

Internet Engineering Task Force (IETF)  
Request for Comments: 8085  
BCP: 145  
Obsoletes: 5405  
Category: Best Current Practice  
ISSN: 2070-1721

L. Eggert  
NetApp  
G. Fairhurst  
University of Aberdeen  
G. Shepherd  
Cisco Systems  
March 2017

## UDP Usage Guidelines

### Abstract

The User Datagram Protocol (UDP) provides a minimal message-passing transport that has no inherent congestion control mechanisms. This document provides guidelines on the use of UDP for the designers of applications, tunnels, and other protocols that use UDP. Congestion control guidelines are a primary focus, but the document also provides guidance on other topics, including message sizes, reliability, checksums, middlebox traversal, the use of Explicit Congestion Notification (ECN), Differentiated Services Code Points (DSCPs), and ports.

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. They may also need to implement additional mechanisms, depending on how they use UDP.

Some guidance is also applicable to the design of other protocols (e.g., protocols layered directly on IP or via IP-based tunnels), especially when these protocols do not themselves provide congestion control.

This document obsoletes RFC 5405 and adds guidelines for multicast UDP usage.

## Status of This Memo

This memo documents an Internet Best Current Practice.

This document is a product of the Internet Engineering Task Force (IETF). It represents the consensus of the IETF community. It has received public review and has been approved for publication by the Internet Engineering Steering Group (IESG). Further information on BCPs is available in Section 2 of RFC 7841.

Information about the current status of this document, any errata, and how to provide feedback on it may be obtained at <http://www.rfc-editor.org/info/rfc8085>.

## Copyright Notice

Copyright (c) 2017 IETF Trust and the persons identified as the document authors. All rights reserved.

This document is subject to BCP 78 and the IETF Trust's Legal Provisions Relating to IETF Documents (<http://trustee.ietf.org/license-info>) in effect on the date of publication of this document. Please review these documents carefully, as they describe your rights and restrictions with respect to this document. Code Components extracted from this document must include Simplified BSD License text as described in Section 4.e of the Trust Legal Provisions and are provided without warranty as described in the Simplified BSD License.

## Table of Contents

1. Introduction .....	3
2. Terminology .....	5
3. UDP Usage Guidelines .....	5
3.1. Congestion Control Guidelines .....	6
3.2. Message Size Guidelines .....	19
3.3. Reliability Guidelines .....	21
3.4. Checksum Guidelines .....	22
3.5. Middlebox Traversal Guidelines .....	25
3.6. Limited Applicability and Controlled Environments .....	27
4. Multicast UDP Usage Guidelines .....	28
4.1. Multicast Congestion Control Guidelines .....	30
4.2. Message Size Guidelines for Multicast .....	32
5. Programming Guidelines .....	32
5.1. Using UDP Ports .....	34
5.2. ICMP Guidelines .....	37
6. Security Considerations .....	38
7. Summary .....	40
8. References .....	42
8.1. Normative References .....	42
8.2. Informative References .....	43
Appendix A. ....	53
Acknowledgments .....	55
Authors' Addresses .....	55

## 1. Introduction

The User Datagram Protocol (UDP) [RFC768] provides a minimal, unreliable, best-effort, message-passing transport to applications and other protocols (such as tunnels) that wish to operate over IP. Both are simply called "applications" in the remainder of this document.

Compared to other transport protocols, UDP and its UDP-Lite variant [RFC3828] are unique in that they do not establish end-to-end connections between communicating end systems. UDP communication consequently does not incur connection establishment and teardown overheads, and there is minimal associated end-system state. Because of these characteristics, UDP can offer a very efficient communication transport to some applications.

A second unique characteristic of UDP is that it provides no inherent congestion control mechanisms. On many platforms, applications can send UDP datagrams at the line rate of the platform's link interface, which is often much greater than the available end-to-end path capacity, and doing so contributes to congestion along the path. [RFC2914] describes the best current practice for congestion control

in the Internet. It identifies two major reasons why congestion control mechanisms are critical for the stable operation of the Internet:

1. The prevention of congestion collapse, i.e., a state where an increase in network load results in a decrease in useful work done by the network.
2. The establishment of a degree of fairness, i.e., allowing multiple flows to share the capacity of a path reasonably equitably.

Because UDP itself provides no congestion control mechanisms, it is up to the applications that use UDP for Internet communication to employ suitable mechanisms to prevent congestion collapse and establish a degree of fairness. [RFC2309] discusses the dangers of congestion-unresponsive flows and states that "all UDP-based streaming applications should incorporate effective congestion avoidance mechanisms." [RFC7567] reaffirms this statement. This is an important requirement, even for applications that do not use UDP for streaming. In addition, congestion-controlled transmission is of benefit to an application itself, because it can reduce self-induced packet loss, minimize retransmissions, and hence reduce delays. Congestion control is essential even at relatively slow transmission rates. For example, an application that generates five 1500-byte UDP datagrams in one second can already exceed the capacity of a 56 Kb/s path. For applications that can operate at higher, potentially unbounded data rates, congestion control becomes vital to prevent congestion collapse and establish some degree of fairness. Section 3 describes a number of simple guidelines for the designers of such applications.

A UDP datagram is carried in a single IP packet and is hence limited to a maximum payload of 65,507 bytes for IPv4 and 65,527 bytes for IPv6. The transmission of large IP packets usually requires IP fragmentation. Fragmentation decreases communication reliability and efficiency and should be avoided. IPv6 allows the option of transmitting large packets ("jumbograms") without fragmentation when all link layers along the path support this [RFC2675]. Some of the guidelines in Section 3 describe how applications should determine appropriate message sizes. Other sections of this document provide guidance on reliability, checksums, middlebox traversal and use of multicast.

This document provides guidelines and recommendations. Although most UDP applications are expected to follow these guidelines, there do exist valid reasons why a specific application may decide not to follow a given guideline. In such cases, it is RECOMMENDED that

application designers cite the respective section(s) of this document in the technical specification of their application or protocol and explain their rationale for their design choice.

[RFC5405] was scoped to provide guidelines for unicast applications only, whereas this document also provides guidelines for UDP flows that use IP anycast, multicast, broadcast, and applications that use UDP tunnels to support IP flows.

Finally, although this document specifically refers to usage of UDP, the spirit of some of its guidelines also applies to other message-passing applications and protocols (specifically on the topics of congestion control, message sizes, and reliability). Examples include signaling, tunnel or control applications that choose to run directly over IP by registering their own IP protocol number with IANA. This document is expected to provide useful background reading to the designers of such applications and protocols.

## 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].

## 3. UDP Usage Guidelines

Internet paths can have widely varying characteristics, including transmission delays, available bandwidths, congestion levels, reordering probabilities, supported message sizes, or loss rates. Furthermore, the same Internet path can have very different conditions over time. Consequently, applications that may be used on the Internet MUST NOT make assumptions about specific path characteristics. They MUST instead use mechanisms that let them operate safely under very different path conditions. Typically, this requires conservatively probing the current conditions of the Internet path they communicate over to establish a transmission behavior that it can sustain and that is reasonably fair to other traffic sharing the path.

These mechanisms are difficult to implement correctly. For most applications, the use of one of the existing IETF transport protocols is the simplest method of acquiring the required mechanisms. Doing so also avoids issues that protocols using a new IP protocol number face when being deployed over the Internet, where middleboxes that only support TCP and UDP are sometimes present. Consequently, the RECOMMENDED alternative to the UDP usage described in the remainder of this section is the use of an IETF transport protocol such as TCP

[RFC793], Stream Control Transmission Protocol (SCTP) [RFC4960], and SCTP Partial Reliability Extension (SCTP-PR) [RFC3758], or Datagram Congestion Control Protocol (DCCP) [RFC4340] with its different congestion control types [RFC4341][RFC4342][RFC5622], or transport protocols specified by the IETF in the future. (UDP-encapsulated SCTP [RFC6951] and DCCP [RFC6773] can offer support for traversing firewalls and other middleboxes where the native protocols are not supported.)

If used correctly, these more fully featured transport protocols are not as "heavyweight" as often claimed. For example, the TCP algorithms have been continuously improved over decades, and they have reached a level of efficiency and correctness that custom application-layer mechanisms will struggle to easily duplicate. In addition, many TCP implementations allow connections to be tuned by an application to its purposes. For example, TCP's "Nagle" algorithm [RFC1122] can be disabled, improving communication latency at the expense of more frequent -- but still congestion controlled -- packet transmissions. Another example is the TCP SYN cookie mechanism [RFC4987], which is available on many platforms. TCP with SYN cookies does not require a server to maintain per-connection state until the connection is established. TCP also requires the end that closes a connection to maintain the TIME-WAIT state that prevents delayed segments from one connection instance from interfering with a later one. Applications that are aware of and designed for this behavior can shift maintenance of the TIME-WAIT state to conserve resources by controlling which end closes a TCP connection [FABER]. Finally, TCP's built-in capacity-probing and awareness of the maximum transmission unit supported by the path (PMTU) results in efficient data transmission that quickly compensates for the initial connection setup delay, in the case of transfers that exchange more than a few segments.

### 3.1. Congestion Control Guidelines

If an application or protocol chooses not to use a congestion-controlled transport protocol, it SHOULD control the rate at which it sends UDP datagrams to a destination host, in order to fulfill the requirements of [RFC2914]. It is important to stress that an application SHOULD perform congestion control over all UDP traffic it sends to a destination, independently from how it generates this traffic. For example, an application that forks multiple worker processes or otherwise uses multiple sockets to generate UDP datagrams SHOULD perform congestion control over the aggregate traffic.

Several approaches to perform congestion control are discussed in the remainder of this section. This section describes generic topics with an intended emphasis on unicast and anycast [RFC1546] usage. Not all approaches discussed below are appropriate for all UDP-transmitting applications. Section 3.1.2 discusses congestion control options for applications that perform bulk transfers over UDP. Such applications can employ schemes that sample the path over several subsequent round-trips during which data is exchanged to determine a sending rate that the path at its current load can support. Other applications only exchange a few UDP datagrams with a destination. Section 3.1.3 discusses congestion control options for such "low data-volume" applications. Because they typically do not transmit enough data to iteratively sample the path to determine a safe sending rate, they need to employ different kinds of congestion control mechanisms. Section 3.1.11 discusses congestion control considerations when UDP is used as a tunneling protocol. Section 4 provides additional recommendations for broadcast and multicast usage.

It is important to note that congestion control should not be viewed as an add-on to a finished application. Many of the mechanisms discussed in the guidelines below require application support to operate correctly. Application designers need to consider congestion control throughout the design of their application, similar to how they consider security aspects throughout the design process.

In the past, the IETF has also investigated integrated congestion control mechanisms that act on the traffic aggregate between two hosts, i.e., a framework such as the Congestion Manager [RFC3124], where active sessions may share current congestion information in a way that is independent of the transport protocol. Such mechanisms have currently failed to see deployment, but would otherwise simplify the design of congestion control mechanisms for UDP sessions, so that they fulfill the requirements in [RFC2914].

#### 3.1.1. Protocol Timer Guidelines

Understanding the latency between communicating endpoints is usually a crucial part of effective congestion control implementations for protocols and applications. Latency estimation can be used in a number of protocol functions, such as calculating a congestion-controlled transmission rate, triggering retransmission, and detecting packet loss. Additional protocol functions, for example, determining an interval for probing a path, determining an interval between keep-alive messages, determining an interval for measuring the quality of experience, or determining if a remote endpoint has

responded to a request to perform an action, typically operate over longer timescales than congestion control and therefore are not covered in this section.

The general recommendation in this document is that applications SHOULD leverage existing congestion control techniques and the latency estimators specified therein (see next subsection). The following guidelines are provided for applications that need to design their own latency estimation mechanisms.

The guidelines are framed in terms of "latency" and not "round-trip time" because some situations require characterizing only the network-based latency (e.g., TCP-Friendly Rate Control (TFRC) [RFC5348]), while other cases necessitate inclusion of the time required by the remote endpoint to provide feedback (e.g., developing an understanding of when to retransmit a message).

The latency between endpoints is generally a dynamic property. Therefore, estimates SHOULD represent some sort of averaging of multiple recent measurement samples to account for variance. Leveraging an Exponentially Weighted Moving Average (EWMA) has proven useful for this purpose (e.g., in TCP [RFC6298] and TFRC [RFC5348]).

Independent latency estimates SHOULD be maintained for each destination with which an endpoint communicates.

Latency samples MUST NOT be derived from ambiguous transactions. The canonical example is in a protocol that retransmits data, but subsequently cannot determine which copy is being acknowledged. This ambiguity makes correct computation of the latency problematic. See the discussion of Karn's algorithm in [RFC6298]. This requirement ensures a sender establishes a sound estimate of the latency without relying on misleading measurements.

When a latency estimate is used to arm a timer that provides loss detection -- with or without retransmission -- expiry of the timer MUST be interpreted as an indication of congestion in the network, causing the sending rate to be adapted to a safe conservative rate (e.g., TCP collapses the congestion window to one segment [RFC5681]).

Some applications require an initial latency estimate before the latency between endpoints can be empirically sampled. For instance, when arming a retransmission timer, an initial value is needed to protect the messages sent before the endpoints sample the latency. This initial latency estimate SHOULD generally be as conservative (large) as possible for the given application. For instance, in the absence of any knowledge about the latency of a path, TCP requires the initial Retransmission Timeout (RTO) to be set to no less than 1

second [RFC6298]. UDP applications SHOULD similarly use an initial latency estimate of 1 second. Values shorter than 1 second can be problematic (see the data analysis in the appendix of [RFC6298]).

