

Network Working Group
Internet-Draft
Intended status: Standards Track
Expires: January 07, 2016

S. Dhesikan
C. Jennings
Cisco Systems
D. Druta, Ed.
AT&T
P. Jones
J. Polk
Cisco Systems
July 06, 2015

DSCP and other packet markings for RTCWeb QoS
draft-ietf-tsvwg-rtcweb-qos-04

Abstract

Many networks, such as service provider and enterprise networks, can provide treatment for individual packets based on Differentiated Services Code Point (DSCP) values on a per-hop basis. This document provides the recommended DSCP values for browsers to use for various classes of traffic.

Status of This Memo

This Internet-Draft is submitted in full conformance with the provisions of [BCP 78](#) and [BCP 79](#).

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

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

This Internet-Draft will expire on January 07, 2016.

Copyright Notice

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

This document is subject to [BCP 78](#) and the IETF Trust's Legal Provisions Relating to IETF Documents (<http://trustee.ietf.org/license-info>) in effect on the date of publication of this document. Please review these documents

Internet-Draft

RTCWeb QoS

July 2015

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	2
2.	Relation to Other Standards	3
3.	Terminology	4
4.	Inputs	4
5.	DSCP Mappings	5
6.	Security Considerations	6
7.	IANA Considerations	7
8.	Downward References	7
9.	Acknowledgements	7
10.	Document History	7
11.	References	7
11.1.	Normative References	7
11.2.	Informative References	8
	Authors' Addresses	8

[1.](#) Introduction

Differentiated Services Code Points (DSCP) [[RFC2474](#)] style packet marking can help provide QoS in some environments. There are many use cases where such marking does not help, but it seldom makes things worse if packets are marked appropriately. In other words, if too many packets, say all audio or all audio and video, are marked for a given network condition then it can prevent desirable results. Either too much other traffic will be starved, or there is not enough capacity for the preferentially marked packets (i.e., audio and/or video).

This specification proposes how WebRTC applications can mark packets. This specification does not contradict or redefine any advice from previous IETF RFCs, but merely provides a simple set of recommendations for implementers based on the previous RFCs

There are some environments where DSCP markings frequently help. These include:

1. Private, wide-area networks.

2. Residential Networks. If the congested link is the broadband uplink in a cable or DSL scenario, often residential routers/NAT support preferential treatment based on DSCP.

3. Wireless Networks. If the congested link is a local wireless network, marking may help.

Traditionally DSCP values have been thought of as being site specific, with each site selecting its own code points for controlling per-hop-behavior to influence the QoS for transport flows. However in the RTCWeb use cases, the browsers need to set them to something when there is no site specific information. In this document, "browsers" is used synonymously with "Interactive User Agent" as defined in the HTML specification, [[W3C.REC-html5-20141028](#)]. This document describes a subset of DSCP code point values drawn from existing RFCs and common usage for use with WebRTC applications. These code points are solely defaults.

This specification defines some inputs that the browser in a WebRTC application can consider to aid in determining how to set the various packet markings and defines the mapping from abstract QoS policies (data type, priority level) to those packet markings.

2. Relation to Other Standards

This document exists as a complement to [[I-D.ietf-dart-dscp-rtp](#)], which describes the interaction between DSCP and real-time communications. It covers the implications of using various DSCP values, particularly focusing on Real-time Transport Protocol (RTP) [[RFC3550](#)] streams that are multiplexed onto a single transport-layer flow.

There are a number of guidelines specified in [[I-D.ietf-dart-dscp-rtp](#)] that should be followed when marking traffic sent by WebRTC applications, as it is common for multiple RTP streams to be multiplexed on the same transport flow. Generally, the RTP streams would be marked with a value as appropriate from Table 1. A WebRTC application might multiplex data channel traffic over the same 5-tuple as RTP streams, which would also be marked as per that table. The guidance in [[I-D.ietf-dart-dscp-rtp](#)] indicates that all data

channel traffic would be marked with a single value that is different than the value(s) used for RTP streams multiplexed with the data channel traffic over the same 5-tuple, assuming RTP streams are marked with a value other than default forwarding (DF).

In the event that RTP streams are carried over a reliable transport (e.g., TCP), all such packets, even there are multiple RTP streams that would be marked differently if sent via an unreliable transport, must be marked with a single DSCP value.

