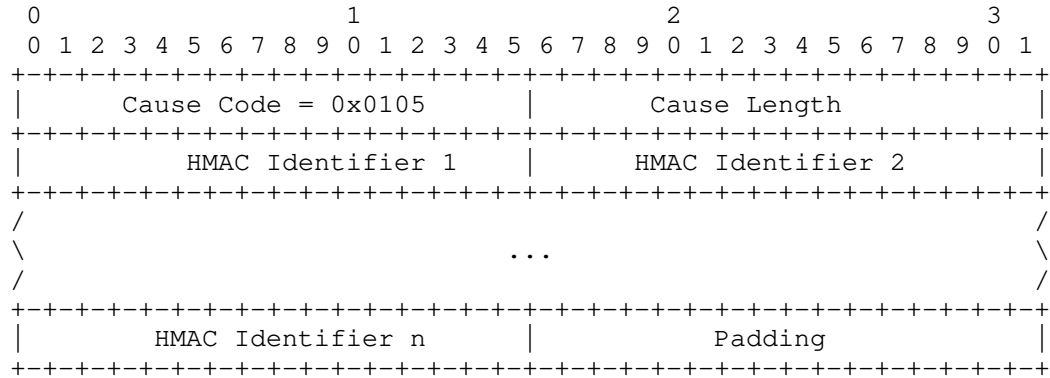


Figure 4: UnSupported HMAC Identifier Error Cause

Cause Code: 2 bytes (unsigned integer)
This value MUST be set to 0x0105.

Cause Length: 2 bytes (unsigned integer)
This value is the number of unsupported HMAC Identifiers multiplied by 2, plus 4.

HMAC Identifier: 2 bytes (unsigned integer)
This value is the HMAC Identifier which is not supported.

Padding: 0 to 2 bytes (unsigned integer)
If the number of unsupported HMAC Identifiers is not even, the sender MUST pad the parameter with all zero bytes to make the parameter 32-bit aligned. The Padding MUST be 0 or 2 bytes long and it MUST be ignored by the receiver.

5. New Chunk Type

This section defines the new chunk type that will be used to authenticate chunks. Table 4 illustrates the new chunk type.

Chunk Type	Chunk Name
0x0F	Authentication Chunk (AUTH)

It should be noted that the AUTH-chunk format requires the receiver to ignore the chunk if it is not understood and silently discard all chunks that follow. This is accomplished (as described in RFC 4960 [RFC4960], Section 3.2.) by the use of the upper bits of the chunk type.

5.1. Authentication Chunk (AUTH)

This chunk is used to hold the result of the HMAC calculation.

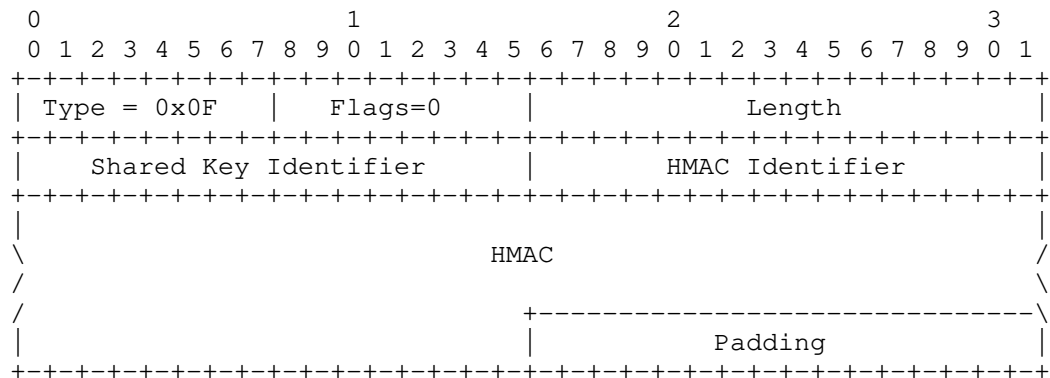


Figure 5: AUTH Chunk

Type: 1 byte (unsigned integer)

This value MUST be set to 0x0F for all AUTH-chunks.

Flags: 1 byte (unsigned integer)

SHOULD be set to zero on transmit and MUST be ignored on receipt.

Length: 2 bytes (unsigned integer)

This value holds the length of the HMAC in bytes plus 8.

Shared Key Identifier: 2 bytes (unsigned integer)

This value describes which endpoint pair shared key is used.

HMAC Identifier: 2 bytes (unsigned integer)

This value describes which message digest is being used. Table 2 shows the currently defined values.

HMAC: n bytes (unsigned integer)

This holds the result of the HMAC calculation.

Padding: 0, 1, 2, or 3 bytes (unsigned integer)

If the length of the HMAC is not a multiple of 4 bytes, the sender MUST pad the chunk with all zero bytes to make the chunk 32-bit

aligned. The Padding MUST NOT be longer than 3 bytes and it MUST be ignored by the receiver.

The control chunk AUTH MUST NOT appear more than once in an SCTP packet. All control and data chunks that are placed after the AUTH chunk in the packet are sent in an authenticated way. Those chunks placed in a packet before the AUTH chunk are not authenticated. Please note that DATA chunks can not appear before control chunks in an SCTP packet.

6. Procedures

6.1. Establishment of an Association Shared Key

An SCTP endpoint willing to receive or send authenticated chunks MUST send one RANDOM parameter in its INIT or INIT-ACK chunk. The RANDOM parameter MUST contain a 32-byte Random Number. Though the section 3.1 explains the random to be of arbitrary length keeping extensibility in mind. But in the context of this document, it needs to be 32-byte in size. The Random Number should be generated in accordance with RFC 4086 [RFC4086]. If the Random Number is not 32 bytes, the association MUST be aborted. The ABORT chunk SHOULD contain the error cause 'Protocol Violation'. In case of INIT collision, the rules governing the handling of this Random Number follow the same pattern as those for the Verification Tag, as explained in Section 5.2.4 of RFC 4960 [RFC4960]. Therefore, each endpoint knows its own Random Number and the peer's Random Number after the association has been established.

An SCTP endpoint has a list of chunks it only accepts if they are received in an authenticated way. This list is included in the INIT and INIT-ACK, and MAY be omitted if it is empty. Since this list does not change during the lifetime of the SCTP endpoint there is no problem in case of INIT collision.

Each SCTP endpoint MUST include in the INIT and INIT-ACK a HMAC-ALGO parameter containing a list of HMAC Identifiers it requests the peer to use. The receiver of an HMAC-ALGO parameter SHOULD use the first listed algorithm it supports. The HMAC algorithm based on SHA-1 MUST be supported and included in the HMAC-ALGO parameter. An SCTP endpoint MUST NOT change the parameters listed in the HMAC-ALGO parameter during the lifetime of the endpoint.

Additionally, local endpoint MUST also include supported extension parameter as defined in RFC 5061 [RFC5061] section 4.2.7 containing Auth chunk type as defined in Section 5.1 in INIT/INIT-ACK chunk to notify the peer endpoint about the negotiation attempt to enable AUTH extension which will be used to encode certain chunks based on

negotiation during the lifetime of this association. This parameter is added to make this procedure consistent with all other SCTP extensions with new chunk types which expects this parameter needs to be sent in INIT/INIT-ACK messages. Some implementations supporting this extension already mandates peer to include this parameter if this procedure needs to be enabled.

Both endpoints of an association MAY have endpoint pair shared keys that are byte vectors and pre-configured or established by another mechanism. They are identified by the Shared Key Identifier. For each endpoint pair shared key, an association shared key is computed. If there is no endpoint pair shared key, only one association shared key is computed by using an empty byte vector as the endpoint pair shared key.

The RANDOM parameter, the CHUNKS parameter, and the HMAC-ALGO parameter sent by each endpoint are concatenated as byte vectors. The CHUNKS parameter received from peer endpoint without stripping off any unrecognized chunks are used for concatenation, since doing so will result in mismatch of CHUNKS parameter used between the local and remote endpoint. These parameters include the parameter type, parameter length, and the parameter value, but padding is omitted; all padding MUST be removed from this concatenation before proceeding with further computation of keys. The concatenation MUST follow the order of the parameters, RANDOM parameter, followed by the CHUNKS parameter, followed by the HMAC-ALGO parameter. Parameters that were not sent are simply omitted from the concatenation process. The resulting two vectors are called the two key vectors.

From the endpoint pair shared keys and the key vectors, the association shared keys are computed. This is performed by selecting the numerically smaller key vector and concatenating it to the endpoint pair shared key, and then concatenating the numerically larger key vector to that. If the key vectors are equal as numbers but differ in length, then the concatenation order is the endpoint shared key, followed by the shorter key vector, followed by the longer key vector. Otherwise, the key vectors are identical, and may be concatenated to the endpoint pair key in any order. The concatenation is performed on byte vectors, and all numerical comparisons use network byte order to convert the key vectors to a number. The result of the concatenation is the association shared key.

6.2. Negotiation of Auth Procedure

Negotiation between SCTP peer endpoints would be required, when either end of the SCTP endpoint doesn't support AUTH extension. If the either end of the SCTP association doesn't support AUTH

extension, then the SCTP association MUST switch to normal mode without AUTH support. Both the end of the association endpoint would discover if the peer endpoint doesn't support the same by the time INIT/INI-ACK message is exchanged between them.

6.2.1. Receiving INIT containing AUTH parameters when AUTH extension is not supported

The endpoint receiving such an INIT would skip the AUTH related parameter based on the upper 2 bits as explained in RFC 4960 [RFC4960] section 3.2.1.

6.2.2. Receiving INIT-ACK without AUTH parameters when AUTH extension was sent in INIT chunk

On receiving such an INIT-ACK, then local endpoint SHOULD disable the AUTH feature and continue normal association establishment considering that the AUTH negotiation has failed. If AUTH is mandatory as per local policies or the interface from the application layer mandates it, then local endpoint MUST ABORT the association establishment procedure by sending ABORT message in response to this INIT-ACK with ABORT cause as User Initiated Abort and release all resources that has been allocated for this association.

6.2.3. Receiving INIT without AUTH parameters when AUTH is supported locally

On receiving such an INIT, endpoint can switch the association to normal association establishment without AUTH enabled realizing that peer endpoint is not AUTH enabled. If AUTH is mandatory as per local policies or the interface from the application layer mandates it, then remote endpoint MUST send ABORT message in response to this INIT chunk with ABORT cause as User Initiated Abort.

6.3. Association Initialization Collision

No special handling is required in case of collision and the procedures as explained in Section 6.2 above applies in this case as well.

6.4. Sending Authenticated Chunks

Endpoints MUST send all requested chunks that have been authenticated where this has been requested by the peer. The other chunks MAY be sent whether or not they have been authenticated. If endpoint pair shared keys are used, one of them MUST be selected for authentication.

To send chunks in an authenticated way, the sender MUST include these chunks after an AUTH chunk. This means that a sender MUST bundle chunks in order to authenticate them.

If the endpoint has no endpoint pair shared key for the peer, it MUST use Shared Key Identifier zero with an empty endpoint pair shared key. If there are multiple endpoint shared keys the sender selects one and uses the corresponding Shared Key Identifier.

The sender MUST calculate the Message Authentication Code (MAC) (as described in RFC 6151 [RFC6151]) using the hash function H as described by the HMAC Identifier and the shared association key K based on the endpoint pair shared key described by the Shared Key Identifier. The 'data' used for the computation of the AUTH-chunk is given by the AUTH chunk with its HMAC field set to zero (as shown in Figure 6) followed by all the chunks that are placed after the AUTH chunk in the SCTP packet.

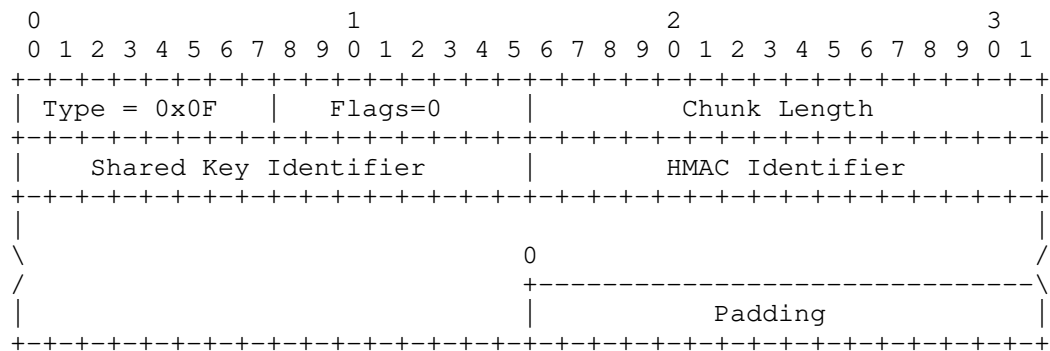


Figure 6: Sending Auth Chunk

Please note that all fields are in network byte order and that the field that will contain the complete HMAC is filled with zeroes. The length of the field shown as zero is the length of the HMAC described by the HMAC Identifier. The padding of all chunks being authenticated MUST be included in the HMAC computation.

The sender fills the HMAC into the HMAC field and sends the packet.

6.5. Receiving Authenticated Chunks

The receiver has a list of chunk types that it expects to be received only after an AUTH-chunk. This list has been sent to the peer during the association setup. It MUST silently discard these chunks if they are not placed after an AUTH chunk in the packet.

The receiver MUST use the HMAC algorithm indicated in the HMAC Identifier field. If this algorithm was not specified by the receiver in the HMAC-ALGO parameter in the INIT or INIT-ACK chunk during association setup, the AUTH chunk and all the chunks after it MUST be discarded and an ERROR chunk SHOULD be sent with the error cause defined in Section 4.1.

If an endpoint with no shared key receives a Shared Key Identifier other than 0, it MUST silently discard all authenticated chunks. If the endpoint has at least one endpoint pair shared key for the peer, it MUST use the key specified by the Shared Key Identifier if a key has been configured for that Shared Key Identifier. If no endpoint pair shared key has been configured for that Shared Key Identifier, all authenticated chunks MUST be silently discarded.

The receiver now performs the same calculation as described for the sender based on Figure 6. If the result of the calculation is the same as given in the HMAC field, all the chunks following the AUTH chunk are processed. If the field does not match the result of the calculation, all the chunks following the AUTH chunk MUST be silently discarded.

It should be noted that if the receiver wants to tear down an association in an authenticated way only, the handling of malformed packets should not result in tearing down the association.