### 3.1.2. Bulk-Transfer Applications

Applications that perform bulk transmission of data to a peer over UDP, i.e., applications that exchange more than a few UDP datagrams per RTT, SHOULD implement TFRC [RFC5348], window-based TCP-like congestion control, or otherwise ensure that the application complies with the congestion control principles.

TFRC has been designed to provide both congestion control and fairness in a way that is compatible with the IETF's other transport protocols. If an application implements TFRC, it need not follow the remaining guidelines in Section 3.1.2, because TFRC already addresses them, but it SHOULD still follow the remaining guidelines in the subsequent subsections of Section 3.

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.

Section 2 of [RFC3551] suggests that applications SHOULD monitor the packet-loss rate to ensure that it is within acceptable parameters. Packet loss is considered acceptable if a TCP flow across the same network path under the same network conditions would achieve an average throughput, measured on a reasonable timescale, that is not less than that of the UDP flow. The comparison to TCP cannot be specified exactly, but is intended as an "order-of-magnitude" comparison in timescale and throughput. The recommendations for managing timers specified in Section 3.1.1 also apply.

Finally, some bulk-transfer applications may choose not to implement any congestion control mechanism and instead rely on transmitting across reserved path capacity (see Section 3.1.9). This might be an acceptable choice for a subset of restricted networking environments, but is by no means a safe practice for operation over the wider Internet. When the UDP traffic of such applications leaks out into unprovisioned Internet paths, it can significantly degrade the performance of other traffic sharing the path and even result in congestion collapse. Applications that support an uncontrolled or unadaptive transmission behavior SHOULD NOT do so by default and SHOULD instead require users to explicitly enable this mode of operation, and they SHOULD verify that sufficient path capacity has been reserved for them.

### 3.1.3. Low Data-Volume Applications

When applications that at any time exchange only a few UDP datagrams with a destination implement TFRC or one of the other congestion control schemes in Section 3.1.2, the network sees little benefit, because those mechanisms perform congestion control in a way that is only effective for longer transmissions.

Applications that at any time exchange only a few UDP datagrams with a destination SHOULD still control their transmission behavior by not sending on average more than one UDP datagram per RTT to a destination. Similar to the recommendation in [RFC1536], an application SHOULD maintain an estimate of the RTT for any destination with which it communicates using the methods specified in Section 3.1.1.

Some applications cannot maintain a reliable RTT estimate for a destination. These applications do not need to or are unable to use protocol timers to measure the RTT (Section 3.1.1). Two cases can be identified:

1. The first case is that of applications that exchange too few UDP datagrams with a peer to establish a statistically accurate RTT estimate but that can monitor the reliability of transmission (Section 3.3). Such applications MAY use a predetermined transmission interval that is exponentially backed off when packets are deemed lost. TCP specifies an initial value of 1 second [RFC6298], which is also RECOMMENDED as an initial value for UDP applications. Some low data-volume applications, e.g., SIP [RFC3261] and General Internet Signaling Transport (GIST) [RFC5971] use an interval of 500 ms, and shorter values are likely problematic in many cases. As in the previous case, note that the initial timeout is not the maximum possible timeout, see Section 3.1.1.
2. A second case of applications cannot maintain an RTT estimate for a destination, because the destination does not send return traffic. Such applications SHOULD NOT send more than one UDP datagram every 3 seconds and SHOULD use an even less aggressive rate when possible. Shorter values are likely problematic in many cases. Note that the sending rate in this case must be more conservative than in the previous cases, because the lack of return traffic prevents the detection of packet loss, i.e., congestion, and the application therefore cannot perform exponential back off to reduce load.

#### 3.1.4. Applications Supporting Bidirectional Communications

Applications that communicate bidirectionally SHOULD employ congestion control for both directions of the communication. For example, for a client-server, request-response-style application, clients SHOULD congestion-control their request transmission to a server, and the server SHOULD congestion-control its responses to the clients. Congestion in the forward and reverse directions is uncorrelated, and an application SHOULD either independently detect and respond to congestion along both directions or limit new and retransmitted requests based on acknowledged responses across the entire round-trip path.

#### 3.1.5. Implications of RTT and Loss Measurements on Congestion Control

Transports such as TCP, SCTP, and DCCP provide timely detection of congestion that results in an immediate reduction of their maximum sending rate when congestion is experienced. This reaction is typically completed 1-2 RTTs after loss/congestion is encountered. Applications using UDP SHOULD implement a congestion control scheme that provides a prompt reaction to signals indicating congestion (e.g., by reducing the rate within the next RTT following a congestion signal).

The operation of a UDP congestion control algorithm can be very different from the way TCP operates. This includes congestion controls that respond on timescales that fit applications that cannot usefully work within the "change rate every RTT" model of TCP. Applications that experience a low or varying RTT are particularly vulnerable to sampling errors (e.g., due to measurement noise or timer accuracy). This suggests the need to average loss/congestion and RTT measurements over a longer interval; however, this also can contribute additional delay in detecting congestion. Some applications may not react by reducing their sending rate immediately for various reasons, including the following: RTT and loss measurements are only made periodically (e.g., using RTCP), additional time is required to filter information, or the application is only able to change its sending rate at predetermined interval (e.g., some video codecs).

When designing a congestion control algorithm, the designer therefore needs to consider the total time taken to reduce the load following a lack of feedback or a congestion event. An application where the most recent RTT measurement is smaller than the actual RTT or the measured loss rate is smaller than the current rate, can result in over estimating the available capacity. Such over-estimation can

result in a sending rate that creates congestion to the application or other flows sharing the path capacity, and can contribute to congestion collapse -- both of these need to be avoided.

A congestion control designed for UDP SHOULD respond as quickly as possible when it experiences congestion, and it SHOULD take into account both the loss rate and the response time when choosing a new rate. The implemented congestion control scheme SHOULD result in bandwidth (capacity) use that is comparable to that of TCP within an order of magnitude, so that it does not starve other flows sharing a common bottleneck.

#### 3.1.6. Burst Mitigation and Pacing

UDP applications SHOULD provide mechanisms to regulate the bursts of transmission that the application may send to the network. Many TCP and SCTP implementations provide mechanisms that prevent a sender from generating long bursts at line-rate, since these are known to induce early loss to applications sharing a common network bottleneck. The use of pacing with TCP [ALLMAN] has also been shown to improve the coexistence of TCP flows with other flows. The need to avoid excessive transmission bursts is also noted in specifications for applications (e.g., [RFC7143]).

Even low data-volume UDP flows may benefit from packet pacing, e.g., an application that sends three copies of a packet to improve robustness to loss is RECOMMENDED to pace out those three packets over several RTTs, to reduce the probability that all three packets will be lost due to the same congestion event (or other event, such as burst corruption).

#### 3.1.7. Explicit Congestion Notification

Internet applications can use Explicit Congestion Notification (ECN) [RFC3168] to gain benefits for the services they support [RFC8087].

Internet transports, such as TCP, provide a set of mechanisms that are needed to utilize ECN. ECN operates by setting an ECN-capable codepoint (ECT(0) or ECT(1)) in the IP header of packets that are sent. This indicates to ECN-capable network devices (routers and other devices) that they may mark (set the congestion experienced, Congestion Experience (CE) codepoint) rather than drop the IP packet as a signal of incipient congestion.

UDP applications can also benefit from enabling ECN, providing that the API supports ECN and that they implement the required protocol mechanisms to support ECN.

The set of mechanisms required for an application to use ECN over UDP are:

- o A sender MUST provide a method to determine (e.g., negotiate) that the corresponding application is able to provide ECN feedback using a compatible ECN method.
- o A receiver that enables the use of ECN for a UDP port MUST check the ECN field at the receiver for each UDP datagram that it receives on this port.
- o The receiving application needs to provide feedback of congestion information to the sending application. This MUST report the presence of datagrams received with a CE-mark by providing a mechanism to feed this congestion information back to the sending application. The feedback MAY also report the presence of ECT(1) and ECT(0)/Not-ECT packets [RFC7560]. ([RFC3168] and [RFC7560] specify methods for TCP.)
- o An application sending ECN-capable datagrams MUST provide an appropriate congestion reaction when it receives feedback indicating that congestion has been experienced. This ought to result in reduction of the sending rate by the UDP congestion control method (see Section 3.1) that is not less than the reaction of TCP under equivalent conditions.
- o A sender SHOULD detect network paths that do not support the ECN field correctly. When detected, they need to either conservatively react to congestion or even fall back to not using ECN [RFC8087]. This method needs to be robust to changes within the network path that may occur over the lifetime of a session.
- o A sender is encouraged to provide a mechanism to detect and react appropriately to misbehaving receivers that fail to report CE-marked packets [RFC8087].

[RFC6679] provides guidance and an example of this support, by describing a method to allow ECN to be used for UDP-based applications using the Real-Time Protocol (RTP). Applications that cannot provide this set of mechanisms, but wish to gain the benefits of using ECN, are encouraged to use a transport protocol that already supports ECN (such as TCP).

#### 3.1.8. Differentiated Services Model

An application using UDP can use the differentiated services (DiffServ) Quality of Service (QoS) framework. To enable differentiated services processing, a UDP sender sets the

Differentiated Services Code Point (DSCP) field [RFC2475] in packets sent to the network. Normally, a UDP source/destination port pair will set a single DSCP value for all packets belonging to a flow, but multiple DSCPs can be used as described later in this section. A DSCP may be chosen from a small set of fixed values (the class selector code points), or from a set of recommended values defined in the Per Hop Behavior (PHB) specifications, or from values that have purely local meanings to a specific network that supports DiffServ. In general, packets may be forwarded across multiple networks between source and destination.

In setting a non-default DSCP value, an application must be aware that DSCP markings may be changed or removed between the traffic source and destination. This has implications on the design of applications that use DSCPs. Specifically, applications **SHOULD** be designed not to rely on implementation of a specific network treatment; they need instead to implement congestion control methods to determine if their current sending rate is inducing congestion in the network.

[RFC7657] describes the implications of using DSCPs and provides recommendations on using multiple DSCPs within a single network five-tuple (source and destination addresses, source and destination ports, and the transport protocol used, in this case, UDP or UDP-Lite), and particularly the expected impact on transport protocol interactions, with congestion control or reliability functionality (e.g., retransmission, reordering). Use of multiple DSCPs can result in reordering by increasing the set of network forwarding resources used by a sender. It can also increase exposure to resource depletion or failure.

#### 3.1.9. QoS, Pre-Provisioned, or Reserved Capacity

The IETF usually specifies protocols for use within the Best Effort General Internet. Sometimes it is relevant to specify protocols with a different applicability. An application using UDP can use the integrated services QoS framework. This framework is usually made available within controlled environments (e.g., within a single administrative domain or bilaterally agreed connection between domains). Applications intended for the Internet **SHOULD NOT** assume that QoS mechanisms are supported by the networks they use, and therefore need to provide congestion control, error recovery, etc., in case the actual network path does not provide provisioned service.

Some UDP applications are only expected to be deployed over network paths that use pre-provisioned capacity or capacity reserved using dynamic provisioning, e.g., through the Resource Reservation Protocol (RSVP). Multicast applications are also used with pre-provisioned

capacity (e.g., IPTV deployments within access networks). These applications MAY choose not to implement any congestion control mechanism and instead rely on transmitting only on paths where the capacity is provisioned and reserved for this use. This might be an acceptable choice for a subset of restricted networking environments, but is by no means a safe practice for operation over the wider Internet. Applications that choose this option SHOULD carefully and in detail describe the provisioning and management procedures that result in the desired containment.

Applications that support an uncontrolled or unadaptive transmission behavior SHOULD NOT do so by default and SHOULD instead require users to explicitly enable this mode of operation.

Applications designed for use within a controlled environment (see Section 3.6) may be able to exploit network management functions to detect whether they are causing congestion, and react accordingly. If the traffic of such applications leaks out into unprovisioned Internet paths, it can significantly degrade the performance of other traffic sharing the path and even result in congestion collapse. Protocols designed for such networks SHOULD provide mechanisms at the network edge to prevent leakage of traffic into unprovisioned Internet paths (e.g., [RFC7510]). To protect other applications sharing the same path, applications SHOULD also deploy an appropriate circuit breaker, as described in Section 3.1.10.

An IETF specification targeting a controlled environment is expected to provide an applicability statement that restricts the application to the controlled environment (see Section 3.6).

#### 3.1.10. Circuit Breaker Mechanisms

A transport circuit breaker is an automatic mechanism that is used to estimate the congestion caused by a flow, and to terminate (or significantly reduce the rate of) the flow when excessive congestion is detected [RFC8084]. This is a safety measure to prevent congestion collapse (starvation of resources available to other flows), essential for an Internet that is heterogeneous and for traffic that is hard to predict in advance.

A circuit breaker is intended as a protection mechanism of last resort. Under normal circumstances, a circuit breaker should not be triggered; it is designed to protect things when there is severe overload. The goal is usually to limit the maximum transmission rate that reflects the available capacity of a network path. Circuit breakers can operate on individual UDP flows or traffic aggregates, e.g., traffic sent using a network tunnel.

[RFC8084] provides guidance and examples on the use of circuit breakers. The use of a circuit breaker in RTP is specified in [RFC8083].

Applications used in the general Internet SHOULD implement a transport circuit breaker if they do not implement congestion control or operate a low data-volume service (see Section 3.6). All applications MAY implement a transport circuit breaker [RFC8084] and are encouraged to consider implementing at least a slow-acting transport circuit breaker to provide a protection of last resort for their network traffic.

### 3.1.11. UDP Tunnels

One increasingly popular use of UDP is as a tunneling protocol [INT-TUNNELS], where a tunnel endpoint encapsulates the packets of another protocol inside UDP datagrams and transmits them to another tunnel endpoint, which decapsulates the UDP datagrams and forwards the original packets contained in the payload. One example of such a protocol is Teredo [RFC4380]. Tunnels establish virtual links that appear to directly connect locations that are distant in the physical Internet topology and can be used to create virtual (private) networks. Using UDP as a tunneling protocol is attractive when the payload protocol is not supported by middleboxes that may exist along the path, because many middleboxes support transmission using UDP.

Well-implemented tunnels are generally invisible to the endpoints that happen to transmit over a path that includes tunneled links. On the other hand, to the routers along the path of a UDP tunnel, i.e., the routers between the two tunnel endpoints, the traffic that a UDP tunnel generates is a regular UDP flow, and the encapsulator and decapsulator appear as regular UDP-sending and UDP-receiving applications. Because other flows can share the path with one or more UDP tunnels, congestion control needs to be considered.

Two factors determine whether a UDP tunnel needs to employ specific congestion control mechanisms: first, whether the payload traffic is IP-based; and second, whether the tunneling scheme generates UDP traffic at a volume that corresponds to the volume of payload traffic carried within the tunnel.

IP-based unicast traffic is generally assumed to be congestion controlled, i.e., it is assumed that the transport protocols generating IP-based unicast traffic at the sender already employ mechanisms that are sufficient to address congestion on the path. Consequently, a tunnel carrying IP-based unicast traffic should

already interact appropriately with other traffic sharing the path, and specific congestion control mechanisms for the tunnel are not necessary.

However, if the IP traffic in the tunnel is known not to be congestion controlled, additional measures are RECOMMENDED to limit the impact of the tunneled traffic on other traffic sharing the path. For the specific case of a tunnel that carries IP multicast traffic, see Section 4.1.

The following guidelines define these possible cases in more detail:

1. A tunnel generates UDP traffic at a volume that corresponds to the volume of payload traffic, and the payload traffic is IP based and congestion controlled.

This is arguably the most common case for Internet tunnels. In this case, the UDP tunnel SHOULD NOT employ its own congestion control mechanism, because congestion losses of tunneled traffic will already trigger an appropriate congestion response at the original senders of the tunneled traffic. A circuit breaker mechanism may provide benefit by controlling the envelope of the aggregated traffic.

Note that this guideline is built on the assumption that most IP-based communication is congestion controlled. If a UDP tunnel is used for IP-based traffic that is known to not be congestion controlled, the next set of guidelines applies.

2. A tunnel generates UDP traffic at a volume that corresponds to the volume of payload traffic, and the payload traffic is not known to be IP based, or is known to be IP based but not congestion controlled.

This can be the case, for example, when some link-layer protocols are encapsulated within UDP (but not all link-layer protocols; some are congestion controlled). Because it is not known that congestion losses of tunneled non-IP traffic will trigger an appropriate congestion response at the senders, the UDP tunnel SHOULD employ an appropriate congestion control mechanism or circuit breaker mechanism designed for the traffic it carries. Because tunnels are usually bulk-transfer applications as far as the intermediate routers are concerned, the guidelines in Section 3.1.2 apply.

3. A tunnel generates UDP traffic at a volume that does not correspond to the volume of payload traffic, independent of whether the payload traffic is IP based or congestion controlled.

Examples of this class include UDP tunnels that send at a constant rate, increase their transmission rates under loss, for example, due to increasing redundancy when Forward Error Correction is used, or are otherwise unconstrained in their transmission behavior. These specialized uses of UDP for tunneling go beyond the scope of the general guidelines given in this document. The implementer of such specialized tunnels SHOULD carefully consider congestion control in the design of their tunneling mechanism and SHOULD consider use of a circuit breaker mechanism.

The type of encapsulated payload might be identified by a UDP port; identified by an Ethernet Type or IP protocol number. A tunnel SHOULD provide mechanisms to restrict the types of flows that may be carried by the tunnel. For instance, a UDP tunnel designed to carry IP needs to filter out non-IP traffic at the ingress. This is particularly important when a generic tunnel encapsulation is used (e.g., one that encapsulates using an EtherType value). Such tunnels SHOULD provide a mechanism to restrict the types of traffic that are allowed to be encapsulated for a given deployment (see [INT-TUNNELS]).

Designing a tunneling mechanism requires significantly more expertise than needed for many other UDP applications, because tunnels are usually intended to be transparent to the endpoints transmitting over them, so they need to correctly emulate the behavior of an IP link [INT-TUNNELS], for example:

- o Requirements for tunnels that carry or encapsulate using ECN code points [RFC6040].
- o Usage of the IP DSCP field by tunnel endpoints [RFC2983].
- o Encapsulation considerations in the design of tunnels [ENCAP].
- o Usage of ICMP messages [INT-TUNNELS].
- o Handling of fragmentation and packet size for tunnels [INT-TUNNELS].
- o Source port usage for tunnels designed to support equal cost multipath (ECMP) routing (see Section 5.1.1).
- o Guidance on the need to protect headers [INT-TUNNELS] and the use of checksums for IPv6 tunnels (see Section 3.4.1).
- o Support for operations and maintenance [INT-TUNNELS].

At the same time, the tunneled traffic is application traffic like any other from the perspective of the networks the tunnel transmits over. This document only touches upon the congestion control considerations for implementing UDP tunnels; a discussion of other required tunneling behavior is out of scope.

### 3.2. Message Size Guidelines

IP fragmentation lowers the efficiency and reliability of Internet communication. The loss of a single fragment results in the loss of an entire fragmented packet, because even if all other fragments are received correctly, the original packet cannot be reassembled and delivered. This fundamental issue with fragmentation exists for both IPv4 and IPv6.

In addition, some network address translators (NATs) and firewalls drop IP fragments. The network address translation performed by a NAT only operates on complete IP packets, and some firewall policies also require inspection of complete IP packets. Even with these being the case, some NATs and firewalls simply do not implement the necessary reassembly functionality; instead, they choose to drop all fragments. Finally, [RFC4963] documents other issues specific to IPv4 fragmentation.

Due to these issues, an application SHOULD NOT send UDP datagrams that result in IP packets that exceed the Maximum Transmission Unit (MTU) along the path to the destination. Consequently, an application SHOULD either use the path MTU information provided by the IP layer or implement Path MTU Discovery (PMTUD) itself [RFC1191] [RFC1981] [RFC4821] to determine whether the path to a destination will support its desired message size without fragmentation.

However, the ICMP messages that enable path MTU discovery are being increasingly filtered by middleboxes (including Firewalls) [RFC4890]. When the path includes a tunnel, some devices acting as a tunnel ingress discard ICMP messages that originate from network devices over which the tunnel passes, preventing these from reaching the UDP endpoint.

Packetization Layer Path MTU Discovery (PLPMTUD) [RFC4821] does not rely upon network support for ICMP messages and is therefore considered more robust than standard PMTUD. It is not susceptible to "black holing" of ICMP messages. To operate, PLPMTUD requires changes to the way the transport is used: both to transmit probe packets and to account for the loss or success of these probes. This not only updates the PMTU algorithm, it also impacts loss recovery, congestion control, etc. These updated mechanisms can be implemented

within a connection-oriented transport (e.g., TCP, SCTP, DCCP), but they are not a part of UDP; this type of feedback is not typically present for unidirectional applications.

Therefore, PLPMTUD places additional design requirements on a UDP application that wishes to use this method. This is especially true for UDP tunnels, because the overhead of sending probe packets needs to be accounted for and may require adding a congestion control mechanism to the tunnel (see Section 3.1.11) as well as complicating the data path at a tunnel decapsulator.

Applications that do not follow the recommendation to do PMTU/PLPMTUD discovery SHOULD still avoid sending UDP datagrams that would result in IP packets that exceed the path MTU. Because the actual path MTU is unknown, such applications SHOULD fall back to sending messages that are shorter than the default effective MTU for sending (EMTU\_S in [RFC1122]). For IPv4, EMTU\_S is the smaller of 576 bytes and the first-hop MTU [RFC1122]. For IPv6, EMTU\_S is 1280 bytes [RFC2460]. The effective PMTU for a directly connected destination (with no routers on the path) is the configured interface MTU, which could be less than the maximum link payload size. Transmission of minimum-sized UDP datagrams is inefficient over paths that support a larger PMTU, which is a second reason to implement PMTU discovery.

To determine an appropriate UDP payload size, applications MUST subtract the size of the IP header (which includes any IPv4 optional headers or IPv6 extension headers) as well as the length of the UDP header (8 bytes) from the PMTU size. This size, known as the Maximum Segment Size (MSS), can be obtained from the TCP/IP stack [RFC1122].

Applications that do not send messages that exceed the effective PMTU of IPv4 or IPv6 need not implement any of the above mechanisms. Note that the presence of tunnels can cause an additional reduction of the effective PMTU [INT-TUNNELS], so implementing PMTU discovery may be beneficial.

Applications that fragment an application-layer message into multiple UDP datagrams SHOULD perform this fragmentation so that each datagram can be received independently, and be independently retransmitted in the case where an application implements its own reliability mechanisms.

### 3.3. Reliability Guidelines

Application designers are generally aware that UDP does not provide any reliability, e.g., it does not retransmit any lost packets. Often, this is a main reason to consider UDP as a transport protocol. Applications that do require reliable message delivery **MUST** implement an appropriate mechanism themselves.

UDP also does not protect against datagram duplication, i.e., an application may receive multiple copies of the same UDP datagram, with some duplicates arriving potentially much later than the first. Application designers **SHOULD** handle such datagram duplication gracefully, and they may consequently need to implement mechanisms to detect duplicates. Even if UDP datagram reception triggers only idempotent operations, applications may want to suppress duplicate datagrams to reduce load.

Applications that require ordered delivery **MUST** reestablish datagram ordering themselves. The Internet can significantly delay some packets with respect to others, e.g., due to routing transients, intermittent connectivity, or mobility. This can cause reordering, where UDP datagrams arrive at the receiver in an order different from the transmission order.

Applications that use multiple transport ports need to be robust to reordering between sessions. Load-balancing techniques within the network, such as Equal Cost Multipath (ECMP) forwarding can also result in a lack of ordering between different transport sessions, even between the same two network endpoints.

It is important to note that the time by which packets are reordered or after which duplicates can still arrive can be very large. Even more importantly, there is no well-defined upper boundary here. [RFC793] defines the maximum delay a TCP segment should experience -- the Maximum Segment Lifetime (MSL) -- as 2 minutes. No other RFC defines an MSL for other transport protocols or IP itself. The MSL value defined for TCP is conservative enough that it **SHOULD** be used by other protocols, including UDP. Therefore, applications **SHOULD** be robust to the reception of delayed or duplicate packets that are received within this 2-minute interval.

Retransmission of lost packets or messages is a common reliability mechanism. Such retransmissions can increase network load in response to congestion, worsening that congestion. Any application that uses retransmission is responsible for congestion control of its retransmissions (as well as the application's original traffic); hence, it is subject to the Congestion Control guidelines in

Section 3.1. Guidance on the appropriate measurement of RTT in Section 3.1.1 also applies for timers used for retransmission packet-loss detection.

Instead of implementing these relatively complex reliability mechanisms by itself, an application that requires reliable and ordered message delivery SHOULD whenever possible choose an IETF standard transport protocol that provides these features.

### 3.4. Checksum Guidelines

The UDP header includes an optional, 16-bit one's complement checksum that provides an integrity check. These checks are not strong from a coding or cryptographic perspective and are not designed to detect physical-layer errors or malicious modification of the datagram [RFC3819]. Application developers SHOULD implement additional checks where data integrity is important, e.g., through a Cyclic Redundancy Check (CRC) or keyed or non-keyed cryptographic hash included with the data to verify the integrity of an entire object/file sent over the UDP service.

The UDP checksum provides a statistical guarantee that the payload was not corrupted in transit. It also allows the receiver to verify that it was the intended destination of the packet, because it covers the IP addresses, port numbers, and protocol number, and it verifies that the packet is not truncated or padded, because it covers the size field. Therefore, it protects an application against receiving corrupted payload data in place of, or in addition to, the data that was sent. More description of the set of checks performed using the checksum field is provided in Section 3.1 of [RFC6396].

Applications SHOULD enable UDP checksums [RFC1122]. For IPv4, [RFC768] permits an option to disable their use, by setting a zero checksum value. An application is permitted to optionally discard UDP datagrams with a zero checksum [RFC1122].

When UDP is used over IPv6, the UDP checksum is relied upon to protect both the IPv6 and UDP headers from corruption (because IPv6 lacks a checksum) and MUST be used as specified in [RFC2460]. Under specific conditions, a UDP application is allowed to use a zero UDP zero-checksum mode with a tunnel protocol (see Section 3.4.1).

Applications that choose to disable UDP checksums MUST NOT make assumptions regarding the correctness of received data and MUST behave correctly when a UDP datagram is received that was originally sent to a different destination or is otherwise corrupted.