An SCTP implementation has to maintain state for each SCTP association. In the following, we call this data structure the SCTP transmission control block (STCB).

When an endpoint requires COOKIE-ECHO chunks to be authenticated, some special procedures have to be followed because the reception of a COOKIE-ECHO chunk might result in the creation of an SCTP association. If a packet arrives containing an AUTH chunk as a first chunk, a COOKIE-ECHO chunk as the second chunk, and possibly more chunks after them, and the receiver does not have an STCB for that packet, then authentication is based on the contents of the COOKIE-ECHO chunk. In this situation, the receiver MUST authenticate the chunks in the packet by using the RANDOM parameters, CHUNKS parameters and HMAC_ALGO parameters obtained from the COOKIE-ECHO chunk, and possibly a local shared secret as inputs to the authentication procedure specified in Section 6.3. If authentication fails, then the packet is discarded. If the authentication is successful, the COOKIE-ECHO and all the chunks after the COOKIE-ECHO MUST be processed. If the receiver has an STCB, it MUST process the AUTH chunk as described above using the STCB from the existing association to authenticate the COOKIE-ECHO chunk and all the chunks after it.

If the receiver does not find an STCB for a packet containing an AUTH chunk as the first chunk and does not find a COOKIE-ECHO chunk as the second chunk, it MUST use the chunks after the AUTH chunk to look up an existing association. If no association is found, the packet MUST be considered as out of the blue. The out of the blue handling MUST be based on the packet without taking the AUTH chunk into account. If an association is found, it MUST process the AUTH chunk using the STCB from the existing association as described earlier.

Requiring ABORT chunks and COOKIE-ECHO chunks to be authenticated makes it impossible for an attacker to bring down or restart an association as long as the attacker does not know the association shared key. But it should also be noted that if an endpoint accepts ABORT chunks only in an authenticated way, it may take longer to detect that the peer is no longer available. If an endpoint accepts COOKIE-ECHO chunks only in an authenticated way, the restart procedure does not work, because the restarting endpoint most likely does not know the association shared key of the old association to be restarted. However, if the restarting endpoint does know the old association shared key, he can successfully send the COOKIE-ECHO chunk in a way that it is accepted by the peer by using this old association shared key for the packet containing the AUTH chunk. After this operation, both endpoints have to use the new association shared key.

If a server has an endpoint pair shared key with some clients, it can request the COOKIE_ECHO chunk to be authenticated and can ensure that only associations from clients with a correct endpoint pair shared key are accepted.

Furthermore, it is important that the cookie contained in an INIT-ACK chunk and in a COOKIE-ECHO chunk MUST NOT contain any endpoint pair shared keys.

7. Examples

This section gives examples of message exchanges for association setup.

The simplest way of using the extension described in this document is given by the following message exchange.

```

----- INIT[RANDOM; CHUNKS[*]; HMAC-ALGO] ----->
<----- INIT-ACK[RANDOM; CHUNKS[*]; HMAC-ALGO] -----
----- COOKIE-ECHO ----->
<----- COOKIE-ACK -----

```

Figure 7: auth_negotiation_flow

Please note that the CHUNKS parameter is optional in the INIT and INIT-ACK.

If the server wants to receive DATA chunks in an authenticated way, the following message exchange is possible:

```

----- INIT[RANDOM; CHUNKS[*]; HMAC-ALGO] ----->
<----- INIT-ACK[RANDOM; CHUNKS[ DATA ]; HMAC-ALGO] -----
----- COOKIE-ECHO; AUTH; DATA ----->
<----- COOKIE-ACK; SACK -----

```

Figure 8: data_chunk_auth_negotiation

Please note that if the endpoint pair shared key depends on the client and the server, and is only known by the upper layer, this message exchange requires an upper layer intervention between the processing of the COOKIE-ECHO chunk and the processing of the AUTH and DATA chunk at the server side. This intervention may be realized by a COMMUNICATION-UP notification followed by the presentation of the endpoint pair shared key by the upper layer to the SCTP stack, see for example Section 10 of RFC 4960 [RFC4960]. If this intervention is not possible due to limitations of the API (for example, the socket API), the server might discard the AUTH and DATA chunk, making a retransmission of the DATA chunk necessary. If the same endpoint pair shared key is used for multiple endpoints and does not depend on the client, this intervention might not be necessary.

8. IANA Considerations

This document (RFC 4895 [RFC4895]) is the reference for all registrations described in this section. All registrations need to be listed in the document available at [sctp-parameters]. The changes are described below.

8.1. A New Chunk Type

A chunk type for the AUTH chunk has been assigned by IANA. IANA has assigned the value (15), as given in Table 4. An additional line has been added in the "CHUNK TYPES" table of [sctp-parameters]:

CHUNK TYPES

ID Value -----	Chunk Type -----	Reference -----
15	Authentication Chunk (AUTH)	[RFC4895]

Figure 9: New Chunk Type

8.2. Three New Parameter Types

Parameter types have been assigned for the RANDOM, CHUNKS, and HMAC-ALGO parameter by IANA. The values are as given in Table 1. This required two modifications to the "CHUNK PARAMETER TYPES" tables in [sctp-parameters]: the first is the addition of three new lines to the "INIT Chunk Parameter Types" table:

Chunk Parameter Type	Value
-----	-----
Random	32770 (0x8002)
Chunk List	32771 (0x8003)
Requested HMAC Algorithm Parameter	32772 (0x8004)

Figure 10: Chunk Parameter Type

The second required change is the addition of the same three lines to the to the "INIT ACK Chunk Parameter Types" table.

8.3. A New Error Cause

An error cause for the Unsupported HMAC Identifier error cause has been assigned. The value (261) has been assigned as in Table 3. This requires an additional line of the "CAUSE CODES" table in [sctp-parameters]:

VALUE	CAUSE CODE	REFERENCE
-----	-----	-----
261 (0x0105)	Unsupported HMAC Identifier	[RFC4895]

Figure 11: New Cause Code

8.4. A New Table for HMAC Identifiers

HMAC Identifiers have to be maintained by IANA. Four initial values have been assigned by IANA as described in Table 2. This required a new table "HMAC IDENTIFIERS" in [sctp-parameters]:

HMAC Identifier	Message Digest Algorithm	REFERENCE
-----	-----	-----
0	Reserved	[RFC4895]
1	SHA-1	[RFC4895]
2	Reserved	[RFC4895]
3	SHA-256	[RFC4895]

Figure 12: HMAC ID List

For registering a new HMAC Identifier with IANA, in this table, a request has to be made to assign such a number. This number must be unique and a message digest algorithm usable with the HMAC defined in RFC 6151 [RFC6151] MUST be specified. The "Specification Required" policy of RFC 8126 [RFC8126] MUST be applied.

9. Security Considerations

Without using endpoint shared keys, this extension only protects against modification or injection of authenticated chunks by attackers who did not capture the initial handshake setting up the SCTP association.

If an endpoint pair shared key is used, even a true man in the middle cannot inject chunks, which are required to be authenticated, even if he intercepts the initial message exchange. The endpoint also knows that it is accepting authenticated chunks from a peer who knows the endpoint pair shared key.

The establishment of endpoint pair shared keys is out of the scope of this document. Other mechanisms can be used, like using TLS or manual configuration.

When an endpoint accepts COOKIE-ECHO chunks only in an authenticated way the restart procedure does not work. Neither an attacker nor a restarted endpoint not knowing the association shared key can perform an restart. However, if the association shared key is known, it is possible to restart the association.

Because SCTP already has a built-in mechanism that handles the reception of duplicated chunks, the presented solution makes use of this functionality and does not provide a method to avoid replay attacks by itself. Of course, this only works within each SCTP association. Therefore, a separate shared key is used for each SCTP association to handle replay attacks covering multiple SCTP associations.

Each endpoint presenting a list of more than one element in the HMAC-ALGO parameter must be prepared for the peer using the weakest algorithm listed.

When an endpoint pair uses non-NULL endpoint pair shared keys and one of the endpoints still accepts a NULL key, an attacker who captured the initial handshake can still inject or modify authenticated chunks by using the NULL key.

10. Acknowledgements

I would like to thank shweta K R for her review and invaluable comments.

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SOCKS Protocol Version 6
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Abstract

The SOCKS protocol is used primarily to proxy TCP connections to arbitrary destinations via the use of a proxy server. Under the latest version of the protocol (version 5), it takes 2 RTTs (or 3, if authentication is used) before data can flow between the client and the server.

This memo proposes SOCKS version 6, which reduces the number of RTTs used, takes full advantage of TCP Fast Open, and adds support for 0-RTT authentication.

Status of This Memo

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

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1. Introduction

Versions 4 and 5 [RFC1928] of the SOCKS protocol were developed two decades ago and are in widespread use for circuit level gateways or as circumvention tools, and enjoy wide support and usage from various software, such as web browsers, SSH clients, and proxifiers. However, their design needs an update in order to take advantage of the new features of transport protocols, such as TCP Fast Open [RFC7413], or to better assist newer transport protocols, such as MPTCP [RFC6824].

One of the main issues faced by SOCKS version 5 is that, when taking into account the TCP handshake, method negotiation, authentication, connection request and grant, it may take up to 5 RTTs for a data exchange to take place at the application layer. This is especially costly in networks with a large delay at the access layer, such as 3G, 4G, or satellite.

The desire to reduce the number of RTTs manifests itself in the design of newer security protocols. TLS version 1.3 [RFC8446] defines a zero round trip (0-RTT) handshake mode for connections if the client and server had previously communicated.

TCP Fast Open [RFC7413] is a TCP option that allows TCP to send data in the SYN and receive a response in the first ACK, and aims at obtaining a data response in one RTT. The SOCKS protocol needs to concern itself with at least two TFO deployment scenarios: First, when TFO is available end-to-end (at the client, at the proxy, and at the server); second, when TFO is active between the client and the proxy, but not at the server.

This document describes the SOCKS protocol version 6. The key improvements over SOCKS version 5 are:

- o The client sends as much information upfront as possible, and does not wait for the authentication process to conclude before requesting the creation of a socket.

- o The connection request also mimics the semantics of TCP Fast Open [RFC7413]. As part of the connection request, the client can supply the potential payload for the initial SYN that is sent out to the server.
- o The protocol can be extended via options without breaking backward-compatibility.
- o The protocol can leverage the aforementioned options to support 0-RTT authentication schemes.

1.1. Revision log

Typos and minor clarifications are not listed.

draft-11

- o Changed intended status to Standards Track
- o Renamed Vendor-specific option range to Experimental
- o Stack options:
 - * Fixed some instances where an unsupported option was indistinguishable from a case where the proxy couldn't or wouldn't honor it (offenders: Happy Eyeballs, IP Fragmentation, UDP Error, Port Parity)
 - * MPTCP: changed semantics w.r.t. TCP BIND: the absence of such an option SHOULD no longer lead the proxy to refuse MPTCP
 - * Port Parity: relaxed restrictions in case the client supplies a specific port

draft-10

- o Removed untrusted sessions
- o IP DF
- o UDP relay:
 - * Support ICMPv6 Too Big
 - * Shifted some fields in the error messages
 - * RTP support

draft-09

- o Revamped UDP relay
 - * Support for ICMP errors: host/net unreachable, TTL exceeded
 - * Datagrams can be sent over TCP
 - * Timeout for the receipt of the initial datagram
- o TTL stack option (intended use: traceroute)
- o Added the "Privacy Considerations" section
- o SOCKS-provided DNS: the proxy may provide a valid bind address and port

draft-08

- o Removed Address Resolution options
- o Happy Eyeballs options
- o DNS provided by SOCKS

draft-07

- o All fields are now aligned.
- o Eliminated version minors
- o Lots of changes to options
 - * 2-byte option kinds
 - * Flattened option kinds/types/reply codes; also renamed some options
 - * Socket options
 - + Proxies MUST always answer them (Clients can probe for support)
 - + MPTCP Options: expanded functionality ("please do/don't do MPTCP on my behalf")
 - + MPTCP Scheduler options removed

- + Listen Backlog options: code changed to 0x03
 - * Revamped Idempotence options
 - * Auth data options limited to one per method
 - o Authentication Reply: all authentication-related information is now in the options
 - * Authentication replies no longer have a field indicating the chosen auth. method
 - * Method that must proceed (or whereby authentication succeeded) indicated in options
 - * Username/password authentication: proxy now sends reply in option
 - o Removed requirements w.r.t. caching authentication methods by multihomed clients
 - o UDP: 8-byte association IDs
 - o Sessions
 - * The proxy is now free to terminate ongoing connections along with the session.
 - * The session-terminating request is not part of the session that it terminated.
 - o Address Resolution options
- draft-06
- o Session options
 - o Options now have a 2-byte length field.
 - o Stack options
 - * Stack options can no longer contain duplicate information.
 - * TFO: Better payload size semantics
 - * TOS: Added missing code field.
 - * MPTCP Scheduler options:

- + Removed support for round-robin
- + "Default" renamed to "Lowest latency first"
- * Listen Backlog options: now tied to sessions, instead of an authenticated user
- o Idempotence options
 - * Now used in the context of a session (no longer tied to an authenticated user)
 - * Idempotence options have a different codepoint: 0x05. (Was 0x04.)
 - * Clarified that implementations that support Idempotence Options must support all Idempotence Option Types.
 - * Shifted Idempotence Option Types by 1. (Makes implementation easier.)
- o Shrunk vendor-specific option range to 32 (down from 64).
- o Removed reference to dropping initial data. (It could no longer be done as of -05.)
- o Initial data size capped at 16KB.
- o Application data is never encrypted by SOCKS 6. (It can still be encrypted by the TLS layer under SOCKS.)
- o Messages now carry the total length of the options, rather than the number of options. Limited options length to 16KB.
- o Security Considerations
 - * Updated the section to reflect the smaller maximum message size.
 - * Added a subsection on resource exhaustion.

draft-05

- o Limited the "slow" authentication negotiations to one (and Authentication Replies to 2)
- o Revamped the handling of the first bytes in the application data stream

- * False starts are now recommended. (Added the "False Start" section.)
 - * Initial data is only available to clients willing to do "slow" authentication. Moved the "Initial data size" field from Requests to Authentication Method options.
 - * Initial data size capped at 2^{13} . Initial data can no longer be dropped by the proxy.
 - * The TFO option can hint at the desired SYN payload size.
 - o Request: clarified the meaning of the Address and Port fields.
 - o Better reverse TCP proxy support: optional listen backlog for TCP BIND
 - o TFO options can no longer be placed inside Operation Replies.
 - o IP TOS stack option
 - o Suggested a range for vendor-specific options.
 - o Revamped UDP functionality
 - * Now using fixed UDP ports
 - * DTLS support
 - o Stack options: renamed Proxy-Server leg to Proxy-Remote leg
- draft-04
- o Moved Token Expenditure Replies to the Authentication Reply.
 - o Shifted the Initial Data Size field in the Request, in order to make it easier to parse.

draft-03

- o Shifted some fields in the Operation Reply to make it easier to parse.
- o Added connection attempt timeout response code to Operation Replies.
- o Proxies send an additional Authentication Reply after the authentication phase. (Useful for token window advertisements.)