### 3.4.1. IPv6 Zero UDP Checksum

[RFC6935] defines a method that enables use of a zero UDP zero-checksum mode with a tunnel protocol, providing that the method satisfies the requirements in [RFC6936]. The application **MUST** implement mechanisms and/or usage restrictions when enabling this mode. This includes defining the scope for usage and measures to prevent leakage of traffic to other UDP applications (see Appendix A and Section 3.6). These additional design requirements for using a zero IPv6 UDP checksum are not present for IPv4, since the IPv4 header validates information that is not protected in an IPv6 packet. Key requirements are:

- o Use of the UDP checksum with IPv6 **MUST** be the default configuration for all implementations [RFC6935]. The receiving endpoint **MUST** only allow the use of UDP zero-checksum mode for IPv6 on a UDP destination port that is specifically enabled.
- o An application that supports a checksum different than that in [RFC2460] **MUST** comply with all implementation requirements specified in Section 4 of [RFC6936] and with the usage requirements specified in Section 5 of [RFC6936].
- o A UDP application **MUST** check that the source and destination IPv6 addresses are valid for any packets with a UDP zero-checksum and **MUST** discard any packet for which this check fails. To protect from misdelivery, new encapsulation designs **SHOULD** include an integrity check at the transport layer that includes at least the IPv6 header, the UDP header and the shim header for the encapsulation, if any [RFC6936].
- o One way to help satisfy the requirements of [RFC6936] may be to limit the usage of such tunnels, e.g., to constrain traffic to an operator network, as discussed in Section 3.6. The encapsulation defined for MPLS in UDP [RFC7510] chooses this approach.

As in IPv4, IPv6 applications that choose to disable UDP checksums **MUST NOT** make assumptions regarding the correctness of received data and **MUST** behave correctly when a UDP datagram is received that was originally sent to a different destination or is otherwise corrupted.

IPv6 datagrams with a zero UDP checksum will not be passed by any middlebox that validates the checksum based on [RFC2460] or that updates the UDP checksum field, such as NATs or firewalls. Changing this behavior would require such middleboxes to be updated to correctly handle datagrams with zero UDP checksums. To ensure end-to-end robustness, applications that may be deployed in the general Internet **MUST** provide a mechanism to safely fall back to using a

checksum when a path change occurs that redirects a zero UDP checksum flow over a path that includes a middlebox that discards IPv6 datagrams with a zero UDP checksum.

### 3.4.2. UDP-Lite

A special class of applications can derive benefit from having partially damaged payloads delivered, rather than discarded, when using paths that include error-prone links. Such applications can tolerate payload corruption and MAY choose to use the Lightweight User Datagram Protocol (UDP-Lite) [RFC3828] variant of UDP instead of basic UDP. Applications that choose to use UDP-Lite instead of UDP should still follow the congestion control and other guidelines described for use with UDP in Section 3.

UDP-Lite changes the semantics of the UDP "payload length" field to that of a "checksum coverage length" field. Otherwise, UDP-Lite is semantically identical to UDP. The interface of UDP-Lite differs from that of UDP by the addition of a single (socket) option that communicates the checksum coverage length: at the sender, this specifies the intended checksum coverage, with the remaining unprotected part of the payload called the "error-insensitive part". By default, the UDP-Lite checksum coverage extends across the entire datagram. If required, an application may dynamically modify this length value, e.g., to offer greater protection to some messages. UDP-Lite always verifies that a packet was delivered to the intended destination, i.e., always verifies the header fields. Errors in the insensitive part will not cause a UDP datagram to be discarded by the destination. Therefore, applications using UDP-Lite MUST NOT make assumptions regarding the correctness of the data received in the insensitive part of the UDP-Lite payload.

A UDP-Lite sender SHOULD select the minimum checksum coverage to include all sensitive payload information. For example, applications that use the Real-Time Protocol (RTP) [RFC3550] will likely want to protect the RTP header against corruption. Applications, where appropriate, MUST also introduce their own appropriate validity checks for protocol information carried in the insensitive part of the UDP-Lite payload (e.g., internal CRCs).

A UDP-Lite receiver MUST set a minimum coverage threshold for incoming packets that is not smaller than the smallest coverage used by the sender [RFC3828]. The receiver SHOULD select a threshold that is sufficiently large to block packets with an inappropriately short coverage field. This may be a fixed value, or it may be negotiated by an application. UDP-Lite does not provide mechanisms to negotiate the checksum coverage between the sender and receiver. Therefore, this needs to be performed by the application.

Applications can still experience packet loss when using UDP-Lite. The enhancements offered by UDP-Lite rely upon a link being able to intercept the UDP-Lite header to correctly identify the partial coverage required. When tunnels and/or encryption are used, this can result in UDP-Lite datagrams being treated the same as UDP datagrams, i.e., result in packet loss. Use of IP fragmentation can also prevent special treatment for UDP-Lite datagrams, and this is another reason why applications SHOULD avoid IP fragmentation (Section 3.2).

UDP-Lite is supported in some endpoint protocol stacks. Current support for middlebox traversal using UDP-Lite is poor, because UDP-Lite uses a different IPv4 protocol number or IPv6 "next header" value than that used for UDP; therefore, few middleboxes are currently able to interpret UDP-Lite and take appropriate actions when forwarding the packet. This makes UDP-Lite less suited for applications needing general Internet support, until such time as UDP-Lite has achieved better support in middleboxes.

### 3.5. Middlebox Traversal Guidelines

NATs and firewalls are examples of intermediary devices ("middleboxes") that can exist along an end-to-end path. A middlebox typically performs a function that requires it to maintain per-flow state. For connection-oriented protocols, such as TCP, middleboxes snoop and parse the connection-management information, and create and destroy per-flow state accordingly. For a connectionless protocol such as UDP, this approach is not possible. Consequently, middleboxes can create per-flow state when they see a packet that -- according to some local criteria -- indicates a new flow, and destroy the state after some time during which no packets belonging to the same flow have arrived.

Depending on the specific function that the middlebox performs, this behavior can introduce a time-dependency that restricts the kinds of UDP traffic exchanges that will be successful across the middlebox. For example, NATs and firewalls typically define the partial path on one side of them to be interior to the domain they serve, whereas the partial path on their other side is defined to be exterior to that domain. Per-flow state is typically created when the first packet crosses from the interior to the exterior, and while the state is present, NATs and firewalls will forward return traffic. Return traffic that arrives after the per-flow state has timed out is dropped, as is other traffic that arrives from the exterior.

Many applications that use UDP for communication operate across middleboxes without needing to employ additional mechanisms. One example is the Domain Name System (DNS), which has a strict request-response communication pattern that typically completes within seconds.

Other applications may experience communication failures when middleboxes destroy the per-flow state associated with an application session during periods when the application does not exchange any UDP traffic. Applications SHOULD be able to gracefully handle such communication failures and implement mechanisms to re-establish application-layer sessions and state.

For some applications, such as media transmissions, this re-synchronization is highly undesirable, because it can cause user-perceivable playback artifacts. Such specialized applications MAY send periodic keep-alive messages to attempt to refresh middlebox state (e.g., [RFC7675]). It is important to note that keep-alive messages are not recommended for general use -- they are unnecessary for many applications and can consume significant amounts of system and network resources.

An application that needs to employ keep-alive messages to deliver useful service over UDP in the presence of middleboxes SHOULD NOT transmit them more frequently than once every 15 seconds and SHOULD use longer intervals when possible. No common timeout has been specified for per-flow UDP state for arbitrary middleboxes. NATs require a state timeout of 2 minutes or longer [RFC4787]. However, empirical evidence suggests that a significant fraction of currently deployed middleboxes unfortunately use shorter timeouts. The timeout of 15 seconds originates with the Interactive Connectivity Establishment (ICE) protocol [RFC5245]. When an application is deployed in a controlled environment, the deployer SHOULD investigate whether the target environment allows applications to use longer intervals, or whether it offers mechanisms to explicitly control middlebox state timeout durations, for example, using the Port Control Protocol (PCP) [RFC6887], Middlebox Communications (MIDCOM) [RFC3303], Next Steps in Signaling (NSIS) [RFC5973], or Universal Plug and Play (UPnP) [UPnP]. It is RECOMMENDED that applications apply slight random variations ("jitter") to the timing of keep-alive transmissions, to reduce the potential for persistent synchronization between keep-alive transmissions from different hosts [RFC7675].

Sending keep-alive messages is not a substitute for implementing a mechanism to recover from broken sessions. Like all UDP datagrams, keep-alive messages can be delayed or dropped, causing middlebox state to time out. In addition, the congestion control guidelines in Section 3.1 cover all UDP transmissions by an application, including the transmission of middlebox keep-alive messages. Congestion control may thus lead to delays or temporary suspension of keep-alive transmission.

Keep-alive messages are NOT RECOMMENDED for general use. They are unnecessary for many applications and may consume significant resources. For example, on battery-powered devices, if an application needs to maintain connectivity for long periods with little traffic, the frequency at which keep-alive messages are sent can become the determining factor that governs power consumption, depending on the underlying network technology.

Because many middleboxes are designed to require keep-alive messages for TCP connections at a frequency that is much lower than that needed for UDP, this difference alone can often be sufficient to prefer TCP over UDP for these deployments. On the other hand, there is anecdotal evidence that suggests that direct communication through middleboxes, e.g., by using ICE [RFC5245], does succeed less often with TCP than with UDP. The trade-offs between different transport protocols -- especially when it comes to middlebox traversal -- deserve careful analysis.

UDP applications that could be deployed in the Internet need to be designed understanding that there are many variants of middlebox behavior, and although UDP is connectionless, middleboxes often maintain state for each UDP flow. Using multiple UDP flows can consume available state space and also can lead to changes in the way the middlebox handles subsequent packets (either to protect its internal resources, or to prevent perceived misuse). The probability of path failure can increase when applications use multiple UDP flows in parallel (see Section 5.1.2 for recommendations on usage of multiple ports).

### 3.6. Limited Applicability and Controlled Environments

Two different types of applicability have been identified for the specification of IETF applications that utilize UDP:

General Internet. By default, IETF specifications target deployment on the general Internet. Experience has shown that successful protocols developed in one specific context or for a particular application tend to become used in a wider range of contexts. For example, a protocol with an initial deployment within a local area

network may subsequently be used over a virtual network that traverses the Internet, or in the Internet in general. Applications designed for general Internet use may experience a range of network device behaviors and, in particular, should consider whether applications need to operate over paths that may include middleboxes.

Controlled Environment. A protocol/encapsulation/tunnel could be designed to be used only within a controlled environment. For example, an application designed for use by a network operator might only be deployed within the network of that single network operator or on networks of an adjacent set of cooperating network operators. The application traffic may then be managed to avoid congestion, rather than relying on built-in mechanisms, which are required when operating over the general Internet. Applications that target a limited applicability use case may be able to take advantage of specific hardware (e.g., carrier-grade equipment) or underlying protocol features of the subnetwork over which they are used.

Specifications addressing a limited applicability use case or a controlled environment SHOULD identify how, in their restricted deployment, a level of safety is provided that is equivalent to that of a protocol designed for operation over the general Internet (e.g., a design based on extensive experience with deployments of particular methods that provide features that cannot be expected in general Internet equipment and the robustness of the design of MPLS to corruption of headers both helped justify use of an alternate UDP integrity check [RFC7510]).

An IETF specification targeting a controlled environment is expected to provide an applicability statement that restricts the application traffic to the controlled environment, and it would be expected to describe how methods can be provided to discourage or prevent escape of corrupted packets from the environment (for example, Section 5 of [RFC7510]).

#### 4. Multicast UDP Usage Guidelines

This section complements Section 3 by providing additional guidelines that are applicable to multicast and broadcast usage of UDP.

Multicast and broadcast transmission [RFC1112] usually employ the UDP transport protocol, although they may be used with other transport protocols (e.g., UDP-Lite).

There are currently two models of multicast delivery: the Any-Source Multicast (ASM) model as defined in [RFC1112] and the Source-Specific Multicast (SSM) model as defined in [RFC4607]. ASM group members will receive all data sent to the group by any source, while SSM constrains the distribution tree to only one single source.

Specialized classes of applications also use UDP for IP multicast or broadcast [RFC919]. The design of such specialized applications requires expertise that goes beyond simple, unicast-specific guidelines, since these senders may transmit to potentially very many receivers across potentially very heterogeneous paths at the same time, which significantly complicates congestion control, flow control, and reliability mechanisms.

This section provides guidance on multicast and broadcast UDP usage. Use of broadcast by an application is normally constrained by routers to the local subnetwork. However, use of tunneling techniques and proxies can and does result in some broadcast traffic traversing Internet paths. These guidelines therefore also apply to broadcast traffic.

The IETF has defined a reliable multicast framework [RFC3048] and several building blocks to aid the designers of multicast applications, such as [RFC3738] or [RFC4654].

Senders to anycast destinations must be aware that successive messages sent to the same anycast IP address may be delivered to different anycast nodes, i.e., arrive at different locations in the topology.

Most UDP tunnels that carry IP multicast traffic use a tunnel encapsulation with a unicast destination address, such as Automatic Multicast Tunneling [RFC7450]. These MUST follow the same requirements as a tunnel carrying unicast data (see Section 3.1.11). There are deployment cases and solutions where the outer header of a UDP tunnel contains a multicast destination address, such as [RFC6513]. These cases are primarily deployed in controlled environments over reserved capacity, often operating within a single administrative domain, or between two domains over a bilaterally agreed upon path with reserved capacity, and so congestion control is OPTIONAL, but circuit breaker techniques are still RECOMMENDED in order to restore some degree of service should the offered load exceed the reserved capacity (e.g., due to misconfiguration).