- o Renamed the section "Connection Requests" to "Requests"
- o Clarified the fact that proxies don't need to support any command in particular.
- o Added the section "TCP Fast Open on the Client-Proxy Leg"
- o Options:
 - * Added constants for option kinds
 - * Salt options removed, along with the relevant section from Security Considerations. (TLS 1.3 Makes AEAD mandatory.)
 - * Limited Authentication Data options to one per method.
 - * Relaxed proxy requirements with regard to handling multiple Authentication Data options. (When the client violates the above bullet point.)
 - * Removed interdependence between Authentication Method and Authentication Data options.
 - * Clients SHOULD omit advertising the "No authentication required" option. (Was MAY.)
 - * Idempotence options:
 - + Token Window Advertisements are now part of successful Authentication Replies (so that the proxy-server RTT has no impact on their timeliness).
 - + Proxies can't advertise token windows of size 0.
 - + Tweaked token expenditure response codes.
 - + Support no longer mandatory on the proxy side.
 - * Revamped Socket options
 - + Renamed Socket options to Stack options.
 - + Banned contradictory socket options.
 - + Added socket level for generic IP. Removed the "socket" socket level.
 - + Stack options no longer use option codes from `setsockopt()`.

- + Changed MPTCP Scheduler constants.

draft-02

- o Made support for Idempotence options mandatory for proxies.
- o Clarified what happens when proxies can not or will not issue tokens.
- o Limited token windows to $2^{31} - 1$.
- o Fixed definition of "less than" for tokens.
- o NOOP commands now trigger Operation Replies.
- o Renamed Authentication options to Authentication Data options.
- o Authentication Data options are no longer mandatory.
- o Authentication methods are now advertised via options.
- o Shifted some Request fields.
- o Option range for vendor-specific options.
- o Socket options.
- o Password authentication.
- o Salt options.

draft-01

- o Added this section.
- o Support for idempotent commands.
- o Removed version numbers from operation replies.
- o Request port number for SOCKS over TLS. Deprecate encryption/encapsulation within SOCKS.
- o Added Version Mismatch Replies.
- o Renamed the AUTH command to NOOP.
- o Shifted some fields to make requests and operation replies easier to parse.

2. Requirements language

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

3. Mode of operation

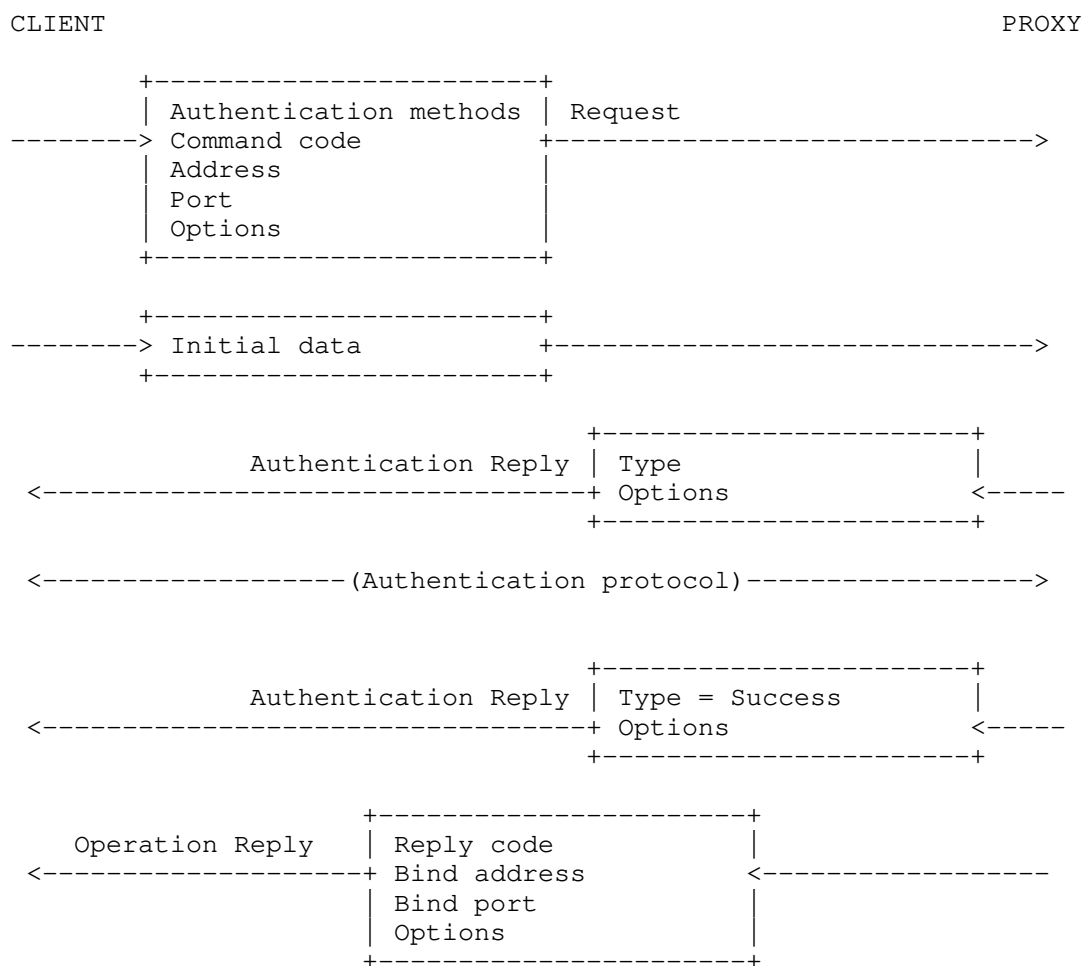


Figure 1: The SOCKS version 6 protocol message exchange

When a TCP-based client wishes to establish a connection to a server, it must open a TCP connection to the appropriate SOCKS port on the

SOCKS proxy. The client then enters a negotiation phase, by sending the request in Figure 1, that contains, in addition to fields present in SOCKS 5 [RFC1928], fields that facilitate low RTT usage and faster authentication negotiation.

Next, the server sends an authentication reply. If the request did not contain the necessary authentication information, the proxy indicates an authentication method that must proceed. This may trigger a longer authentication sequence that could include tokens for ulterior faster authentications. The part labeled "Authentication protocol" is specific to the authentication method employed and is not expected to be employed for every connection between a client and its proxy server. The authentication protocol typically takes up 1 RTT or more.

If the authentication is successful, an operation reply is generated by the proxy. It indicates whether the proxy was successful in creating the requested socket or not.

In the fast case, when authentication is properly set up, the proxy attempts to create the socket immediately after the receipt of the request, thus achieving an operational connection in one RTT (provided TFO functionality is available at the client, proxy, and server).

4. Requests

The client starts by sending a request to the proxy.

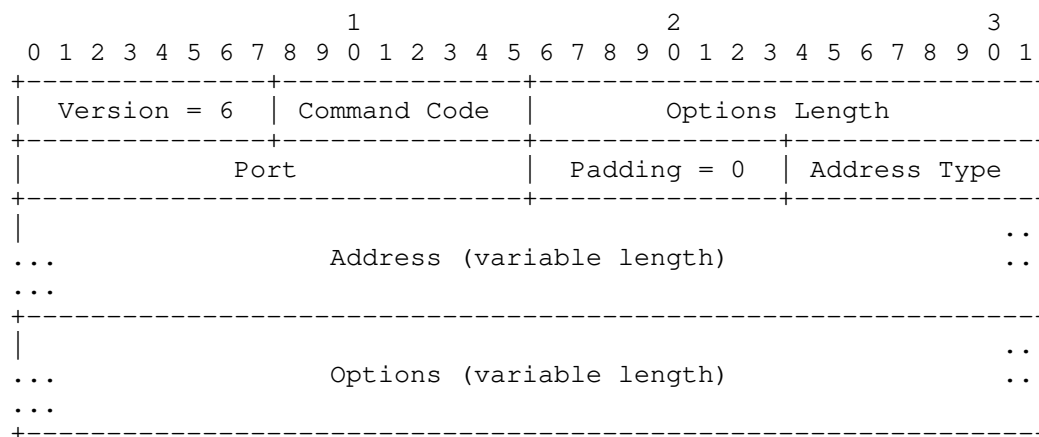


Figure 2: SOCKS 6 Request

- o Version: 6
- o Command Code:
 - * 0x00 NOOP: does nothing.
 - * 0x01 CONNECT: requests the establishment of a TCP connection. TFO MUST NOT be used unless explicitly requested.
 - * 0x02 BIND: requests the establishment of a TCP port binding.
 - * 0x03 UDP ASSOCIATE: requests a UDP port association.
- o Address Type:
 - * 0x01: IPv4
 - * 0x03: Domain Name
 - * 0x04: IPv6
- o Address: this field's format depends on the address type:
 - * IPv4: a 4-byte IPv4 address
 - * Domain Name: one byte that contains the length of the FQDN, followed by the FQDN itself. The string is not NUL-terminated, but padded by NUL characters, if needed.
 - * IPv6: a 16-byte IPv6 address
- o Port: the port in network byte order.
- o Padding: set to 0
- o Options Length: the total size of the SOCKS options that appear in the Options field. MUST NOT exceed 16KB.
- o Options: see Section 8.

The Address and Port fields have different meanings based on the Command Code:

- o NOOP: The fields have no meaning. The Address Type field MUST be either 0x01 (IPv4) or 0x04 (IPv6). The Address and Port fields MUST be 0.

- o CONNECT: The fields signify the address and port to which the client wishes to connect.
- o BIND, UDP ASSOCIATE: The fields indicate the desired bind address and port. If the client does not require a certain address, it can set the Address Type field to 0x01 (IPv4) or 0x04 (IPv6), and the Address field to 0. Likewise, if the client does not require a certain port, it can set the Port field to 0.

Clients can advertise their supported authentication methods by including an Authentication Method Advertisement option (see Section 8.2).

5. Version Mismatch Replies

Upon receipt of a request starting with a version number other than 6, the proxy sends the following response:

```

 0 1 2 3 4 5 6 7
+-----+
| Version = 6 |
+-----+

```

Figure 3: SOCKS 6 Version Mismatch Reply

- o Version: 6

A client **MUST** close the connection after receiving such a reply.

6. Authentication Replies

Upon receipt of a valid request, the proxy sends an Authentication Reply:

										1										2										3									
0	1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6	7	8	9	0	1								
+-----+-----+-----+-----+																																							
Version = 6										Type										Options Length																			
+-----+-----+-----+-----+																																							
																														..									
...										Options (variable length)																				..									
...																																							
+-----+-----+-----+-----+																																							

Figure 4: SOCKS 6 Authentication Reply

- o Version: 6
- o Type:
 - * 0x00: authentication successful.
 - * 0x01: authentication failed.
- o Options Length: the total size of the SOCKS options that appear in the Options field. MUST NOT exceed 16KB.
- o Options: see Section 8.

If the server signals that the authentication has failed and does not signal that any authentication negotiation can continue (via an Authentication Method Selection option), the client MUST close the connection.

The client and proxy begin a method-specific negotiation. During such negotiations, the proxy MAY supply information that allows the client to authenticate a future request using an Authentication Data option. Application data is not subject to any encryption negotiated during this phase. Descriptions of such negotiations are beyond the scope of this memo.

When the negotiation is complete (either successfully or unsuccessfully), the proxy sends a second Authentication Reply. The second Authentication Reply MUST NOT allow for further negotiations.

7. Operation Replies

After the authentication negotiations are complete, the proxy sends an Operation Reply:

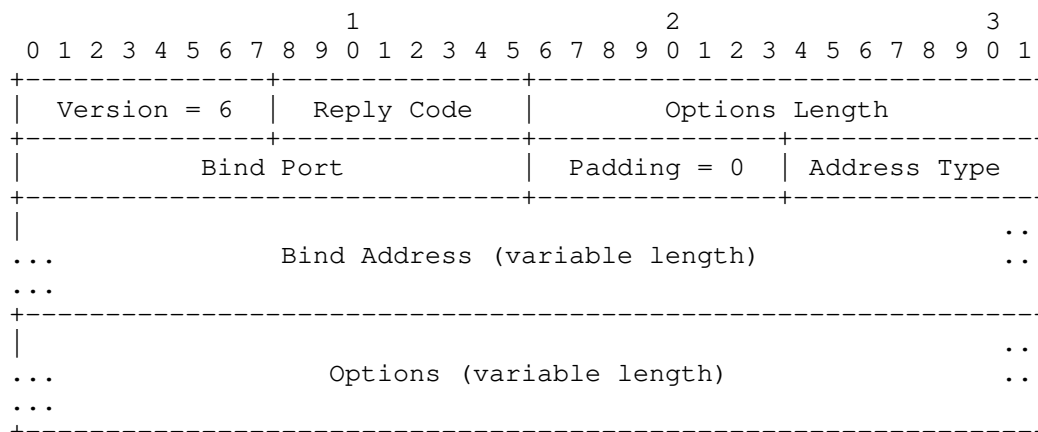


Figure 5: SOCKS 6 Operation Reply

- o Version: 6
- o Reply Code:
 - * 0x00: Success
 - * 0x01: General SOCKS server failure
 - * 0x02: Connection not allowed by ruleset
 - * 0x03: Network unreachable
 - * 0x04: Host unreachable
 - * 0x05: Connection refused
 - * 0x06: TTL expired
 - * 0x07: Command not supported
 - * 0x08: Address type not supported
 - * 0x09: Connection attempt timed out
- o Bind Port: the proxy bound port in network byte order.
- o Padding: set to 0
- o Address Type:

- * 0x01: IPv4
- * 0x03: Domain Name
- * 0x04: IPv6
- o Bind Address: the proxy bound address in the following format:
 - * IPv4: a 4-byte IPv4 address
 - * Domain Name: one byte that contains the length of the FQDN, followed by the FQDN itself. The string is not NUL-terminated, but padded by NUL characters, if needed.
 - * IPv6: a 16-byte IPv6 address
- o Options Length: the total size of the SOCKS options that appear in the Options field. MUST NOT exceed 16KB.
- o Options: see Section 8.

Proxy implementations MAY support any subset of the client commands listed in Section 4.

If the proxy returns a reply code other than "Success", the client MUST close the connection.

If the client issued an NOOP command, the client MUST close the connection after receiving the Operation Reply.

7.1. Handling CONNECT

In case the client has issued a CONNECT request, data can now pass.

7.2. Handling BIND

In case the client has issued a BIND request, it must wait for a second Operation reply from the proxy, which signifies that a host has connected to the bound port. The Bind Address and Bind Port fields contain the address and port of the connecting host. Afterwards, application data may pass.

7.3. Handling UDP ASSOCIATE

Proxies offering UDP functionality may be configured with a UDP port used for relaying UDP datagrams to and from the client, and/or a port used for relaying datagrams over DTLS.

Following a successful Operation Reply, the client and the proxy begin exchanging messages with the following header:

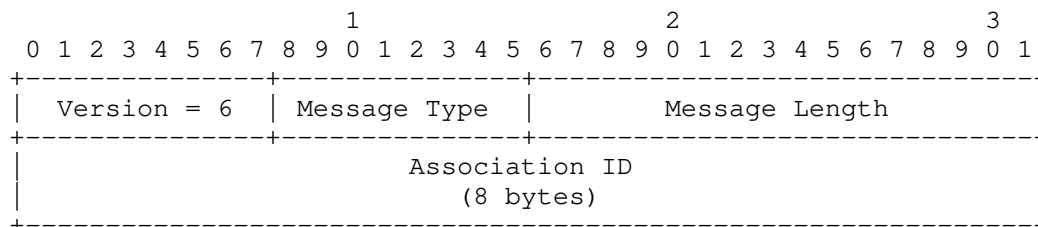


Figure 6: UDP Association Header

o Message Type:

- * 0x01: Association Initialization
- * 0x02: Association Confirmation
- * 0x03: Datagram
- * 0x04: Error

o Message Length: the total length of the message

o Association ID: the identifier of the UDP association

First, the proxy picks an Association ID sends a an Association Initialization message:

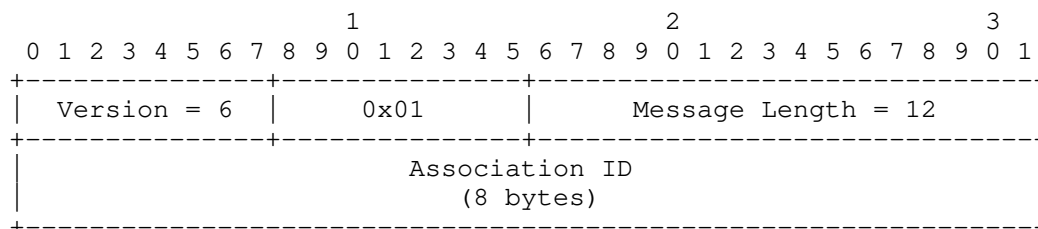


Figure 7: UDP Association Initialization

Proxy implementations SHOULD generate Association IDs randomly or pseudo-randomly.

Clients may start sending datagrams to the proxy either:

- o over the TCP connection,
- o in plaintext, using the proxy's configured UDP port(s), or
- o over an established DTLS session.

A client's datagrams are prefixed by a Datagram Header, indicating the remote host's address and port:

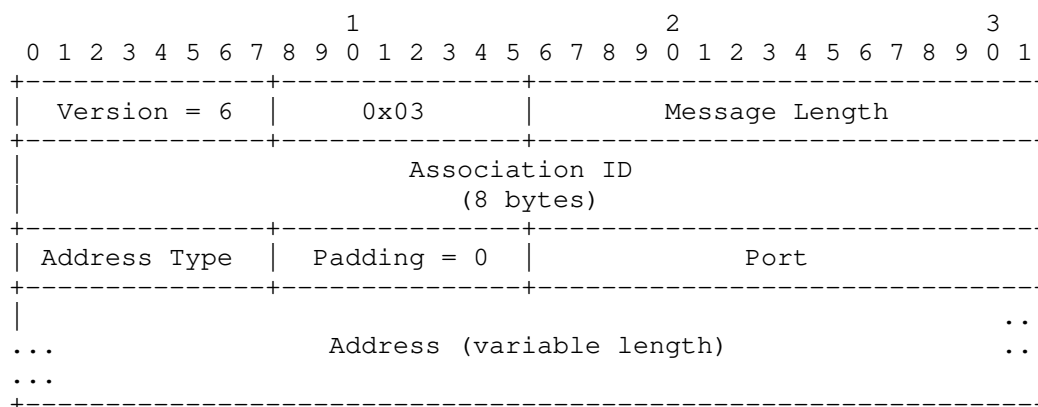


Figure 8: Datagram Header

- o Version: 0x06
- o Association ID: the identifier of the UDP association
- o Address Type:
 - * 0x01: IPv4
 - * 0x03: Domain Name
 - * 0x04: IPv6
- o Address: this field's format depends on the address type:
 - * IPv4: a 4-byte IPv4 address
 - * Domain Name: one byte that contains the length of the FQDN, followed by the FQDN itself. The string is not NUL-terminated.
 - * IPv6: a 16-byte IPv6 address

- o Port: the port in network byte order.

Datagrams sent over UDP MAY be padded with arbitrary data (i. e., the Message Length MAY be smaller than the actual UDP/DTLS payload). Client and proxy implementations MUST ignore the padding. If the Message Length is larger than the size of the UDP or DTLS payload, the message MUST be silently ignored.

Following the receipt of the first datagram from the client, the proxy makes a one-way mapping between the Association ID and:

- o the TCP connection, if it was received over TCP, or
- o the 5-tuple of the UDP conversation, if the datagram was received over plain UDP, or
- o the DTLS connection, if the datagram was received over DTLS. The DTLS connection is identified either by its 5-tuple, or some other mechanism, like [I-D.ietf-tls-dtls-connection-id].

The proxy SHOULD close the TCP connection if the initial datagram is not received after a timeout.

Further datagrams carrying the same Association ID, but not matching the established mapping, are silently dropped.

The proxy then sends an UDP Association Confirmation message over the TCP connection with the client:

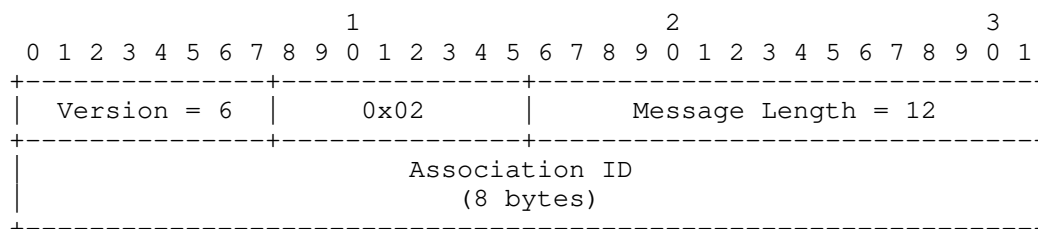


Figure 9: UDP Association Confirmation

Following the confirmation message, UDP packets bound for the proxy's bind address and port are relayed to the client, also prefixed by a Datagram Header.

The UDP association remains active for as long as the TCP connection between the client and the proxy is kept open.

7.3.1. Proxying UDP servers

Under some circumstances (e.g. when hosting a server), the SOCKS client expects the remote host to send UDP datagrams first. As such, the SOCKS client must trigger a UDP Association Confirmation without having the proxy relay any datagrams on its behalf.

To that end, it sends an empty datagram prefixed by a Datagram Header with an IP address and port consisting of zeroes. If it is using UDP, the client **SHOULD** resend the empty datagram if an UDP Association Confirmation is not received after a timeout.

7.3.2. Proxying multicast traffic

The use of multicast addresses is permitted for UDP traffic only.

7.3.3. Reporting ICMP Errors

If a client has opted in (see Section 8.1.8), the proxy **MAY** relay information contained in some ICMP Error packets. The message format is as follows:

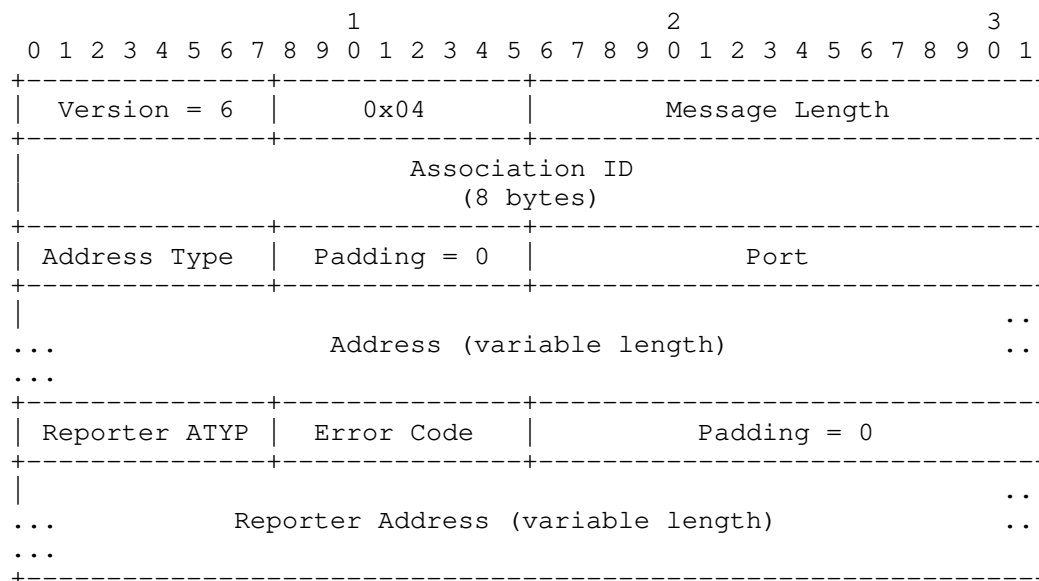


Figure 10: Datagram Error Message

- o Address: The destination address of the IP header contained in the ICMP payload

- o Address Type: Either 0x01 (IPv4) or 0x04 (IPv6)
- o Port: The destination port of the UDP header contained in the ICMP payload
- o Reporter Address: The IP address of the host that issued the ICMP error
- o Reporter Address Type (ATYP): Either 0x01 (IPv4) or 0x04 (IPv6)
- o Error code:
 - * 0x01: Network unreachable
 - * 0x02: Host unreachable
 - * 0x03: TTL expired
 - * 0x04: Datagram too big (IPv6 only)

It is possible for ICMP Error packets to be spurious, and not be related to any UDP packet that was sent out. The proxy is not required to check the validity of ICMP Error packets before reporting them to the client.

Clients **MUST NOT** send Datagram Error messages to the proxy. Proxies **MUST NOT** send Error messages unless the clients have opted in.

8. SOCKS Options

SOCKS options have the following format:

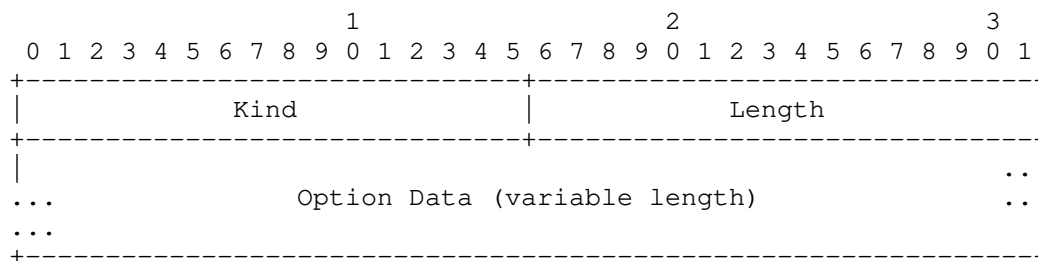


Figure 11: SOCKS 6 Option

- o Kind: Allocated by IANA. (See Section 15.)
- o Length: The total length of the option. **MUST** be a multiple of 4.

- o Option Data: The contents are specific to each option kind.

Unless otherwise noted, client and proxy implementations MAY omit supporting any of the options described in this document. Upon encountering an unsupported option, a SOCKS endpoint MUST silently ignore it.

8.1. Stack options

Stack options can be used by clients to alter the behavior of the protocols on top of which SOCKS is running, as well the protocols used by the proxy to communicate with the remote host (i.e. IP, TCP, UDP). A Stack option can affect either the proxy's protocol on the client-proxy leg or on the proxy-remote leg. Clients can only place Stack options inside SOCKS Requests.

Proxies MAY choose not to honor any Stack options sent by the client.

Proxies include Stack options in their Operation Replies to signal their behavior, and MUST do so for every supported Stack option sent by the client. Said options MAY also be unsolicited, i. e. the proxy MAY send them to signal behavior that was not explicitly requested by the client.

If a particular Stack option is unsupported, the proxy MUST silently ignore it.

In case of UDP ASSOCIATE, the stack options refer to the UDP traffic relayed by the proxy.

Stack options that are part of the same message MUST NOT contradict one another or contain duplicate information.