#### 4.1. Multicast Congestion Control Guidelines

Unicast congestion-controlled transport mechanisms are often not applicable to multicast distribution services, or simply do not scale to large multicast trees, since they require bidirectional communication and adapt the sending rate to accommodate the network conditions to a single receiver. In contrast, multicast distribution trees may fan out to massive numbers of receivers, which limits the scalability of an in-band return channel to control the sending rate, and the one-to-many nature of multicast distribution trees prevents adapting the rate to the requirements of an individual receiver. For this reason, generating TCP-compatible aggregate flow rates for Internet multicast data, either native or tunneled, is the responsibility of the application implementing the congestion control.

Applications using multicast SHOULD provide appropriate congestion control. Multicast congestion control needs to be designed using mechanisms that are robust to the potential heterogeneity of both the multicast distribution tree and the receivers belonging to a group. Heterogeneity may manifest itself in some receivers experiencing more loss than others, higher delay, and/or less ability to respond to network conditions. Congestion control is particularly important for any multicast session where all or part of the multicast distribution tree spans an access network (e.g., a home gateway). Two styles of congestion control have been defined in the RFC Series:

- o Feedback-based congestion control, in which the sender receives multicast or unicast UDP messages from the receivers allowing it to assess the level of congestion and then adjust the sender rate(s) (e.g., [RFC5740],[RFC4654]). Multicast methods may operate on longer timescales than for unicast (e.g., due to the higher group RTT of a heterogeneous group). A control method could decide not to reduce the rate of the entire multicast group in response to a control message received from a single receiver (e.g., a sender could set a minimum rate and decide to request a congested receiver to leave the multicast group and could also decide to distribute content to these congested receivers at a lower rate using unicast congestion control).
- o Receiver-driven congestion control, which does not require a receiver to send explicit UDP control messages for congestion control (e.g., [RFC3738], [RFC5775]). Instead, the sender distributes the data across multiple IP multicast groups (e.g., using a set of {S,G} channels). Each receiver determines its own level of congestion and controls its reception rate using only multicast join/leave messages sent in the network control plane. This method scales to arbitrary large groups of receivers.

Any multicast-enabled receiver may attempt to join and receive traffic from any group. This may imply the need for rate limits on individual receivers or the aggregate multicast service. Note, at the transport layer, there is no way to prevent a join message propagating to the next-hop router.

Some classes of multicast applications support applications that can monitor the user-level quality of the transfer at the receiver. Applications that can detect a significant reduction in user quality SHOULD regard this as a congestion signal (e.g., to leave a group using layered multicast encoding); if not, they SHOULD use this signal to provide a circuit breaker to terminate the flow by leaving the multicast group.

#### 4.1.1. Bulk-Transfer Multicast Applications

Applications that perform bulk transmission of data over a multicast distribution tree, i.e., applications that exchange more than a few UDP datagrams per RTT, SHOULD implement a method for congestion control. The currently RECOMMENDED IETF methods are as follows: Asynchronous Layered Coding (ALC) [RFC5775], TCP-Friendly Multicast Congestion Control (TFMCC) [RFC4654], Wave and Equation Based Rate Control (WEBRC) [RFC3738], NACK-Oriented Reliable Multicast (NORM) transport protocol [RFC5740], File Delivery over Unidirectional Transport (FLUTE) [RFC6726], Real Time Protocol/Control Protocol (RTP/RTCP) [RFC3550].

An application can alternatively implement another congestion control scheme following the guidelines of [RFC2887] and utilizing the framework of [RFC3048]. Bulk-transfer applications that choose not to implement [RFC4654], [RFC5775], [RFC3738], [RFC5740], [RFC6726], or [RFC3550] SHOULD implement a congestion control scheme that results in bandwidth use that competes fairly with TCP within an order of magnitude.

Section 2 of [RFC3551] states that multimedia applications SHOULD monitor the packet-loss rate to ensure that it is within acceptable parameters. Packet loss is considered acceptable if a TCP flow across the same network path under the same network conditions would achieve an average throughput, measured on a reasonable timescale, that is not less than that of the UDP flow. The comparison to TCP cannot be specified exactly, but is intended as an "order-of-magnitude" comparison in timescale and throughput.

#### 4.1.2. Low Data-Volume Multicast Applications

All the recommendations in Section 3.1.3 are also applicable to low data-volume multicast applications.

#### 4.2. Message Size Guidelines for Multicast

A multicast application SHOULD NOT send UDP datagrams that result in IP packets that exceed the effective MTU as described in Section 3 of [RFC6807]. Consequently, an application SHOULD either use the effective MTU information provided by the "Population Count Extensions to Protocol Independent Multicast (PIM)" [RFC6807] or implement path MTU discovery itself (see Section 3.2) to determine whether the path to each destination will support its desired message size without fragmentation.

#### 5. Programming Guidelines

The de facto standard application programming interface (API) for TCP/IP applications is the "sockets" interface [POSIX]. Some platforms also offer applications the ability to directly assemble and transmit IP packets through "raw sockets" or similar facilities. This is a second, more cumbersome method of using UDP. The guidelines in this document cover all such methods through which an application may use UDP. Because the sockets API is by far the most common method, the remainder of this section discusses it in more detail.

Although the sockets API was developed for UNIX in the early 1980s, a wide variety of non-UNIX operating systems also implement it. The sockets API supports both IPv4 and IPv6 [RFC3493]. The UDP sockets API differs from that for TCP in several key ways. Because application programmers are typically more familiar with the TCP sockets API, this section discusses these differences. [STEVENS] provides usage examples of the UDP sockets API.

UDP datagrams may be directly sent and received, without any connection setup. Using the sockets API, applications can receive packets from more than one IP source address on a single UDP socket. Some servers use this to exchange data with more than one remote host through a single UDP socket at the same time. Many applications need to ensure that they receive packets from a particular source address; these applications MUST implement corresponding checks at the application layer or explicitly request that the operating system filter the received packets.

Many operating systems also allow a UDP socket to be connected, i.e., to bind a UDP socket to a specific pair of addresses and ports. This is similar to the corresponding TCP sockets API functionality. However, for UDP, this is only a local operation that serves to simplify the local send/receive functions and to filter the traffic for the specified addresses and ports. Binding a UDP socket does not establish a connection -- UDP does not notify the remote end when a

local UDP socket is bound. Binding a socket also allows configuring options that affect the UDP or IP layers, for example, use of the UDP checksum or the IP Timestamp option. On some stacks, a bound socket also allows an application to be notified when ICMP error messages are received for its transmissions [RFC1122].

If a client/server application executes on a host with more than one IP interface, the application SHOULD send any UDP responses with an IP source address that matches the IP destination address of the UDP datagram that carried the request (see [RFC1122], Section 4.1.3.5). Many middleboxes expect this transmission behavior and drop replies that are sent from a different IP address, as explained in Section 3.5.

A UDP receiver can receive a valid UDP datagram with a zero-length payload. Note that this is different from a return value of zero from a read() socket call, which for TCP indicates the end of the connection.

UDP provides no flow-control, i.e., the sender at any given time does not know whether the receiver is able to handle incoming transmissions. This is another reason why UDP-based applications need to be robust in the presence of packet loss. This loss can also occur within the sending host, when an application sends data faster than the line rate of the outbound network interface. It can also occur at the destination, where receive calls fail to return all the data that was sent when the application issues them too infrequently (i.e., such that the receive buffer overflows). Robust flow control mechanisms are difficult to implement, which is why applications that need this functionality SHOULD consider using a full-featured transport protocol such as TCP.

When an application closes a TCP, SCTP, or DCCP socket, the transport protocol on the receiving host is required to maintain TIME-WAIT state. This prevents delayed packets from the closed connection instance from being mistakenly associated with a later connection instance that happens to reuse the same IP address and port pairs. The UDP protocol does not implement such a mechanism. Therefore, UDP-based applications need to be robust to reordering and delay. One application may close a socket or terminate, followed in time by another application receiving on the same port. This later application may then receive packets intended for the first application that were delayed in the network.

### 5.1. Using UDP Ports

The rules and procedures for the management of the "Service Name and Transport Protocol Port Number Registry" are specified in [RFC6335]. Recommendations for use of UDP ports are provided in [RFC7605].

A UDP sender SHOULD NOT use a source port value of zero. A source port number that cannot be easily determined from the address or payload type provides protection at the receiver from data injection attacks by off-path devices. A UDP receiver SHOULD NOT bind to port zero.

Applications SHOULD implement receiver port and address checks at the application layer or explicitly request that the operating system filter the received packets to prevent receiving packets with an arbitrary port. This measure is designed to provide additional protection from data injection attacks from an off-path source (where the port values may not be known).

Applications SHOULD provide a check that protects from off-path data injection, avoiding an application receiving packets that were created by an unauthorized third party. TCP stacks commonly use a randomized source port to provide this protection [RFC6056]; UDP applications should follow the same technique. Middleboxes and end systems often make assumptions about the system ports or user ports; hence, it is recommended to use randomized ports in the Dynamic and/or Private Port range. Setting a "randomized" source port also provides greater assurance that reported ICMP errors originate from network systems on the path used by a particular flow. Some UDP applications choose to use a predetermined value for the source port (including some multicast applications), these applications need to therefore employ a different technique. Protection from off-path data attacks can also be provided by randomizing the initial value of another protocol field within the datagram payload, and checking the validity of this field at the receiver (e.g., RTP has random initial sequence number and random media timestamp offsets [RFC3550]).

When using multicast, IP routers perform a reverse-path forwarding (RPF) check for each multicast packet. This provides protection from off-path data injection, restricting opportunities to forge a packet's source address. When a receiver joins a multicast group and filters based on the source address the filter verifies the sender's IP address. This is always the case when using an SSM {S,G} channel.

#### 5.1.1. Usage of UDP for Source Port Entropy and the IPv6 Flow Label

Some applications use the UDP datagram header as a source of entropy for network devices that implement ECMP [RFC6438]. A UDP tunnel application targeting this usage encapsulates an inner packet using UDP, where the UDP source port value forms a part of the entropy that can be used to balance forwarding of network traffic by the devices that use ECMP. A sending tunnel endpoint selects a source port value in the UDP datagram header that is computed from the inner flow information (e.g., the encapsulated packet headers). To provide sufficient entropy, the sending tunnel endpoint maps the encapsulated traffic to one of a range of UDP source values. The value SHOULD be within the ephemeral port range, i.e., 49152 to 65535, where the high order two bits of the port are set to one. The available source port entropy of 14 bits (using the ephemeral port range) plus the outer IP addresses seems sufficient for entropy for most ECMP applications [ENCAP].

To avoid reordering within an IP flow, the same UDP source port value SHOULD be used for all packets assigned to an encapsulated flow (e.g., using a hash of the relevant headers). The entropy mapping for a flow MAY change over the lifetime of the encapsulated flow [ENCAP]. For instance, this could be changed as a Denial of Service (DOS) mitigation, or as a means to effect routing through the ECMP network. However, the source port selected for a flow SHOULD NOT change more than once in every thirty seconds (e.g., as in [RFC8086]).

The use of the source port field for entropy has several side effects that need to be considered, including:

- o It can increase the probability of misdelivery of corrupted packets, which increases the need for checksum computation or an equivalent mechanism to protect other UDP applications from misdelivery errors Section 3.4.
- o It is expected to reduce the probability of successful middlebox traversal Section 3.5. This use of the source port field will often not be suitable for applications targeting deployment in the general Internet.
- o It can prevent the field being usable to protect from off-path attacks (described in Section 5.1). Designers therefore need to consider other mechanisms to provide equivalent protection (e.g., to restrict use to a controlled environment [RFC7510] Section 3.6).

The UDP source port number field has also been leveraged to produce entropy with IPv6. However, in the case of IPv6, the "flow label" [RFC6437] may also alternatively be used to provide entropy for load balancing [RFC6438]. This use of the flow label for load balancing is consistent with the definition of the field, although further clarity was needed to ensure the field can be consistently used for this purpose. Therefore, an updated IPv6 flow label [RFC6437] and ECMP routing [RFC6438] usage was specified.

To ensure future opportunities to use the flow label, UDP applications SHOULD set the flow label field, even when an entropy value is also set in the source port field (e.g., An IPv6 tunnel endpoint could copy the source port flow entropy value to the IPv6 flow label field [RFC8086]). Router vendors are encouraged to start using the IPv6 flow label as a part of the flow hash, providing support for IP-level ECMP without requiring use of UDP. The end-to-end use of flow labels for load balancing is a long-term solution. Even if the usage of the flow label has been clarified, there will be a transition time before a significant proportion of endpoints start to assign a good quality flow label to the flows that they originate. The use of load balancing using the transport header fields will likely continue until widespread deployment is finally achieved.

#### 5.1.2. Applications Using Multiple UDP Ports

A single application may exchange several types of data. In some cases, this may require multiple UDP flows (e.g., multiple sets of flows, identified by different five-tuples). [RFC6335] recommends application developers not to apply to IANA to be assigned multiple well-known ports (user or system). It does not discuss the implications of using multiple flows with the same well-known port or pairs of dynamic ports (e.g., identified by a service name or signaling protocol).