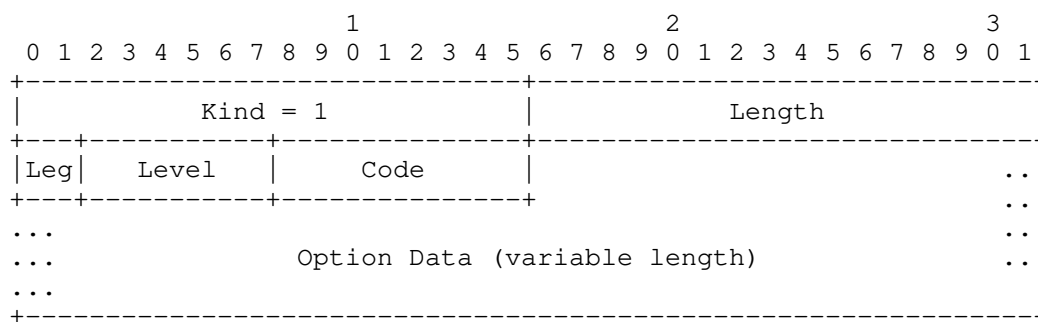


Figure 12: Stack Option

- o Leg:
 - * 1: Client-Proxy Leg
 - * 2: Proxy-Remote Leg
 - * 3: Both Legs
- o Level:
 - * 1: IP: options that apply to either IPv4 or IPv6
 - * 2: IPv4
 - * 3: IPv6
 - * 4: TCP
 - * 5: UDP
- o Code: Option code
- o Option Data: Option-specific data

8.1.1. IP TOS options

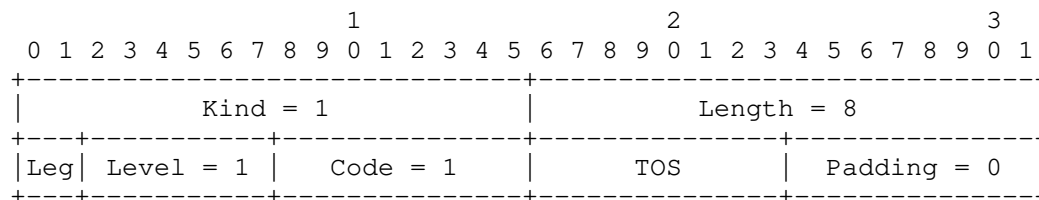


Figure 13: IP TOS Option

- o TOS: The IP TOS code

The client can use IP TOS options to request that the proxy use a certain value for the IP TOS field. Likewise, the proxy can use IP TOS options to advertise the TOS values being used.

8.1.2. Happy Eyeballs options

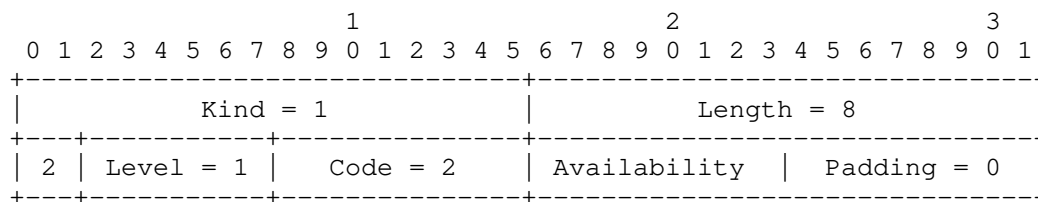


Figure 14: Happy Eyeballs Option

o Availability:

- * 0x01: Happy Eyeballs is not desired (client) or was not performed (proxy)
- * 0x02: Happy Eyeballs is desired (client) or was attempted (proxy)

This memo provides enough features for clients to implement a mechanism analogous to Happy Eyeballs [RFC8305] over SOCKS. However, when the delay between the client and the proxy, or the proxy's vantage point, is high, doing so can become impractical or inefficient.

In such cases, the client can instruct the proxy to employ the Happy Eyeballs technique on its behalf when connecting to a remote host.

The client **MUST** supply a Domain Name as part of its Request. Otherwise, the proxy **MUST** silently ignore the option.

TODO: Figure out which knobs to include.

8.1.3. TTL options

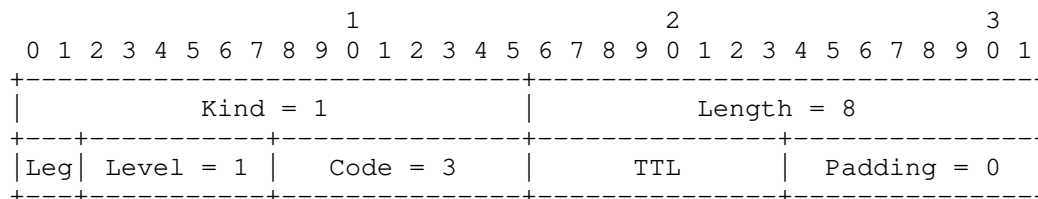


Figure 15: IP TTL Option

o TTL: The IP TTL or Hop Limit

8.1.4. No Fragmentation options

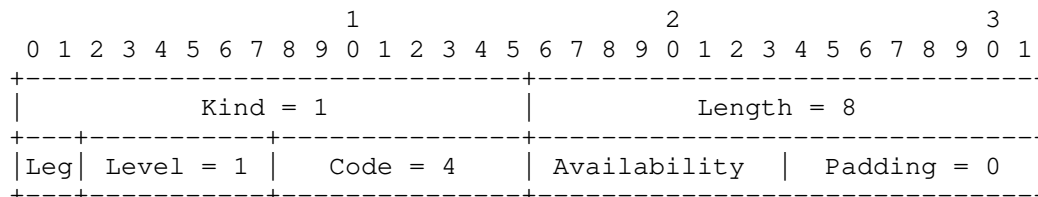


Figure 16: No Fragmentation Option

o Availability:

- * 0x01: IP fragmentation is allowed (client) or the lack thereof is not enforced (proxy)
- * 0x02: IP fragmentation is not desired (client) or avoidance of fragmentation is enforced (proxy)

A No Fragmentation option can be used to instruct the proxy to avoid IP fragmentation. In the case of IPv4, this also entails setting the DF bit on outgoing packets.

8.1.5. TFO options

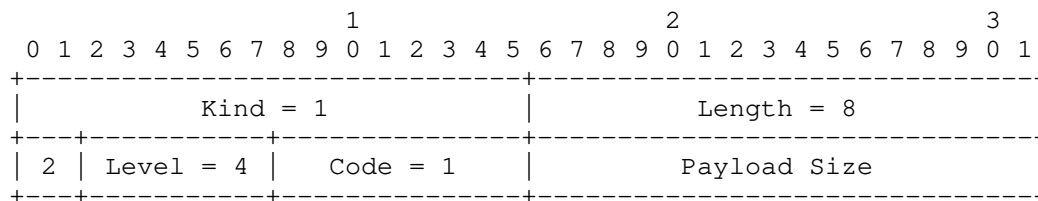


Figure 17: TFO Option

- o Payload Size: The desired payload size of the TFO SYN. Ignored in case of a BIND command.

If a SOCKS Request contains a TFO option, the proxy SHOULD attempt to use TFO in case of a CONNECT command, or accept TFO in case of a BIND command. Otherwise, the proxy MUST NOT attempt to use TFO in case of a CONNECT command, or accept TFO in case of a BIND command.

In case of a CONNECT command, the client can indicate the desired payload size of the SYN. If the field is 0, the proxy can use an

arbitrary payload size. If the field is non-zero, the proxy MUST NOT use a payload size larger than the one indicated. The proxy MAY use a smaller payload size than the one indicated.

8.1.6. Multipath options

In case of a CONNECT or BIND command, the client can inform the proxy whether MPTCP is desired on the proxy-remote leg by sending a Multipath option.

Conversely, the proxy can use a Multipath option to convey the following information:

- o whether or not the connection uses MPTCP or not, when replying to a CONNECT command, or in the second Operation reply to a BIND command, or
- o whether an MPTCP connection will be accepted, when first replying to a BIND command.

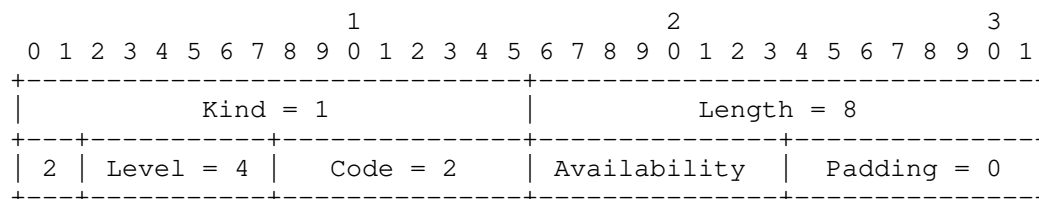


Figure 18: Multipath Option

- o Availability:
 - * 0x01: MPTCP is not desired (client) or available (proxy)
 - * 0x02: MPTCP is desired (client) or available (proxy)

In the absence of such an option, the proxy SHOULD NOT enable MPTCP for CONNECT commands.

8.1.7. Listen Backlog options

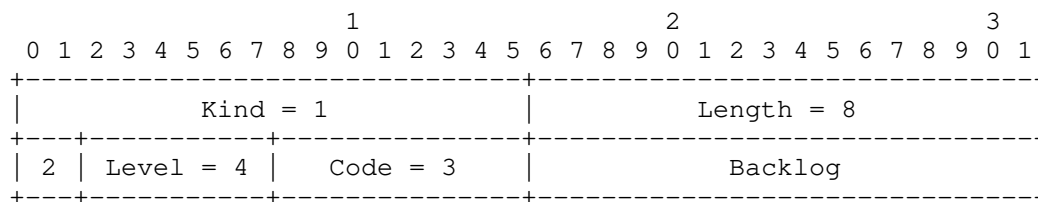


Figure 19: Listen Backlog Option

- o Backlog: The length of the listen backlog.

The default behavior of the BIND does not allow a client to simultaneously handle multiple connections to the same bind address. A client can alter BIND's behavior by adding a TCP Listen Backlog Option to a BIND Request, provided that the Request is part of a Session.

In response, the proxy sends a TCP Listen Backlog Option as part of the Operation Reply, with the Backlog field signaling the actual backlog used. The proxy SHOULD NOT use a backlog longer than requested.

Following the successful negotiation of a backlog, the proxy listens for incoming connections for as long as the initial connection stays open. The initial connection is not used to relay data between the client and a remote host.

To accept connections, the client issues further BIND Requests using the bind address and port supplied by the proxy in the initial Operation Reply. Said BIND requests must belong to the same Session as the original Request.

If no backlog is issued, the proxy signals a backlog length of 0, and BIND's behavior remains unaffected.

8.1.8. UDP Error options

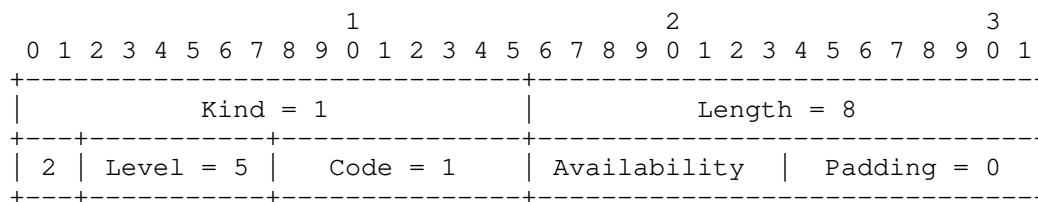


Figure 20: UDP Error Option

o Availability:

- * 0x01: Error reporting is not desired (client) or will not be performed (proxy)
- * 0x02: Error reporting is desired (client) or will be performed (proxy)

Clients can use this option to turn on error reporting for a particular UDP association. See Section 7.3.3.

8.1.9. Port Parity options

The RTP specification [RFC3550] recommends running the protocol on consecutive UDP ports, where the even port is the lower of the two.

SOCKS clients can specify the desired port parity when issuing a UDP ASSOCIATE command, and request that the port's counterpart be reserved.

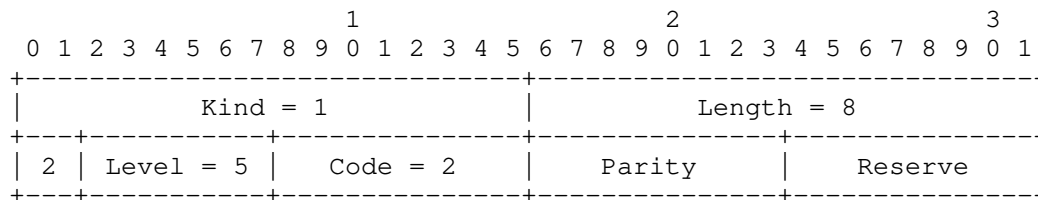


Figure 21: Port Parity Option

o Parity:

- * 0x00: No particular parity
- * 0x01: Even

- * 0x02: Odd
- o Reserve: whether or not to reserve the port's counterpart
- * 0x00: Don't reserve
- * 0x01: Reserve

If the UDP ASSOCIATE request does not have the Port field set to 0 (indicating that an arbitrary port can be chosen), the proxy MUST ignore the suggested parity.

A port's counterpart is determined as follows:

- o for even ports, it is the next higher port and
- o for odd ports, it is the next lower port.

If the proxy can not or will not comply with the requested parity, it also does not reserve the allocated port's counterpart.

Port reservations are in place until either:

- o the original association ends, or
- o an association involving the reserved port is made.

An association involving a reserved port can only be made if a client explicitly requests said port. Further, if the original association is part of a session (see Section 8.4), the reserved port can only be claimed from within the same session.

8.2. Authentication Method options

A client that is willing to go through the authentication phase MUST include an Authentication Method Advertisement option in its Request. In case of a CONNECT Request, the option is also used to specify the amount of initial data supplied before any method-specific authentication negotiations take place.

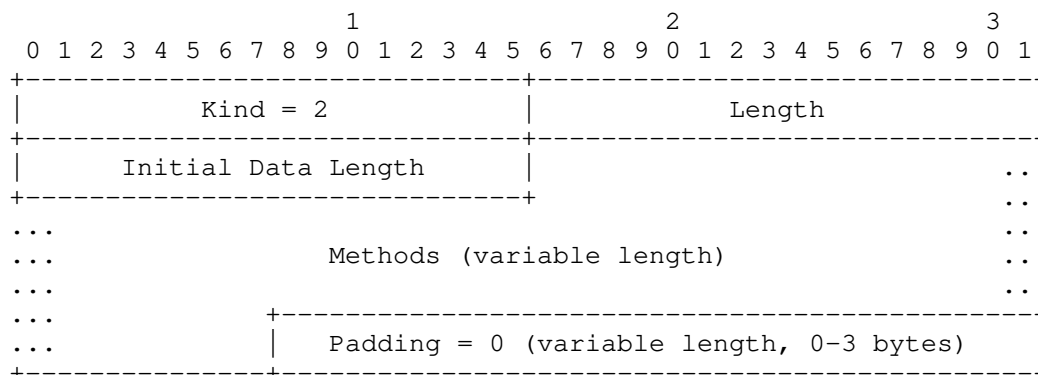


Figure 22: Authentication Method Advertisement Option

- o Initial Data Size: A two-byte number in network byte order. In case of CONNECT, this is the number of bytes of initial data that are supplied by the client immediately following the Request. This number MUST NOT be larger than 2^{14} .
- o Methods: One byte per advertised method. Method numbers are assigned by IANA.
- o Padding: A minimally-sized sequence of zeroes, such that the option length is a multiple of 4. Note that 0 coincides with the value for "No Authentication Required".

Clients MUST support the "No authentication required" method. Clients SHOULD omit advertising the "No authentication required" option.

The proxy indicates which authentication method must proceed by sending an Authentication Method Selection option in the corresponding Authentication Reply:

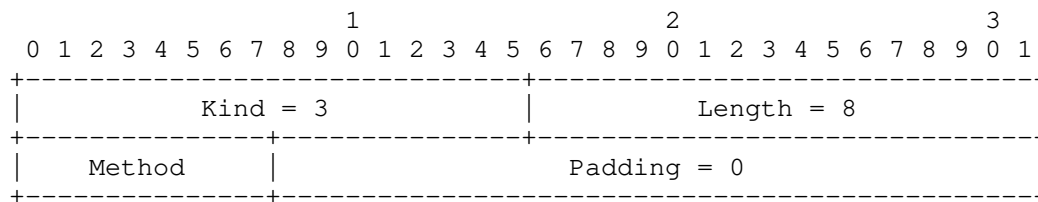


Figure 23: Authentication Method Selection Option

- o Method: The selected method.

If the proxy selects "No Acceptable Methods", the client MUST close the connection.

If authentication is successful via some other means, or not required at all, the proxy silently ignores the Authentication Method Advertisement option.

8.3. Authentication Data options

Authentication Data options carry method-specific authentication data. They can be part of SOCKS Requests and Authentication Replies.

Authentication Data options have the following format:

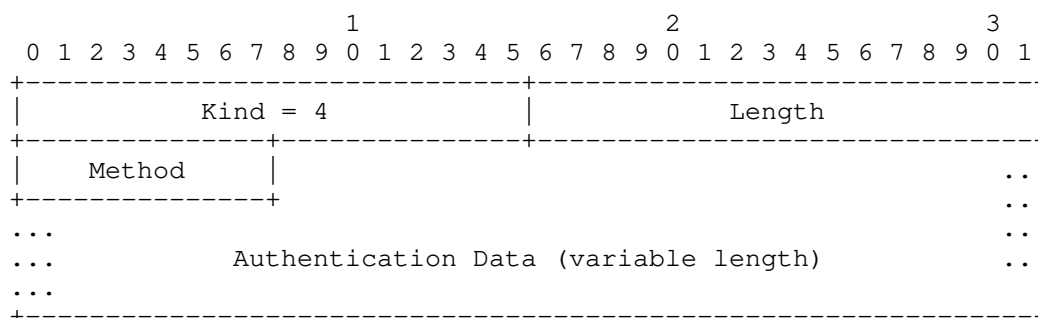


Figure 24: Authentication Data Option

- o Method: The number of the authentication method. These numbers are assigned by IANA.
- o Authentication Data: The contents are specific to each method.

Clients MUST only place one Authentication Data option per authentication method.

8.4. Session options

Clients and proxies can establish SOCKS sessions, which span one or more Requests. All session-related negotiations are done via Session Options, which are placed in Requests and Authentication Replies by the client and, respectively, by the proxy.

Client and proxy implementations MUST either support all Session Option Types, or none.

8.4.1. Session initiation

A client can initiate a session by sending a Session Request Option:

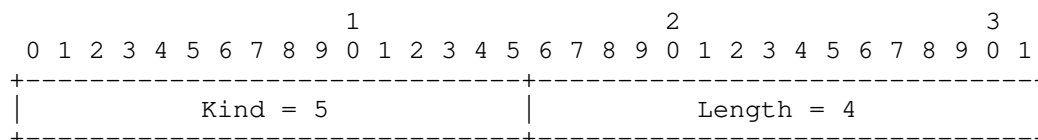


Figure 25: Session Request Option

The proxy then replies with a Session ID Option in the successful Operation Reply:

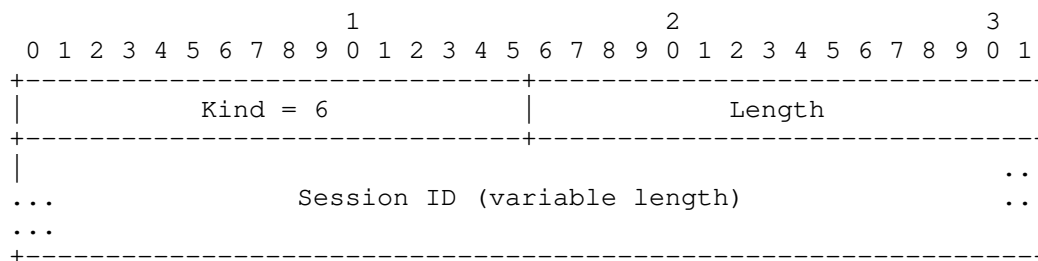


Figure 26: Session ID Option

- o Session ID: An opaque sequence of bytes specific to the session. The size MUST be a multiple of 4. MUST NOT be empty.

The Session ID serves to identify the session and is opaque to the client.

The credentials, or lack thereof, used to initiate the session are tied to the session.

The SOCKS Request that initiated the session is considered part of the session. A client MUST NOT attempt to initiate a session from within a different session.

If the proxy can not or will not honor the Session Request, it does so silently.

8.4.2. Further SOCKS Requests

Any further SOCKS Requests that are part of the session MUST include a Session ID Option (as seen in Figure 26). The proxy MUST silently ignore any authentication attempt in the Request, and MUST NOT require any authentication.

The proxy then replies by placing a Session OK option in the successful Authentication Reply:

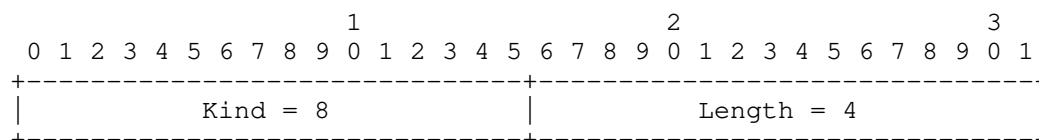


Figure 27: Session OK Option

If the Session ID is invalid, the first Authentication Reply MUST signal that authentication failed and can not continue (by setting the Type field to 0x01). Further, it SHALL contain a Session Invalid option:

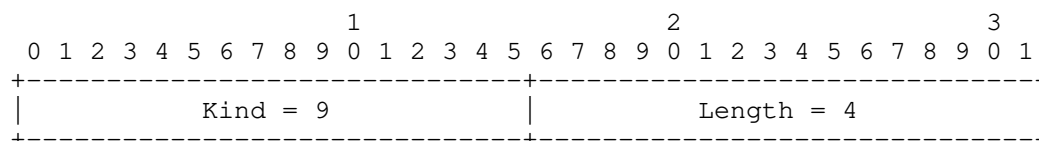


Figure 28: Session Invalid Option

8.4.3. Tearing down the session

Proxies can, at their discretion, tear down a session and free all associated state. Proxy implementations SHOULD feature a timeout mechanism that destroys sessions after a period of inactivity. When a session is terminated, the proxy MAY close all connections associated with said session.

Clients can signal that a session is no longer needed, and can be torn down, by sending a Session Teardown option in addition to the Session ID option:

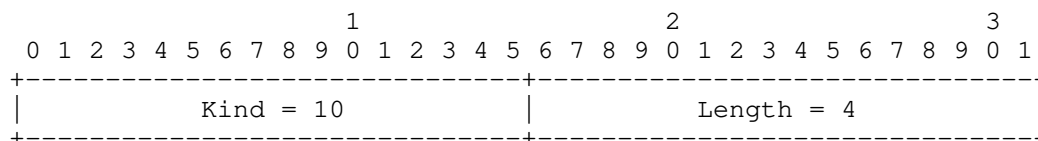


Figure 29: Session Teardown Option

After sending such an option, the client **MUST** assume that the session is no longer valid. The proxy **MUST** treat the session-terminating request as if it were not part of any session.

8.5. Idempotence options

To protect against duplicate SOCKS Requests, clients can request, and then spend, idempotence tokens. A token can only be spent on a single SOCKS request.

Tokens are 4-byte unsigned integers in a modular 4-byte space. Therefore, if x and y are tokens, x is less than y if $0 < (y - x) < 2^{31}$ in unsigned 32-bit arithmetic.

Proxies grant contiguous ranges of tokens called token windows. Token windows are defined by their base (the first token in the range) and size.

All token-related operations are done via Idempotence options.

Idempotence options are only valid in the context of a SOCKS Session. If a SOCKS Request is not part of a Session (either by supplying a valid Session ID or successfully initiating one via a Session Request), the proxy **MUST** silently ignore any Idempotence options.

Token windows are tracked by the proxy on a per-session basis. There can be at most one token window for every session and its tokens can only be spent from within said session.

Client and proxy implementations **MUST** either support all Idempotence Option Types, or none.

8.5.1. Requesting a token window

A client can obtain a window of tokens by sending an Idempotence Request option as part of a SOCKS Request:

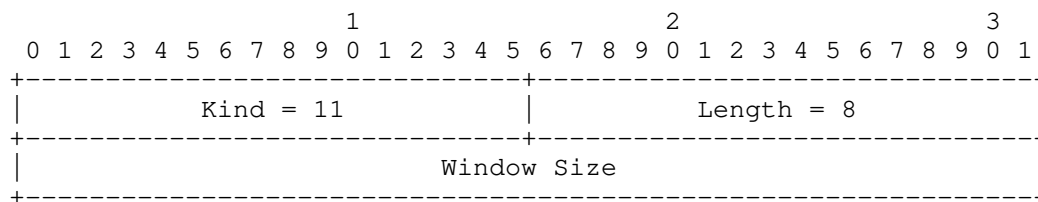


Figure 30: Token Request

- o Window Size: The requested window size.

Once a token window is issued, the proxy MUST include an Idempotence Window option in all subsequent successful Authentication Replies:

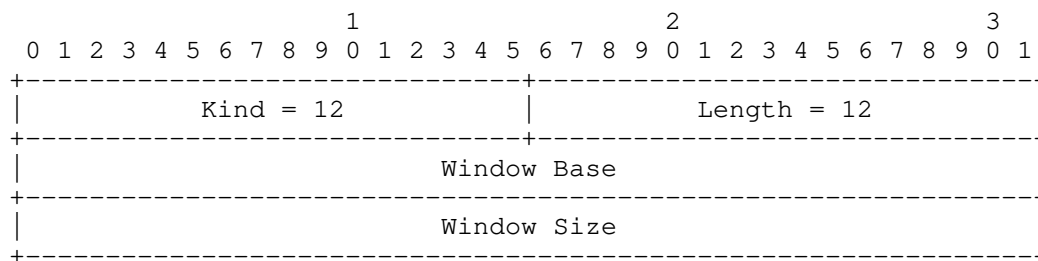


Figure 31: Idempotence Window

- o Window Base: The first token in the window.
- o Window Size: The window size. This value MAY differ from the requested window size. Window sizes MUST be less than 2^{31} . Window sizes MUST NOT be 0.

If no token window is issued, the proxy MUST silently ignore the Token Request. If there is already a token window associated with the session, the proxy MUST NOT issue a new window.

8.5.2. Spending a token

The client can attempt to spend a token by including a Idempotence Expenditure option in its SOCKS request:

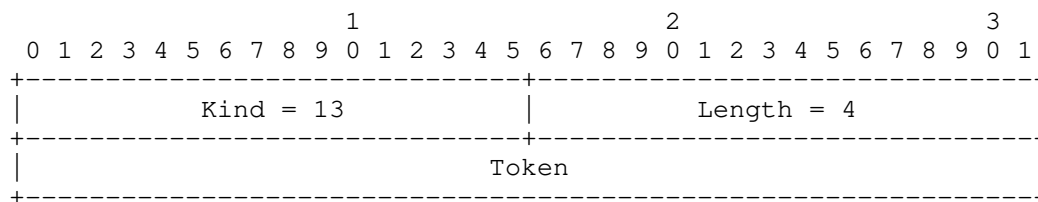


Figure 32: Idempotence Expenditure

- o Kind: 13 (Idempotence Expenditure option)
- o Length: 8
- o Token: The token being spent.

Clients SHOULD prioritize spending the smaller tokens.

The proxy responds by sending either an Idempotence Accepted or Rejected option as part of the Authentication Reply:

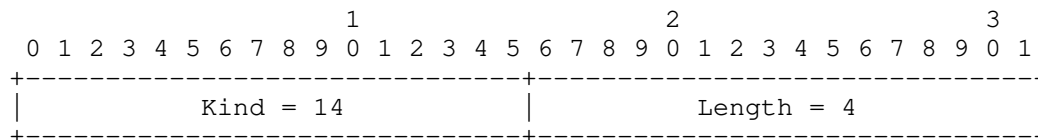


Figure 33: Idempotence Accepted

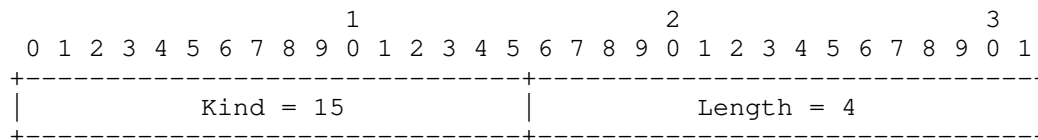


Figure 34: Idempotence Rejected

If eligible, the token is spent before attempting to honor the Request. If the token is not eligible for spending, the Authentication Reply MUST indicate failure.

8.5.3. Shifting windows

Windows can be shifted (i. e. have their base increased, while retaining their size) unilaterally by the proxy.

Proxy implementations SHOULD shift the window: * as soon as the lowest-order token in the window is spent and * when a sufficiently high-order token is spent.

Proxy implementations SHOULD NOT shift the window's base beyond the highest unspent token.

8.5.4. Out-of-order Window Advertisements

Even though the proxy increases the window's base monotonically, there is no mechanism whereby a SOCKS client can receive the Token Window Advertisements in order. As such, clients SHOULD disregard Token Window Advertisements with a Window Base less than the previously known value.

9. Username/Password Authentication

Username/Password authentication is carried out as in [RFC1929].

Clients can also attempt to authenticate by placing the Username/Password request in an Authentication Data Option.