Use of multiple flows can affect the network in several ways:

- o Starting a series of successive connections can increase the number of state bindings in middleboxes (e.g., NAT or Firewall) along the network path. UDP-based middlebox traversal usually relies on timeouts to remove old state, since middleboxes are unaware when a particular flow ceases to be used by an application.
- o Using several flows at the same time may result in seeing different network characteristics for each flow. It cannot be assumed both follow the same path (e.g., when ECMP is used, traffic is intentionally hashed onto different parallel paths based on the port numbers).

- o Using several flows can also increase the occupancy of a binding or lookup table in a middlebox (e.g., NAT or Firewall), which may cause the device to change the way it manages the flow state.
- o Further, using excessive numbers of flows can degrade the ability of a unicast congestion control to react to congestion events, unless the congestion state is shared between all flows in a session. A receiver-driven multicast congestion control requires the sending application to distribute its data over a set of IP multicast groups, each receiver is therefore expected to receive data from a modest number of simultaneously active UDP ports.

Therefore, applications MUST NOT assume consistent behavior of middleboxes when multiple UDP flows are used; many devices respond differently as the number of used ports increases. Using multiple flows with different QoS requirements requires applications to verify that the expected performance is achieved using each individual flow (five-tuple), see Section 3.1.9.

## 5.2. ICMP Guidelines

Applications can utilize information about ICMP error messages that the UDP layer passes up for a variety of purposes [RFC1122]. Applications SHOULD appropriately validate the payload of ICMP messages to ensure these are received in response to transmitted traffic (i.e., a reported error condition that corresponds to a UDP datagram actually sent by the application). This requires context, such as local state about communication instances to each destination, that although readily available in connection-oriented transport protocols is not always maintained by UDP-based applications. Note that not all platforms have the necessary APIs to support this validation, and some platforms already perform this validation internally before passing ICMP information to the application.

Any application response to ICMP error messages SHOULD be robust to temporary routing failures (sometimes called "soft errors"), e.g., transient ICMP "unreachable" messages ought to not normally cause a communication abort.

ICMP messages are being increasingly filtered by middleboxes. A UDP application therefore SHOULD NOT rely on their delivery for correct and safe operation.

## 6. Security Considerations

UDP does not provide communications security. Applications that need to protect their communications against eavesdropping, tampering, or message forgery SHOULD employ end-to-end security services provided by other IETF protocols.

UDP applications SHOULD provide protection from off-path data injection attacks using a randomized source port or equivalent technique (see Section 5.1).

Applications that respond to short requests with potentially large responses are a potential vector for amplification attacks, and SHOULD take steps to minimize their potential for being abused as part of a DoS attack. That could mean authenticating the sender before responding; noting that the source IP address of a request is not a useful authenticator, because it can easily be spoofed. Or it may mean otherwise limiting the cases where short unauthenticated requests produce large responses. Applications MAY also want to offer ways to limit the number of requests they respond to in a time interval, in order to cap the bandwidth they consume.

One option for securing UDP communications is with IPsec [RFC4301], which can provide authentication for flows of IP packets through the Authentication Header (AH) [RFC4302] and encryption and/or authentication through the Encapsulating Security Payload (ESP) [RFC4303]. Applications use the Internet Key Exchange (IKE) [RFC7296] to configure IPsec for their sessions. Depending on how IPsec is configured for a flow, it can authenticate or encrypt the UDP headers as well as UDP payloads. If an application only requires authentication, ESP with no encryption but with authentication is often a better option than AH, because ESP can operate across middleboxes. An application that uses IPsec requires the support of an operating system that implements the IPsec protocol suite, and the network path must permit IKE and IPsec traffic. This may become more common with IPv6 deployments [RFC6092].

Although it is possible to use IPsec to secure UDP communications, not all operating systems support IPsec or allow applications to easily configure it for their flows. A second option for securing UDP communications is through Datagram Transport Layer Security (DTLS) [RFC6347][RFC7525]. DTLS provides communication privacy by encrypting UDP payloads. It does not protect the UDP headers. Applications can implement DTLS without relying on support from the operating system.

Many other options for authenticating or encrypting UDP payloads exist. For example, the GSS-API security framework [RFC2743] or Cryptographic Message Syntax (CMS) [RFC5652] could be used to protect UDP payloads. There exist a number of security options for RTP [RFC3550] over UDP, especially to accomplish key-management, see [RFC7201]. These options covers many usages, including point-to-point, centralized group communication as well as multicast. In some applications, a better solution is to protect larger stand-alone objects, such as files or messages, instead of individual UDP payloads. In these situations, CMS [RFC5652], S/MIME [RFC5751] or OpenPGP [RFC4880] could be used. In addition, there are many non-IETF protocols in this area.

Like congestion control mechanisms, security mechanisms are difficult to design and implement correctly. It is hence RECOMMENDED that applications employ well-known standard security mechanisms such as DTLS or IPsec, rather than inventing their own.

The Generalized TTL Security Mechanism (GTSM) [RFC5082] may be used with UDP applications when the intended endpoint is on the same link as the sender. This lightweight mechanism allows a receiver to filter unwanted packets.

In terms of congestion control, [RFC2309] and [RFC2914] discuss the dangers of congestion-unresponsive flows to the Internet. [RFC8084] describes methods that can be used to set a performance envelope that can assist in preventing congestion collapse in the absence of congestion control or when the congestion control fails to react to congestion events. This document provides guidelines to designers of UDP-based applications to congestion-control their transmissions, and does not raise any additional security concerns.

Some network operators have experienced surges of UDP attack traffic that are multiple orders of magnitude above the baseline traffic rate for UDP. This can motivate operators to limit the data rate or packet rate of UDP traffic. This may in turn limit the throughput that an application can achieve using UDP and could also result in higher packet loss for UDP traffic that would not be experienced if other transport protocols had been used.

A UDP application with a long-lived association between the sender and receiver, ought to be designed so that the sender periodically checks that the receiver still wants ("consents") to receive traffic and need to be designed to stop if there is no explicit confirmation of this [RFC7675]. Applications that require communications in two directions to implement protocol functions (such as reliability or

congestion control) will need to independently check both directions of communication, and may have to exchange keep-alive messages to traverse middleboxes (see Section 3.5).

## 7. Summary

This section summarizes the key guidelines made in Sections 3 - 6 in a tabular format (Table 1) for easy referencing.

Recommendation	Section
MUST tolerate a wide range of Internet path conditions SHOULD use a full-featured transport (e.g., TCP)	3
SHOULD control rate of transmission SHOULD perform congestion control over all traffic	3.1
for bulk transfers, SHOULD consider implementing TFRC else, SHOULD in other ways use bandwidth similar to TCP	3.1.2
for non-bulk transfers, SHOULD measure RTT and transmit max. 1 datagram/RTT else, SHOULD send at most 1 datagram every 3 seconds SHOULD back-off retransmission timers following loss	3.1.3 3.1.1
SHOULD provide mechanisms to regulate the bursts of transmission	3.1.6
MAY implement ECN; a specific set of application mechanisms are REQUIRED if ECN is used.	3.1.7
for DiffServ, SHOULD NOT rely on implementation of PHBs	3.1.8
for QoS-enabled paths, MAY choose not to use CC	3.1.9
SHOULD NOT rely solely on QoS for their capacity non-CC controlled flows SHOULD implement a transport circuit breaker MAY implement a circuit breaker for other applications	3.1.10
for tunnels carrying IP traffic, SHOULD NOT perform congestion control MUST correctly process the IP ECN field	3.1.11

for non-IP tunnels or rate not determined by traffic, SHOULD perform CC or use circuit breaker SHOULD restrict types of traffic transported by the tunnel	3.1.11
SHOULD NOT send datagrams that exceed the PMTU, i.e., SHOULD discover PMTU or send datagrams < minimum PMTU; Specific application mechanisms are REQUIRED if PLPMTUD is used.	3.2
SHOULD handle datagram loss, duplication, reordering SHOULD be robust to delivery delays up to 2 minutes	3.3
SHOULD enable IPv4 UDP checksum	3.4
SHOULD enable IPv6 UDP checksum; Specific application mechanisms are REQUIRED if a zero IPv6 UDP checksum is used.	3.4.1
SHOULD provide protection from off-path attacks else, MAY use UDP-Lite with suitable checksum coverage	5.1 3.4.2
SHOULD NOT always send middlebox keep-alive messages MAY use keep-alives when needed (min. interval 15 sec)	3.5
Applications specified for use in limited use (or controlled environments) SHOULD identify equivalent mechanisms and describe their use case.	3.6
Bulk-multicast apps SHOULD implement congestion control	4.1.1
Low volume multicast apps SHOULD implement congestion control	4.1.2
Multicast apps SHOULD use a safe PMTU	4.2
SHOULD avoid using multiple ports MUST check received IP source address	5.1.2
SHOULD validate payload in ICMP messages	5.2
SHOULD use a randomized source port or equivalent technique, and, for client/server applications, SHOULD send responses from source address matching request 5.1	6
SHOULD use standard IETF security protocols when needed	6

Table 1: Summary of Recommendations

## 8. References

### 8.1. Normative References

- [RFC768] Postel, J., "User Datagram Protocol", STD 6, RFC 768, DOI 10.17487/RFC0768, August 1980, <<http://www.rfc-editor.org/info/rfc768>>.
- [RFC793] Postel, J., "Transmission Control Protocol", STD 7, RFC 793, DOI 10.17487/RFC0793, September 1981, <<http://www.rfc-editor.org/info/rfc793>>.
- [RFC1122] Braden, R., Ed., "Requirements for Internet Hosts - Communication Layers", STD 3, RFC 1122, DOI 10.17487/RFC1122, October 1989, <<http://www.rfc-editor.org/info/rfc1122>>.
- [RFC1191] Mogul, J. and S. Deering, "Path MTU discovery", RFC 1191, DOI 10.17487/RFC1191, November 1990, <<http://www.rfc-editor.org/info/rfc1191>>.
- [RFC1981] McCann, J., Deering, S., and J. Mogul, "Path MTU Discovery for IP version 6", RFC 1981, DOI 10.17487/RFC1981, August 1996, <<http://www.rfc-editor.org/info/rfc1981>>.
- [RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, DOI 10.17487/RFC2119, March 1997, <<http://www.rfc-editor.org/info/rfc2119>>.
- [RFC2460] Deering, S. and R. Hinden, "Internet Protocol, Version 6 (IPv6) Specification", RFC 2460, DOI 10.17487/RFC2460, December 1998, <<http://www.rfc-editor.org/info/rfc2460>>.
- [RFC2914] Floyd, S., "Congestion Control Principles", BCP 41, RFC 2914, DOI 10.17487/RFC2914, September 2000, <<http://www.rfc-editor.org/info/rfc2914>>.
- [RFC3828] Larzon, L-A., Degermark, M., Pink, S., Jonsson, L-E., Ed., and G. Fairhurst, Ed., "The Lightweight User Datagram Protocol (UDP-Lite)", RFC 3828, DOI 10.17487/RFC3828, July 2004, <<http://www.rfc-editor.org/info/rfc3828>>.
- [RFC4787] Audet, F., Ed. and C. Jennings, "Network Address Translation (NAT) Behavioral Requirements for Unicast UDP", BCP 127, RFC 4787, DOI 10.17487/RFC4787, January 2007, <<http://www.rfc-editor.org/info/rfc4787>>.

- [RFC4821] Mathis, M. and J. Heffner, "Packetization Layer Path MTU Discovery", RFC 4821, DOI 10.17487/RFC4821, March 2007, <<http://www.rfc-editor.org/info/rfc4821>>.
- [RFC5348] Floyd, S., Handley, M., Padhye, J., and J. Widmer, "TCP Friendly Rate Control (TFRC): Protocol Specification", RFC 5348, DOI 10.17487/RFC5348, September 2008, <<http://www.rfc-editor.org/info/rfc5348>>.
- [RFC5405] Eggert, L. and G. Fairhurst, "Unicast UDP Usage Guidelines for Application Designers", BCP 145, RFC 5405, DOI 10.17487/RFC5405, November 2008, <<http://www.rfc-editor.org/info/rfc5405>>.
- [RFC6040] Briscoe, B., "Tunnelling of Explicit Congestion Notification", RFC 6040, DOI 10.17487/RFC6040, November 2010, <<http://www.rfc-editor.org/info/rfc6040>>.
- [RFC6298] Paxson, V., Allman, M., Chu, J., and M. Sargent, "Computing TCP's Retransmission Timer", RFC 6298, DOI 10.17487/RFC6298, June 2011, <<http://www.rfc-editor.org/info/rfc6298>>.
- [RFC8084] Fairhurst, G., "Network Transport Circuit Breakers", BCP 208, RFC 8084, DOI 10.17487/RFC8084, March 2017, <<http://www.rfc-editor.org/info/rfc8084>>.

## 8.2. Informative References

- [ALLMAN] Allman, M. and E. Blanton, "Notes on burst mitigation for transport protocols", March 2005.
- [BEHAVE-APP]  
Ford, B., "Application Design Guidelines for Traversal through Network Address Translators", Work in Progress, draft-ford-behave-app-05, March 2007.
- [ENCAP] Nordmark, E., Ed., Tian, A., Gross, J., Hudson, J., Kreeger, L., Garg, P., Thaler, P., and T. Herbert, "Encapsulation Considerations", Work in Progress, draft-ietf-rtgwg-dt-encap-02, October 2016.
- [FABER] Faber, T., Touch, J., and W. Yue, "The TIME-WAIT State in TCP and Its Effect on Busy Servers", Proc. IEEE Infocom, March 1999.