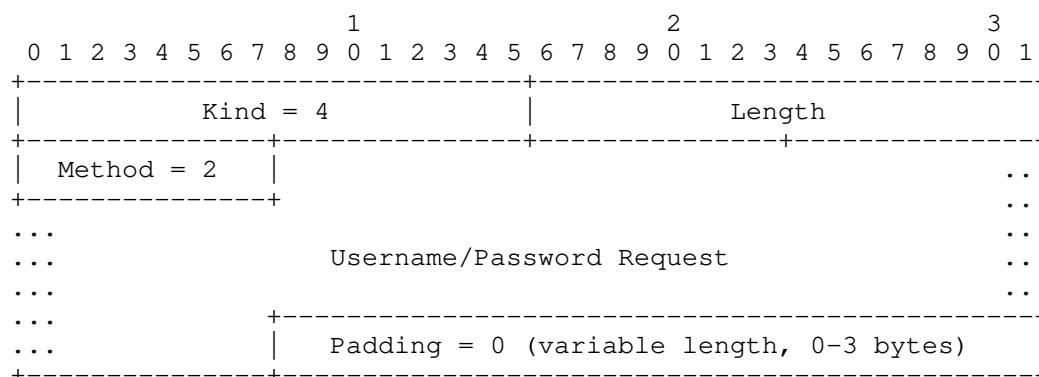


Figure 35: Password authentication via a SOCKS Option

- o Username/Password Request: The Username/Password Request, as described in [RFC1929].

Proxies reply by including a Authentication Data Option in the next Authentication Reply which contains the Username/Password reply:

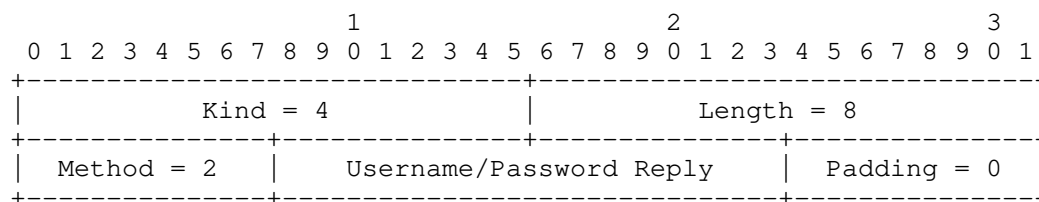


Figure 36: Reply to password authentication via a SOCKS Option

- o Username/Password Reply: The Username/Password Reply, as described in [RFC1929].

10. TCP Fast Open on the Client-Proxy Leg

TFO breaks TCP semantics, causing replays of the data in the SYN's payload under certain rare circumstances [RFC7413]. A replayed SOCKS Request could itself result in a replayed connection on behalf of the client.

As such, client implementations SHOULD NOT use TFO on the client-proxy leg unless:

- o The protocol running on top of SOCKS tolerates the risks of TFO, or
- o The SYN's payload does not contain any application data (so that no data is replayed to the server, even though duplicate connections are still possible), or
- o The client uses Idempotence Options, making replays very unlikely, or
- o SOCKS is running on top of TLS and Early Data is not used.

11. False Starts

In case of CONNECT Requests, the client MAY start sending application data as soon as possible, as long as doing so does not incur the risk of breaking the SOCKS protocol.

Clients must work around the authentication phase by doing any of the following:

- o If the Request does not contain an Authentication Method Advertisement option, the authentication phase is guaranteed not to happen. In this case, application data MAY be sent immediately after the Request.
- o Application data MAY be sent immediately after receiving an Authentication Reply indicating success.
- o When performing a method-specific authentication sequence, application data MAY be sent immediately after the last client message.

12. DNS provided by SOCKS

Clients may require information typically obtained from DNS servers, albeit from the proxy's vantage point.

While the CONNECT command can work with domain names, some clients' workflows require that addresses be resolved as a separate step prior to connecting. Moreover, the SOCKS Datagram Header, as described in Section 7.3, can be reduced in size by providing the resolved destination IP address, rather than the FQDN.

Emerging techniques may also make use of DNS to deliver server-specific information to clients. For example, Encrypted SNI [I-D.ietf-tls-esni] relies on DNS to publish encryption keys.

Proxy implementations MAY provide a default plaintext DNS service. A client looking to make use of it issues a CONNECT Request to IP address 0.0.0.0 or 0:0:0:0:0:0:0:0 on port 53. Following successful authentication, the Operation Reply MAY indicate an unspecified bind address (0.0.0.0 or ::) and port (0). The client and proxy then behave as per [RFC7766].

The service itself can be provided directly by the proxy daemon, or by proxying the client's request to a pre-configured DNS server.

If the proxy does not implement such functionality, it MAY return an error code signaling "Connection refused".

13. Security Considerations

13.1. Large requests

Given the format of the request message, a malicious client could craft a request that is in excess of 16 KB and proxies could be prone to DDoS attacks.

To mitigate such attacks, proxy implementations SHOULD be able to incrementally parse the requests. Proxies MAY close the connection to the client if:

- o the request is not fully received after a certain timeout, or
- o the number of options or their size exceeds an imposed hard cap.

13.2. Replay attacks

In TLS 1.3, early data (which is likely to contain a full SOCKS request) is prone to replay attacks.

While Token Expenditure options can be used to mitigate replay attacks, anything prior to the initial Token Request is still vulnerable. As such, client implementations SHOULD NOT make use of TLS early data unless the Request attempts to spend a token.

13.3. Resource exhaustion

Malicious clients can issue a large number of Session Requests, forcing the proxy to keep large amounts of state.

To mitigate this, the proxy MAY implement policies restricting the number of concurrent sessions on a per-IP or per-user basis, or barring unauthenticated clients from establishing sessions.

14. Privacy Considerations

The timing of Operation Replies can reveal some information about a proxy's recent usage:

- o The DNS resolver used by the proxy may cache the answer to recent queries. As such, subsequent connection attempts to the same hostname are likely to be slightly faster, even if requested by different clients.
- o Likewise, the proxy's OS typically caches TFO cookies. Repeated TFO connection attempts tend to be sped up, regardless of the client.

15. IANA Considerations

This document requests that IANA allocate 2-byte option kinds for SOCKS 6 options. Further, this document requests the following option kinds:

- o Unassigned: 0

- o Stack: 1
- o Authentication Method Advertisement: 2
- o Authentication Method Selection: 3
- o Authentication Data: 4
- o Session Request: 5
- o Session ID: 6
- o Session OK: 8
- o Session Invalid: 9
- o Session Teardown: 10
- o Idempotence Request: 11
- o Idempotence Window: 12
- o Idempotence Expenditure: 13
- o Idempotence Accepted: 14
- o Idempotence Rejected: 15
- o Resolution Request: 16
- o IPv4 Resolution: 17
- o IPv6 Resolution: 18
- o Experimental: 64512-0xFFFF

This document also requests that IANA allocate a TCP and UDP port for SOCKS over TLS and DTLS, respectively.

16. Acknowledgments

The protocol described in this draft builds upon and is a direct continuation of SOCKS 5 [RFC1928].

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Artificial Intelligence (AI) based ECN adaptive reconfiguration for
datacenter networks
draft-zhuang-tsvwg-ai-ecn-for-dcn-00

Abstract

This document is to provide an artificial intelligence (AI) based ECN adaptive reconfiguration for datacenter networks.

Status of This Memo

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1. Introduction

1.1. Background

As defined in [RFC3168], Explicit Congestion Notification is introduced for IP to allow congestion to be signaled before dropping packets. As such, the latency of applications is reduced due to less retransmission of the dropped packets. Besides, MPLS also supports ECN defined in [RFC6679]. For tunneling, [RFC6040] defines how ECN should be constructed in the case of IP-in-IP tunnels.

Meanwhile, the upper layer transports protocols, like TCP in [RFC3168] and UDP based protocols DCCP in [RFC4341][RFC4342][RFC5632] and RTP in [RFC6679] are defined to support ECN-capable functions.

With ECN marking, active queue management (AQM) can choose a non-packet loss way to indicate congestion on the device, rather than dropping packets which might ask for packet retransmission and increase the latency. By using AQM in network devices, it can signal to common congestion-controlled transports to manage the queue length in the buffer and reduce the latency of traffics. Random Early Detection (RED) specified in [RFC2309] is one of the AQM algorithms that recommended to be implemented in routers.

As stated in [RFC7567], with proper parameters, RED can be an effective algorithm. However, dynamically predicting the set of

parameters (minimum threshold and maximum threshold) is difficult. As a result, its present use in the Internet is limited. Other AQM algorithms have also been developed, while how to find proper parameters of algorithms for application traffics is still difficult and affect the network performance.

For data center networks, traffic patterns change with the deployment of applications like storage and high performance computing and changes of corresponding traffics which make the network more dynamic, while such applications have more restrict requirements on high throughput and ultra-low latency. In this area, a set of static ECN configurations suitable for all traffics at all time challenges.

With this, this document is to provide a way to seek ECN adaptive reconfiguration by using AI technologies in running data center network environment.

1.2. Intent

Our intent is to seek proper parameters of ECN adaptive reconfiguration by using artificial intelligence technologies to achieve self-tuning in a running data center network, so as to accommodate the changes of network resources to improve the network performance.

We also offer this as a starting point for seeking adaptive parameters for algorithms and network reconfigurations by using advanced technologies of AI. We do not change the way ECN works defined in [RFC3168]. With this, this document is to provide a way to achieve ECN adaptive reconfiguration by using AI technologies in dynamic data center network environment.

1.3. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

2. Architecture of the AI ECN datacenter networks

The following is a simple 2 layer data center network architecture with an analyzer to process the AI ECN adaptive reconfiguration with the changes of network traffics.

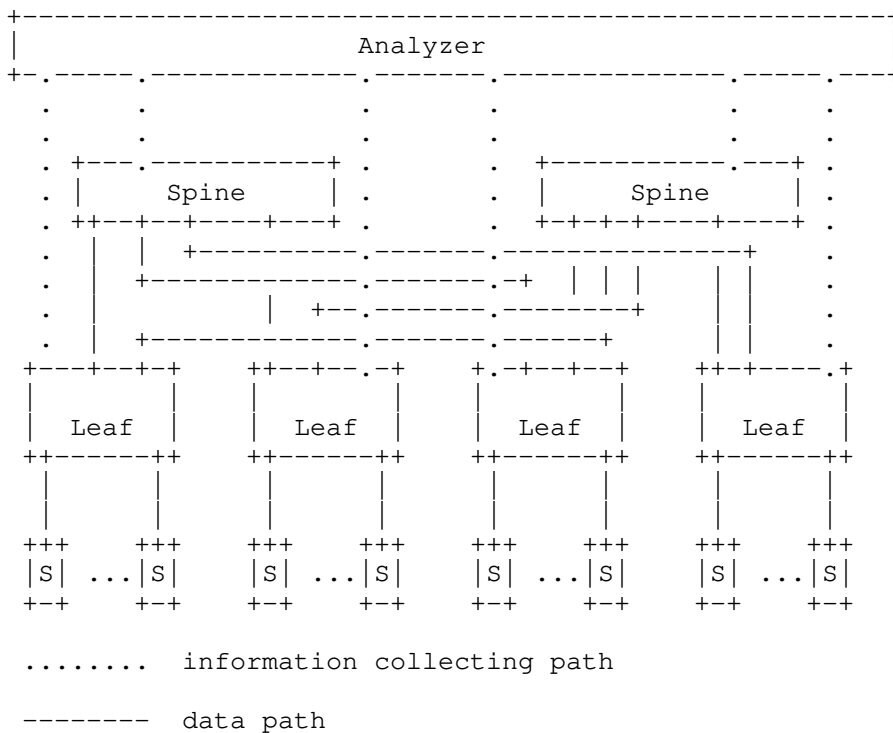


Figure 1. The architecture of a 2-layer data center network

The analyzer can be integrated with spine or can be an independent device which is left for implementation. In this design, it is responsible for collecting device information and conducting the induction for proper parameters for ECN adaptive reconfiguration periodically.

3. Scene-based ECN adaptive reconfiguration with AI

The idea of AI ECN in this document is to identify the "scene" of the current network at some time based on the collected information over a period. The identified scene (which can also be considered as a network traffic pattern) is one of the scenes that are collected and learned from datacenter networks running different traffics of various applications in training process. The ECN settings of these scenes are decided based on human experience. As such, the ECN parameters of current network can be tuned to the settings of the identified scene. This adaptive reconfiguration process is running periodically to accommodate changes of the running network environment due to traffic changes.

3.1. Scene Training

Scene training is the first process in the procedure. It composes of two steps. Firstly, construct typical scenes and generate a learning model to identify these scenes based on a set of network performance indicators. Secondly, provide proper ECN settings for these typical scenes based on human experience.

In the first step, it might need the network operator to select some typical applications and the combinations of traffics based on experience to be used as the typical training scenes. For these typical scenes, we run a learning algorithm (for example, neutral network) to learn the characteristics of these scenes from periodically collected network performance indicators.

The selected network performance indicators can be device's port bandwidth, queue size, etc al. which might be related to the applications and traffics in the networks.

While in the second step, human experience from network administrators can be used to provide proper ECN configurations for these typical scenes. AI technologies can also be used to enrich the scene sets based on these human experience, which is left for implementation.

3.2. Scene Identification and ECN Adaptive Reconfiguration

In the practical network, the analyzer periodically collects information of selected network performance indicators from network nodes. The information is then used as input to the pre-learned model and get the identified scene. The ECN settings of network devices will then be adaptively reconfigured to the parameters of the identified scene periodically.

The adaptive cycle of the period can be decided according to experience or it can be a training result in previous process defined in section 3.1.

4. Data collection and AI ECN adaptive reconfiguration

4.1. Data collection

In both training and adaptive reconfiguration process, the analyzer needs to collect information of the network i.e. a set of network performance indicators.

The data collection can be achieved by grpc or yang-push or other protocols.

4.2. ECN adaptive Reconfiguration

The adaptive reconfiguration of ECN in a running network environment can be achieved by control-plane protocols such as netconf.

5. Security Considerations

TBD

6. Manageability Consideration

TBD

7. IANA Considerations

No IANA action

8. References

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An Open Congestion Control Architecture for high performance fabrics
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Abstract

This document describes an open congestion control architecture of high performance fabrics for the cloud operators and algorithm developers to deploy or develop new congestion control algorithms as well as make appropriate configurations for traffics on smart NICs in a more efficient and flexible way.

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1. Introduction

The datacenter networks (DCNs) nowadays is not only providing traffic transmission for tenants using TCP/IP network protocol stack, but also is required to provide RDMA traffic for High Performance Computing (HPC) and distributed storage accessing applications which requires low latency and high throughput.

Thus, for datacenter application nowadays, the requirements of latency and throughput are more critical than the normal internet traffics, while network congestion and queuing caused by incast is the point that increases the traffic latency and affect the network throughput. With this, congestion control algorithms aimed for low latency and high bandwidth are proposed such as DCTCP[RFC8257], [BBR] for TCP, [DCQCN] for [RoCEv2].

Besides, the CPU utilization on NICs is another point to improve the efficiency of traffic transmission for low latency applications. By offloading some protocol processing into smart NICs and bypassing CPU, applications can directly write to hardware which reduces the latency of traffic transmission. RDMA and RoCEv2 is currently a good example to show the benefit of bypassing kernel/CPU while TCP offloading is also under discussion in [NVMe-oF].

In general, one hand, the cloud operators or application developers are working on new congestion control algorithms to fit requirements of applications like HPC, AI, storage in high performance fabrics; while on the other hand, smart NIC vendors are working on offloading functions of data plane and control plane onto hardware so as to reduce the process latency and improve the performance. In this case, it comes up with the question that how smart NICs can be optimized by offloading some functions onto the hardware while still being able to provide flexibility to customers to develop or change their congestion control algorithms and run their experiments more easily.

That said, it might be good to have an open and modular-based design for congestion control on smart NICs to be able to develop and deploy new algorithms while take the advantage of hardware offloading in a generic way.

This document is to describes an open congestion control architecture of high performance fabrics on smart NICs for the cloud operators and application developers to install or develop new congestion control algorithms as well as select appropriate controls in a more efficient and flexible way.

It only focus on the basic functionality and discuss some common interfaces to network environments and also administrators and application developers while the detailed implementations should be vendors' specific designs and are out of scope.

Discussions of new congestion control algorithms and improved active queue management (AQM) are also out of scope for this document.

2. Conventions

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

3. Abbreviations

IB - InfiniBand

HPC - High Performance Computing

ECN - Explicit Congestion Notification

AI/HPC - Artificial Intelligence/High-Performance computing

RDMA - Remote Direct Memory Access

NIC - Network Interface Card

AQM - Active Queue Management

4. Observations in storage network

Besides the benefits of easing the development of new congestion control algorithms by developers while taking advantage of hardware offloading improvement by NIC vendors, we notice that there are also benefits to choose proper algorithms for specific traffic patterns.

As stated, there are several congestion control algorithms for low latency high throughput datacenter applications and the industry is still working on enhanced algorithms for requirements of new applications in the high performance area. Then, a question might be asked, how to select a proper congestion algorithm for the network, or whether a selected algorithm is efficient and sufficient to all traffics in the network.

With this question, we use a simplified storage network as a use case for study. In this typical network, it mainly includes two traffic types: query and backup. Query is latency sensitive traffic while backup is high throughput traffic. We select several well-known congestion control algorithms (including Reno[RFC5681], Cubic[RFC8312], DCTCP[RFC8257], and BBR[BBR]) of TCP for this study.

Two set of experiments were run to see the performance of these algorithms for different traffic types (i.e. traffic patterns). The first set is to study the performance when one algorithm is used for both traffic types; the second set is to run the two traffics with combinations of congestion algorithms. The detailed experiments and testing results can be found in appendix A.

According to the result in first experiment set, BBR performs better than others when applied for both traffics; while in the second experiment set, some algorithm combinations show better performance than the same one for both, even compared with BBR.

As such, we think there are benefits for different traffic patterns to select their own algorithm in the same network to achieve better performance. This can also be a reason from cloud operation perspective to have an open congestion control on the NIC to select proper algorithms for different traffic patterns.

5. Requirements of the open congestion control architecture

According to the observation, the architecture design is suggested to follow some principles:

- o Can support developers to write their congestion control algorithms onto NICs while keep the benefit of congestion control offloading provided by NIC vendors.
- o Can support vendors to optimize the NIC performance by hardware offloading while allow users to deploy and select new congestion control algorithms.
- o Can support settings of congestion controls by administrators according to traffic patterns.
- o Can support settings from applications to provide some QoS requirements.
- o Be transport protocol independent, for example can support both TCP and RoCE.

6. Open Congestion Control (OpenCC) Architecture Overview

The architecture shown in Figure 1 only states the congestion control related components while components for other functions are omitted. The OpenCC architecture includes three layers.

The bottom layer is called the congestion control engine which provides common function blocks independent of transport protocols which can be implemented in hardware, while the middle layer is the congestion control platform in which different congestion control algorithms will be deployed here. These algorithms can be installed by NIC vendors or can be developed by algorithm developers. At last, the top layer provides all interfaces (i.e. APIs) to users, while the users can be administrators that can select proper algorithms and set proper parameters for their networks, applications that can indicate their QoS requirements which can be further mapped to some runtime settings of congestion control parameters, and the algorithm developers that can write their own algorithms.

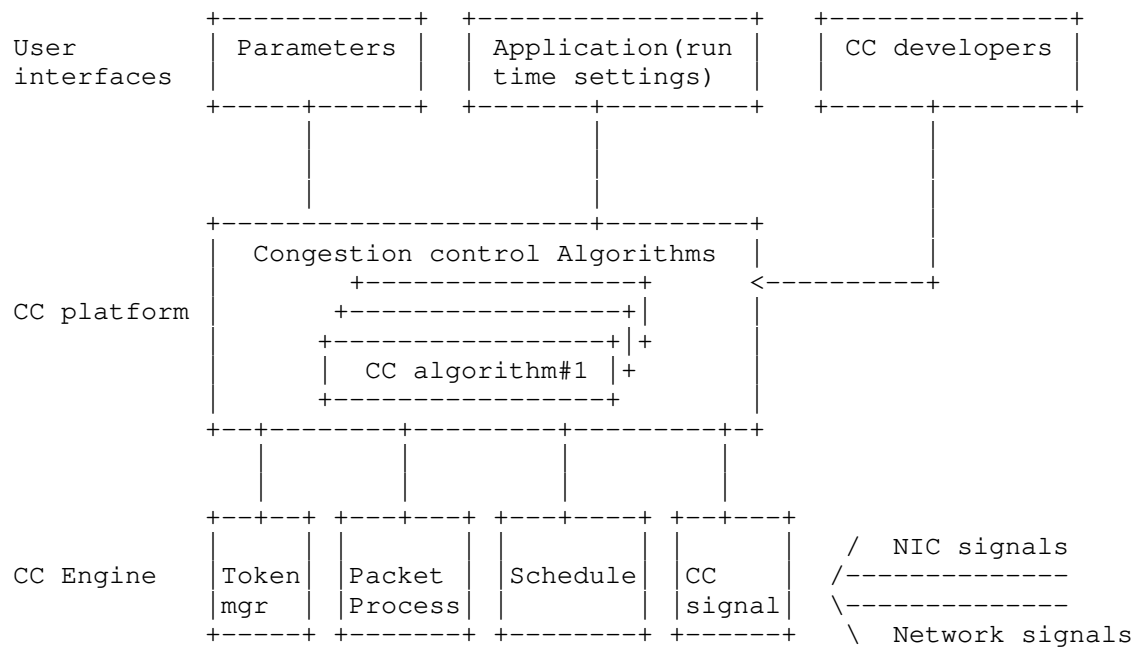


Figure 1. The architecture of open congestion control

6.1. Congestion Control Platform and its user interfaces

The congestion control platform is a software environment to deploy and configure various congestion control algorithms. It contains three types of interfaces to the user layer for different usage.

One is for administrators, which is to select proper congestion control algorithms for their network traffics and configure corresponding parameters of the selected algorithms.

The second one can be an interface defined by NIC vendors or developers that provide some APIs for application developers to define their QoS requirements which will be further mapped to some runtime configuration of the controls.

The last one is for algorithm developers to write their own algorithm in the system. It is suggested to have a defined common language to write algorithms which can be further compiled by vendor specific environments (in which some toolkits or library can be provided) to generate the platform dependent codes.

6.2. Congestion Control Engine (CCE) and its interfaces

Components in the congestion control engine can be offloaded to the hardware to improve the performance. As such, it is suggested to provide some common and basic functions while the upper platform can provide more extensibility and more flexibility for more functions.

The CCE includes basic modules of packet transmission and corresponding control. Several function blocks are illustrated here while the detailed implementation is out of scope for this document and left for NIC vendors. A token manager is used to distribute tokens to traffics while the schedule block is to schedule the transmission time for these traffics. The packet process block is to edit or process the packet before transmission. The congestion control signal block is to collect or monitor signals from both network and other NICs which will be fed to congestion control algorithms.

As such, an interface to get congestion control signal in the congestion control should be defined to receive signals from both other NICs and networks for existing congestion control algorithms and new extensions. These information will be used as inputs of control algorithms to adjust the sending rate and operate the loss recovery et.al.

7. Interoperability Consideration

7.1. Negotiate the congestion control algorithm

Since there will be several congestion control algorithms, the host might negotiate their supported congestion control capability during the session setup phase. However, it should use the existing way of congestion control as default to provide compatibility with legacy devices.

Also, the network devices on the path should be capable to indicate their capability of any specific signals that the congestion control algorithm needs. The capability negotiation between NICs and Switches can be considered either some in-band ECN-like negotiations or out-of-band individual message negotiations.

Alternatively, the system can also use a centralized administration platform to configure the algorithms on NICs and network devices.

7.2. Negotiate the congestion control parameters

The parameters might be set by administrators to meet their traffic patterns and network environments or be set by mappings from application requirements. Hence, these parameters might be changed after the session is set up. As such, hosts should be able to negotiate their parameters when changed or be configured to keep consistent.

8. Security Considerations

TBD

9. Manageability Consideration

TBD

10. IANA Considerations

No IANA action

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Appendix A. Experiments

This section includes two sets of experiments to study the performance of congestion control algorithms in a simplified storage network. The first set is to study one algorithm applied for both query and backup traffics while the second set is to study the performance when different algorithms are used for query traffic and backup traffic. The metrics include throughput of backup traffic, average completion time of query traffic and 95% percentile query completion time.

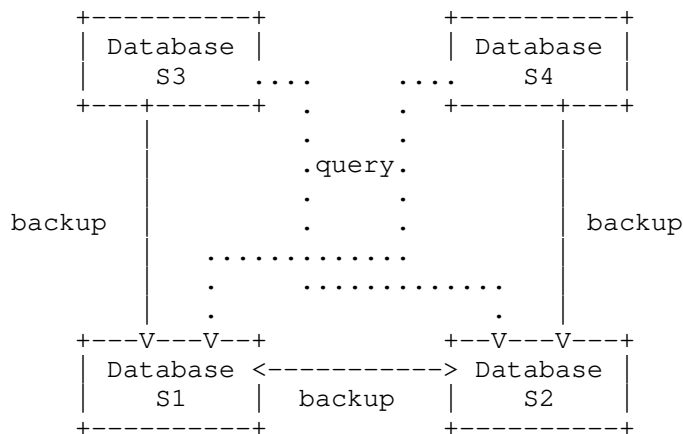


Figure 2. Simplified storage network topology

All experiments are a full implementation of congestion control algorithms on NICs, including Reno, Cubic, DCTCP and BBR. Our experiments includes 4 servers connecting to one switch. Each server with a 10Gbps NIC connected to a 10Gbps port on the switch. However, we limit all ports to 1Gbps to make congestion points. In the experiments, the database server S1 receives backup traffics from both S3 and S2 and one query traffic from S4. The server S2 gets back traffics from S1 and S4 and one query traffic from S3. In the experiments, three traffic flows are transmitted to S1 from one egress port on the switch, which might cause congestion.

In the first experiment set, we test one algorithm for both traffics. The result is shown below in table 1.

	reno	cubic	bbr	dctcp
Throughput MB/s	64.92	65.97	75.25	70.06
Avg. comp ms	821.61	858.05	85.68	99.90
95% comp ms	894.65	911.23	231.75	273.92

Table 1. Performance when use one cc for both query and backup traffics

As we can see, the average completion time of BBR and DCTCP is 10 times better than that of reno and cubic. BBR is the best to keep high throughput.

In the second set, we test all the combinations of algorithms for the two traffics.

1. Reno for query traffic

reno@query

@backup	cubic	bbr	dctcp	reno
Throughput MB/s	66.00	76.19	64.00	64.92
Avg. comp ms	859.61	81.87	18.38	821.61
95% comp ms	917.80	149.88	20.38	894.65

Table 2. reno @ query and cubic, bbr, dctcp @ backup

It shows that given reno used for query traffic, bbr for backup traffic gets better throughput compared with other candidates. However, dctcp for backup traffic gets much better average completion time and 95% completion time, almost 6 times better than those of bbr even its throughput is less than bbr. The reason for this might be bbr does not consider lost packets and congestion levels which might cause much retransmission. In this test set, dctcp for backup traffic gets better performance.

2. Cubic for query traffic

cubic@query

@backup	reno	bbr	dctcp	cubic
Throughput MB/s	64.92	75.02	65.29	65.97
Avg. comp ms	819.23	83.50	18.42	858.05
95% comp ms	902.66	170.96	20.99	911.23

Table 3. cubic @ query and reno, bbr, dctcp @ backup

The results of cubic for query traffic are similar to those of reno. Even with less throughput, dctcp has almost 6 times better than bbr in average completion time and 95% completion time, and nearly 10 times better than those of reno and cubic.

3. Bbr for query traffic

bbr@query

@backup	reno	cubic	dctcp	bbr
Throughput MB/s	64.28	66.61	65.29	75.25
Avg. comp ms	866.05	895.12	18.49	85.68
95% comp ms	925.06	967.67	20.86	231.75

Table 4. bbr @ query and reno, cubi, dctcp @ backup

The results still match those we get from reno and cubic. In the last two columns, dctcp for backup shows better performance even when we compared with bbr used for backup. It indicates that bbr @ query and dctcp @ backup is better than bbr @ query and backup.

4. Dctcp for query traffic

dctcp@query

@backup	reno	cubic	bbr	dctcp
Throughput MB/s	60.93	64.49	76.15	70.06
Avg. comp ms	2817.53	3077.20	816.45	99.90
95% comp ms	3448.53	3639.94	2362.72	273.92

Table 5. dctcp @ query and reno, cubi, bbr @ backup

The results for dctcp@query look worse than others in completion time, since we don't introduce L4S in the experiments which means dctcp will back off most of the time when congestion happens which makes the query traffic bares long latency. The best performance in this test set happens at dctcp@backup. In this setting, both traffics have use the same mechanism to back off their traffics. However, the number is still worse than when other algorithms are used for query and dctcp used for backup.

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