## [INT-TUNNELS]

Touch, J. and W. Townsley, "IP Tunnels in the Internet Architecture", Work in Progress, draft-ietf-intarea-tunnels-03, July 2016.

[POSIX] IEEE Std. 1003.1-2001, , "Standard for Information Technology - Portable Operating System Interface (POSIX)", Open Group Technical Standard: Base Specifications Issue 6, ISO/IEC 9945:2002, December 2001.

[RFC919] Mogul, J., "Broadcasting Internet Datagrams", STD 5, RFC 919, DOI 10.17487/RFC0919, October 1984, <<http://www.rfc-editor.org/info/rfc919>>.

[RFC1112] Deering, S., "Host extensions for IP multicasting", STD 5, RFC 1112, DOI 10.17487/RFC1112, August 1989, <<http://www.rfc-editor.org/info/rfc1112>>.

[RFC1536] Kumar, A., Postel, J., Neuman, C., Danzig, P., and S. Miller, "Common DNS Implementation Errors and Suggested Fixes", RFC 1536, DOI 10.17487/RFC1536, October 1993, <<http://www.rfc-editor.org/info/rfc1536>>.

[RFC1546] Partridge, C., Mendez, T., and W. Milliken, "Host Anycasting Service", RFC 1546, DOI 10.17487/RFC1546, November 1993, <<http://www.rfc-editor.org/info/rfc1546>>.

[RFC2309] Braden, B., Clark, D., Crowcroft, J., Davie, B., Deering, S., Estrin, D., Floyd, S., Jacobson, V., Minshall, G., Partridge, C., Peterson, L., Ramakrishnan, K., Shenker, S., Wroclawski, J., and L. Zhang, "Recommendations on Queue Management and Congestion Avoidance in the Internet", RFC 2309, DOI 10.17487/RFC2309, April 1998, <<http://www.rfc-editor.org/info/rfc2309>>.

[RFC2475] Blake, S., Black, D., Carlson, M., Davies, E., Wang, Z., and W. Weiss, "An Architecture for Differentiated Services", RFC 2475, DOI 10.17487/RFC2475, December 1998, <<http://www.rfc-editor.org/info/rfc2475>>.

[RFC2675] Borman, D., Deering, S., and R. Hinden, "IPv6 Jumbograms", RFC 2675, DOI 10.17487/RFC2675, August 1999, <<http://www.rfc-editor.org/info/rfc2675>>.

[RFC2743] Linn, J., "Generic Security Service Application Program Interface Version 2, Update 1", RFC 2743, DOI 10.17487/RFC2743, January 2000, <<http://www.rfc-editor.org/info/rfc2743>>.

- [RFC2887] Handley, M., Floyd, S., Whetten, B., Kermode, R., Vicisano, L., and M. Luby, "The Reliable Multicast Design Space for Bulk Data Transfer", RFC 2887, DOI 10.17487/RFC2887, August 2000, <<http://www.rfc-editor.org/info/rfc2887>>.
- [RFC2983] Black, D., "Differentiated Services and Tunnels", RFC 2983, DOI 10.17487/RFC2983, October 2000, <<http://www.rfc-editor.org/info/rfc2983>>.
- [RFC3048] Whetten, B., Vicisano, L., Kermode, R., Handley, M., Floyd, S., and M. Luby, "Reliable Multicast Transport Building Blocks for One-to-Many Bulk-Data Transfer", RFC 3048, DOI 10.17487/RFC3048, January 2001, <<http://www.rfc-editor.org/info/rfc3048>>.
- [RFC3124] Balakrishnan, H. and S. Seshan, "The Congestion Manager", RFC 3124, DOI 10.17487/RFC3124, June 2001, <<http://www.rfc-editor.org/info/rfc3124>>.
- [RFC3168] Ramakrishnan, K., Floyd, S., and D. Black, "The Addition of Explicit Congestion Notification (ECN) to IP", RFC 3168, DOI 10.17487/RFC3168, September 2001, <<http://www.rfc-editor.org/info/rfc3168>>.
- [RFC3261] Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M., and E. Schooler, "SIP: Session Initiation Protocol", RFC 3261, DOI 10.17487/RFC3261, June 2002, <<http://www.rfc-editor.org/info/rfc3261>>.
- [RFC3303] Srisuresh, P., Kuthan, J., Rosenberg, J., Molitor, A., and A. Rayhan, "Middlebox communication architecture and framework", RFC 3303, DOI 10.17487/RFC3303, August 2002, <<http://www.rfc-editor.org/info/rfc3303>>.
- [RFC3493] Gilligan, R., Thomson, S., Bound, J., McCann, J., and W. Stevens, "Basic Socket Interface Extensions for IPv6", RFC 3493, DOI 10.17487/RFC3493, February 2003, <<http://www.rfc-editor.org/info/rfc3493>>.
- [RFC3550] Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications", STD 64, RFC 3550, DOI 10.17487/RFC3550, July 2003, <<http://www.rfc-editor.org/info/rfc3550>>.

- [RFC3551] Schulzrinne, H. and S. Casner, "RTP Profile for Audio and Video Conferences with Minimal Control", STD 65, RFC 3551, DOI 10.17487/RFC3551, July 2003, <<http://www.rfc-editor.org/info/rfc3551>>.
- [RFC3738] Luby, M. and V. Goyal, "Wave and Equation Based Rate Control (WEBRC) Building Block", RFC 3738, DOI 10.17487/RFC3738, April 2004, <<http://www.rfc-editor.org/info/rfc3738>>.
- [RFC3758] Stewart, R., Ramalho, M., Xie, Q., Tuexen, M., and P. Conrad, "Stream Control Transmission Protocol (SCTP) Partial Reliability Extension", RFC 3758, DOI 10.17487/RFC3758, May 2004, <<http://www.rfc-editor.org/info/rfc3758>>.
- [RFC3819] Karn, P., Ed., Bormann, C., Fairhurst, G., Grossman, D., Ludwig, R., Mahdavi, J., Montenegro, G., Touch, J., and L. Wood, "Advice for Internet Subnetwork Designers", BCP 89, RFC 3819, DOI 10.17487/RFC3819, July 2004, <<http://www.rfc-editor.org/info/rfc3819>>.
- [RFC4301] Kent, S. and K. Seo, "Security Architecture for the Internet Protocol", RFC 4301, DOI 10.17487/RFC4301, December 2005, <<http://www.rfc-editor.org/info/rfc4301>>.
- [RFC4302] Kent, S., "IP Authentication Header", RFC 4302, DOI 10.17487/RFC4302, December 2005, <<http://www.rfc-editor.org/info/rfc4302>>.
- [RFC4303] Kent, S., "IP Encapsulating Security Payload (ESP)", RFC 4303, DOI 10.17487/RFC4303, December 2005, <<http://www.rfc-editor.org/info/rfc4303>>.
- [RFC4340] Kohler, E., Handley, M., and S. Floyd, "Datagram Congestion Control Protocol (DCCP)", RFC 4340, DOI 10.17487/RFC4340, March 2006, <<http://www.rfc-editor.org/info/rfc4340>>.
- [RFC4341] Floyd, S. and E. Kohler, "Profile for Datagram Congestion Control Protocol (DCCP) Congestion Control ID 2: TCP-like Congestion Control", RFC 4341, DOI 10.17487/RFC4341, March 2006, <<http://www.rfc-editor.org/info/rfc4341>>.

- [RFC4342] Floyd, S., Kohler, E., and J. Padhye, "Profile for Datagram Congestion Control Protocol (DCCP) Congestion Control ID 3: TCP-Friendly Rate Control (TFRC)", RFC 4342, DOI 10.17487/RFC4342, March 2006, <<http://www.rfc-editor.org/info/rfc4342>>.
- [RFC4380] Huitema, C., "Teredo: Tunneling IPv6 over UDP through Network Address Translations (NATs)", RFC 4380, DOI 10.17487/RFC4380, February 2006, <<http://www.rfc-editor.org/info/rfc4380>>.
- [RFC4607] Holbrook, H. and B. Cain, "Source-Specific Multicast for IP", RFC 4607, DOI 10.17487/RFC4607, August 2006, <<http://www.rfc-editor.org/info/rfc4607>>.
- [RFC4654] Widmer, J. and M. Handley, "TCP-Friendly Multicast Congestion Control (TFMCC): Protocol Specification", RFC 4654, DOI 10.17487/RFC4654, August 2006, <<http://www.rfc-editor.org/info/rfc4654>>.
- [RFC4880] Callas, J., Donnerhacke, L., Finney, H., Shaw, D., and R. Thayer, "OpenPGP Message Format", RFC 4880, DOI 10.17487/RFC4880, November 2007, <<http://www.rfc-editor.org/info/rfc4880>>.
- [RFC4890] Davies, E. and J. Mohacsi, "Recommendations for Filtering ICMPv6 Messages in Firewalls", RFC 4890, DOI 10.17487/RFC4890, May 2007, <<http://www.rfc-editor.org/info/rfc4890>>.
- [RFC4960] Stewart, R., Ed., "Stream Control Transmission Protocol", RFC 4960, DOI 10.17487/RFC4960, September 2007, <<http://www.rfc-editor.org/info/rfc4960>>.
- [RFC4963] Heffner, J., Mathis, M., and B. Chandler, "IPv4 Reassembly Errors at High Data Rates", RFC 4963, DOI 10.17487/RFC4963, July 2007, <<http://www.rfc-editor.org/info/rfc4963>>.
- [RFC4987] Eddy, W., "TCP SYN Flooding Attacks and Common Mitigations", RFC 4987, DOI 10.17487/RFC4987, August 2007, <<http://www.rfc-editor.org/info/rfc4987>>.
- [RFC5082] Gill, V., Heasley, J., Meyer, D., Savola, P., Ed., and C. Pignataro, "The Generalized TTL Security Mechanism (GTSM)", RFC 5082, DOI 10.17487/RFC5082, October 2007, <<http://www.rfc-editor.org/info/rfc5082>>.

- [RFC5245] Rosenberg, J., "Interactive Connectivity Establishment (ICE): A Protocol for Network Address Translator (NAT) Traversal for Offer/Answer Protocols", RFC 5245, DOI 10.17487/RFC5245, April 2010, <<http://www.rfc-editor.org/info/rfc5245>>.
- [RFC5622] Floyd, S. and E. Kohler, "Profile for Datagram Congestion Control Protocol (DCCP) Congestion ID 4: TCP-Friendly Rate Control for Small Packets (TFRC-SP)", RFC 5622, DOI 10.17487/RFC5622, August 2009, <<http://www.rfc-editor.org/info/rfc5622>>.
- [RFC5652] Housley, R., "Cryptographic Message Syntax (CMS)", STD 70, RFC 5652, DOI 10.17487/RFC5652, September 2009, <<http://www.rfc-editor.org/info/rfc5652>>.
- [RFC5681] Allman, M., Paxson, V., and E. Blanton, "TCP Congestion Control", RFC 5681, DOI 10.17487/RFC5681, September 2009, <<http://www.rfc-editor.org/info/rfc5681>>.
- [RFC5740] Adamson, B., Bormann, C., Handley, M., and J. Macker, "NACK-Oriented Reliable Multicast (NORM) Transport Protocol", RFC 5740, DOI 10.17487/RFC5740, November 2009, <<http://www.rfc-editor.org/info/rfc5740>>.
- [RFC5751] Ramsdell, B. and S. Turner, "Secure/Multipurpose Internet Mail Extensions (S/MIME) Version 3.2 Message Specification", RFC 5751, DOI 10.17487/RFC5751, January 2010, <<http://www.rfc-editor.org/info/rfc5751>>.
- [RFC5775] Luby, M., Watson, M., and L. Vicisano, "Asynchronous Layered Coding (ALC) Protocol Instantiation", RFC 5775, DOI 10.17487/RFC5775, April 2010, <<http://www.rfc-editor.org/info/rfc5775>>.
- [RFC5971] Schulzrinne, H. and R. Hancock, "GIST: General Internet Signalling Transport", RFC 5971, DOI 10.17487/RFC5971, October 2010, <<http://www.rfc-editor.org/info/rfc5971>>.
- [RFC5973] Stiernerling, M., Tschofenig, H., Aoun, C., and E. Davies, "NAT/Firewall NSIS Signaling Layer Protocol (NSLP)", RFC 5973, DOI 10.17487/RFC5973, October 2010, <<http://www.rfc-editor.org/info/rfc5973>>.
- [RFC6056] Larsen, M. and F. Gont, "Recommendations for Transport-Protocol Port Randomization", BCP 156, RFC 6056, DOI 10.17487/RFC6056, January 2011, <<http://www.rfc-editor.org/info/rfc6056>>.

- [RFC6092] Woodyatt, J., Ed., "Recommended Simple Security Capabilities in Customer Premises Equipment (CPE) for Providing Residential IPv6 Internet Service", RFC 6092, DOI 10.17487/RFC6092, January 2011, <<http://www.rfc-editor.org/info/rfc6092>>.
- [RFC6335] Cotton, M., Eggert, L., Touch, J., Westerlund, M., and S. Cheshire, "Internet Assigned Numbers Authority (IANA) Procedures for the Management of the Service Name and Transport Protocol Port Number Registry", BCP 165, RFC 6335, DOI 10.17487/RFC6335, August 2011, <<http://www.rfc-editor.org/info/rfc6335>>.
- [RFC6347] Rescorla, E. and N. Modadugu, "Datagram Transport Layer Security Version 1.2", RFC 6347, DOI 10.17487/RFC6347, January 2012, <<http://www.rfc-editor.org/info/rfc6347>>.
- [RFC6396] Blunk, L., Karir, M., and C. Labovitz, "Multi-Threaded Routing Toolkit (MRT) Routing Information Export Format", RFC 6396, DOI 10.17487/RFC6396, October 2011, <<http://www.rfc-editor.org/info/rfc6396>>.
- [RFC6437] Amante, S., Carpenter, B., Jiang, S., and J. Rajahalme, "IPv6 Flow Label Specification", RFC 6437, DOI 10.17487/RFC6437, November 2011, <<http://www.rfc-editor.org/info/rfc6437>>.
- [RFC6438] Carpenter, B. and S. Amante, "Using the IPv6 Flow Label for Equal Cost Multipath Routing and Link Aggregation in Tunnels", RFC 6438, DOI 10.17487/RFC6438, November 2011, <<http://www.rfc-editor.org/info/rfc6438>>.
- [RFC6513] Rosen, E., Ed. and R. Aggarwal, Ed., "Multicast in MPLS/BGP IP VPNs", RFC 6513, DOI 10.17487/RFC6513, February 2012, <<http://www.rfc-editor.org/info/rfc6513>>.
- [RFC6679] Westerlund, M., Johansson, I., Perkins, C., O'Hanlon, P., and K. Carlberg, "Explicit Congestion Notification (ECN) for RTP over UDP", RFC 6679, DOI 10.17487/RFC6679, August 2012, <<http://www.rfc-editor.org/info/rfc6679>>.
- [RFC6726] Paila, T., Walsh, R., Luby, M., Roca, V., and R. Lehtonen, "FLUTE - File Delivery over Unidirectional Transport", RFC 6726, DOI 10.17487/RFC6726, November 2012, <<http://www.rfc-editor.org/info/rfc6726>>.

- [RFC6773] Phelan, T., Fairhurst, G., and C. Perkins, "DCCP-UDP: A Datagram Congestion Control Protocol UDP Encapsulation for NAT Traversal", RFC 6773, DOI 10.17487/RFC6773, November 2012, <<http://www.rfc-editor.org/info/rfc6773>>.
- [RFC6807] Farinacci, D., Shepherd, G., Venaas, S., and Y. Cai, "Population Count Extensions to Protocol Independent Multicast (PIM)", RFC 6807, DOI 10.17487/RFC6807, December 2012, <<http://www.rfc-editor.org/info/rfc6807>>.
- [RFC6887] Wing, D., Ed., Cheshire, S., Boucadair, M., Penno, R., and P. Selkirk, "Port Control Protocol (PCP)", RFC 6887, DOI 10.17487/RFC6887, April 2013, <<http://www.rfc-editor.org/info/rfc6887>>.
- [RFC6935] Eubanks, M., Chimento, P., and M. Westerlund, "IPv6 and UDP Checksums for Tunneled Packets", RFC 6935, DOI 10.17487/RFC6935, April 2013, <<http://www.rfc-editor.org/info/rfc6935>>.
- [RFC6936] Fairhurst, G. and M. Westerlund, "Applicability Statement for the Use of IPv6 UDP Datagrams with Zero Checksums", RFC 6936, DOI 10.17487/RFC6936, April 2013, <<http://www.rfc-editor.org/info/rfc6936>>.
- [RFC6951] Tuexen, M. and R. Stewart, "UDP Encapsulation of Stream Control Transmission Protocol (SCTP) Packets for End-Host to End-Host Communication", RFC 6951, DOI 10.17487/RFC6951, May 2013, <<http://www.rfc-editor.org/info/rfc6951>>.
- [RFC7143] Chadalapaka, M., Satran, J., Meth, K., and D. Black, "Internet Small Computer System Interface (iSCSI) Protocol (Consolidated)", RFC 7143, DOI 10.17487/RFC7143, April 2014, <<http://www.rfc-editor.org/info/rfc7143>>.
- [RFC7201] Westerlund, M. and C. Perkins, "Options for Securing RTP Sessions", RFC 7201, DOI 10.17487/RFC7201, April 2014, <<http://www.rfc-editor.org/info/rfc7201>>.
- [RFC7296] Kaufman, C., Hoffman, P., Nir, Y., Eronen, P., and T. Kivinen, "Internet Key Exchange Protocol Version 2 (IKEv2)", STD 79, RFC 7296, DOI 10.17487/RFC7296, October 2014, <<http://www.rfc-editor.org/info/rfc7296>>.
- [RFC7450] Bumgardner, G., "Automatic Multicast Tunneling", RFC 7450, DOI 10.17487/RFC7450, February 2015, <<http://www.rfc-editor.org/info/rfc7450>>.

- [RFC7510] Xu, X., Sheth, N., Yong, L., Callon, R., and D. Black, "Encapsulating MPLS in UDP", RFC 7510, DOI 10.17487/RFC7510, April 2015, <<http://www.rfc-editor.org/info/rfc7510>>.
- [RFC7525] Sheffer, Y., Holz, R., and P. Saint-Andre, "Recommendations for Secure Use of Transport Layer Security (TLS) and Datagram Transport Layer Security (DTLS)", BCP 195, RFC 7525, DOI 10.17487/RFC7525, May 2015, <<http://www.rfc-editor.org/info/rfc7525>>.
- [RFC7560] Kuehlewind, M., Ed., Scheffenegger, R., and B. Briscoe, "Problem Statement and Requirements for Increased Accuracy in Explicit Congestion Notification (ECN) Feedback", RFC 7560, DOI 10.17487/RFC7560, August 2015, <<http://www.rfc-editor.org/info/rfc7560>>.
- [RFC7567] Baker, F., Ed. and G. Fairhurst, Ed., "IETF Recommendations Regarding Active Queue Management", BCP 197, RFC 7567, DOI 10.17487/RFC7567, July 2015, <<http://www.rfc-editor.org/info/rfc7567>>.
- [RFC7605] Touch, J., "Recommendations on Using Assigned Transport Port Numbers", BCP 165, RFC 7605, DOI 10.17487/RFC7605, August 2015, <<http://www.rfc-editor.org/info/rfc7605>>.
- [RFC7657] Black, D., Ed. and P. Jones, "Differentiated Services (Diffserv) and Real-Time Communication", RFC 7657, DOI 10.17487/RFC7657, November 2015, <<http://www.rfc-editor.org/info/rfc7657>>.
- [RFC7675] Perumal, M., Wing, D., Ravindranath, R., Reddy, T., and M. Thomson, "Session Traversal Utilities for NAT (STUN) Usage for Consent Freshness", RFC 7675, DOI 10.17487/RFC7675, October 2015, <<http://www.rfc-editor.org/info/rfc7675>>.
- [RFC8083] Perkins, C. and V. Singh, "Multimedia Congestion Control: Circuit Breakers for Unicast RTP Sessions", RFC 8083, DOI 10.17487/RFC8083, March 2017, <<http://www.rfc-editor.org/info/rfc8083>>.
- [RFC8086] Yong, L., Ed., Crabbe, E., Xu, X., and T. Herbert, "GRE-in-UDP Encapsulation", RFC 8086, DOI 10.17487/RFC8086, March 2017, <<http://www.rfc-editor.org/info/rfc8086>>.

- [RFC8087] Fairhurst, G. and M. Welzl, "The Benefits of Using Explicit Congestion Notification (ECN)", RFC 8087, DOI 10.17487/RFC8087, March 2017, <<http://www.rfc-editor.org/info/rfc8087>>.
- [STEVENS] Stevens, W., Fenner, B., and A. Rudoff, "UNIX Network Programming, The sockets Networking API", Addison-Wesley, 2004.
- [UPnP] UPnP Forum, , "Internet Gateway Device (IGD) Standardized Device Control Protocol V 1.0", November 2001.

## Appendix A. Case Study of the Use of IPv6 UDP Zero-Checksum Mode

This appendix provides a brief review of MPLS-in-UDP as an example of a UDP Tunnel Encapsulation that defines a UDP encapsulation. The purpose of the appendix is to provide a concrete example of which mechanisms were required in order to safely use UDP zero-checksum mode for MPLS-in-UDP tunnels over IPv6. By default, UDP requires a checksum for use with IPv6. An option has been specified that permits a zero IPv6 UDP checksum when used in specific environments, specified in [RFC7510], and defines a set of operational constraints for use of this mode. These are summarized below:

A UDP tunnel or encapsulation using a zero-checksum mode with IPv6 must only be deployed within a single network (with a single network operator) or networks of an adjacent set of cooperating network operators where traffic is managed to avoid congestion, rather than over the Internet where congestion control is required. MPLS-in-UDP has been specified for networks under single administrative control (such as within a single operator's network) where it is known (perhaps through knowledge of equipment types and lower-layer checks) that packet corruption is exceptionally unlikely and where the operator is willing to take the risk of undetected packet corruption.

The tunnel encapsulator SHOULD use different IPv6 addresses for each UDP tunnel that uses the UDP zero-checksum mode, regardless of the decapsulator, to strengthen the decapsulator's check of the IPv6 source address (i.e., the same IPv6 source address SHOULD NOT be used with more than one IPv6 destination address, independent of whether that destination address is a unicast or multicast address). Use of MPLS-in-UDP may be extended to networks within a set of closely cooperating network administrations (such as network operators who have agreed to work together to jointly provide specific services) [RFC7510].

The requirement for MPLS-in-UDP endpoints to check the source IPv6 address in addition to the destination IPv6 address, plus the strong recommendation against reuse of source IPv6 addresses among MPLS-in-UDP tunnels collectively provide some mitigation for the absence of UDP checksum coverage of the IPv6 header. In addition, the MPLS data plane only forwards packets with valid labels (i.e., labels that have been distributed by the tunnel egress Label Switched Router, LSR), providing some additional opportunity to detect MPLS-in-UDP packet misdelivery when the misdelivered packet contains a label that is not valid for forwarding at the receiving LSR. The expected result for IPv6 UDP zero-checksum mode for MPLS-in-UDP is that corruption of the destination IPv6 address will usually cause packet discard, as offsetting corruptions to the source IPv6 and/or MPLS top label are unlikely.

Additional assurance is provided by the restrictions in the above exceptions that limit usage of IPv6 UDP zero-checksum mode to well-managed networks for which MPLS packet corruption has not been a problem in practice. Hence, MPLS-in-UDP is suitable for transmission over lower layers in well-managed networks that are allowed by the exceptions stated above and the rate of corruption of the inner IP packet on such networks is not expected to increase by comparison to MPLS traffic that is not encapsulated in UDP. For these reasons, MPLS-in-UDP does not provide an additional integrity check when UDP zero-checksum mode is used with IPv6, and this design is in accordance with requirements 2, 3, and 5 specified in Section 5 of [RFC6936].

The MPLS-in-UDP encapsulation does not provide a mechanism to safely fall back to using a checksum when a path change occurs that redirects a tunnel over a path that includes a middlebox that discards IPv6 datagrams with a zero UDP checksum. In this case, the MPLS-in-UDP tunnel will be black-holed by that middlebox. Recommended changes to allow firewalls, NATs and other middleboxes to support use of an IPv6 zero UDP checksum are described in Section 5 of [RFC6936]. MPLS does not accumulate incorrect state as a consequence of label-stack corruption. A corrupt MPLS label results in either packet discard or forwarding (and forgetting) of the packet without accumulation of MPLS protocol state. Active monitoring of MPLS-in-UDP traffic for errors is REQUIRED because the occurrence of errors will result in some accumulation of error information outside the MPLS protocol for operational and management purposes. This design is in accordance with requirement 4 specified in Section 5 of [RFC6936]. In addition, IPv6 traffic with a zero UDP checksum MUST be actively monitored for errors by the network operator.

Operators SHOULD also deploy packet filters to prevent IPv6 packets with a zero UDP checksum from escaping from the network due to misconfiguration or packet errors. In addition, IPv6 traffic with a zero UDP checksum MUST be actively monitored for errors by the network operator.

## Acknowledgments

The middlebox traversal guidelines in Section 3.5 incorporate ideas from Section 5 of [BEHAVE-APP] by Bryan Ford, Pyda Srisuresh, and Dan Kegel. The protocol timer guidelines in Section 3.1.1 were largely contributed by Mark Allman.

G. Fairhurst received funding from the European Union's Horizon 2020 research and innovation program 2014-2018 under grant agreement No. 644334 (NEAT). Lars Eggert has received funding from the European Union's Horizon 2020 research and innovation program 2014-2018 under grant agreement No. 644866 (SSICLOPS). This document reflects only the authors' views and the European Commission is not responsible for any use that may be made of the information it contains.

## Authors' Addresses

Lars Eggert  
NetApp  
Sonnenallee 1  
Kirchheim 85551  
Germany

Phone: +49 151 120 55791  
Email: [lars@netapp.com](mailto:lars@netapp.com)  
URI: <https://eggert.org/>

Godred Fairhurst  
University of Aberdeen  
Department of Engineering  
Fraser Noble Building  
Aberdeen AB24 3UE  
Scotland

Email: [gorry@erg.abdn.ac.uk](mailto:gorry@erg.abdn.ac.uk)  
URI: <http://www.erg.abdn.ac.uk/>

Greg Shepherd  
Cisco Systems  
Tasman Drive  
San Jose  
United States of America

Email: [gjshep@gmail.com](mailto:gjshep@gmail.com)