This specification does not change or override the advice in any other standards about setting packet markings. It simply selects a

subset of DSCP values that is relevant in the RTCWeb context. In some cases, such as DSCP where the normative RFC leaves open multiple options from which to choose, this clarifies which values should be used in the RTCWeb context. This document also specifies the inputs that are needed by the browser to provide to the media engine.

The DSCP value set by the endpoint is not always trusted by the network. Therefore, the DSCP value may be remarked at any place in the network for a variety of reasons to any other DSCP value, including default forwarding (DF) value to provide basic best effort service. The mitigation for such action is through an authorization mechanism. Such authorization mechanism is outside the scope of this document. There is benefit in marking traffic even if it only benefits the first few hops.

[3.](#) Terminology

The key words "MUST", "MUST NOT", "SHOULD", "SHOULD NOT", and "MAY" in this document are to be interpreted as described in [[RFC2119](#)].

[4.](#) Inputs

The below uses the concept of a media flow, however this is usually not equivalent to a transport flow defined by a 5-tuple (source address, destination address, source port, destination port, and protocol). Instead each media flow contains all the packets associated with an independent media entity within one 5-tuple. There may be multiple media flows within the same 5-tuple. These media flows might consist of different media types and have different

levels of importance to the application and, therefore, each potentially marked using different DSCP values than for another media flow multiplexed over the same transport flow. The following are the inputs that the browser provides to the media engine:

- o Data Type: The browser provides this input as it knows if the flow is audio, interactive video with or without audio, non-interactive video with or without audio, or data.
- o Application Priority: Another input is the relative importance of the flow within that data type. Many applications have multiple media flows of the same data type and often some flows are more important than others. For example, in a video conference where there are usually audio and video flows, the audio flow may be more important than the video flow. JavaScript applications can tell the browser whether a particular media flow is high, medium, low or very low importance to the application.

[I-D.ietf-rtcweb-transports] defines in more detail what an individual media flow is within the WebRTC context.

As an example of different media flows that might be multiplexed over the same transport flow, packets related to one RTP stream (e.g., an audio flow) carried over UDP might be one media flow, packets related to a second RTP stream (e.g., presentation video) carried over UDP might be a second media flow, and finally data channel packets carried via SCTP over DTLS might be third media flow.

5. DSCP Mappings

Below is a table of DSCP markings for each data type of interest to RTCWeb. These DSCP values for each data type listed are a reasonable subset of code point values taken from [RFC4594]. A web browser SHOULD use these values to mark the appropriate media packets. More information on EF can be found in [RFC3246]. More information on AF can be found in [RFC2597]. DF is default forwarding which provides the basic best effort service.

Data Type	Very Low	Low	Medium	High
Audio	CS1	DF	EF (46)	EF (46)

	(8)	(0)		
Interactive Video with or without audio	CS1 (8)	DF (0)	AF42, AF43 (36, 38)	AF41, AF42 (34, 36)
Non-Interactive Video with or without audio	CS1 (8)	DF (0)	AF32, AF33 (28, 30)	AF31, AF32 (26, 28)
Data	CS1 (8)	DF (0)	AF1x (10, 12, 14)	AF2x (18, 20, 22)

Table 1

The columns "very low", "low", "Medium" and "high" signify the relative importance of the media flow within the application and is an input that the browser receives to assist it in selecting the DSCP value. These are referred to as application priority in this document. Application priority does not refer to priority in the network transport.

The browser SHOULD first select the data type of the media flow. Within the data type, the relative importance of the media flow SHOULD be used to select the appropriate DSCP value. All packets

within a media flow SHOULD have the same application priority. In some cases, the selected cell may have multiple DSCP values, such as AF41 and AF42. These offer different drop precedences. One may select difference drop precedences for the different packets in the same media flow. Therefore, all packets in the media flow SHOULD be marked with the same application priority, but can have difference drop precedences.

The combination of data type and application priority provides specificity and helps in selecting the right DSCP value for the media flow. In some cases, the different drop precedence values provides additional granularity in classifying packets within a media flow. For example, in a video conference, the video media flow may have medium application priority. If so, either AF42 or AF43 may be selected. If the I-frames in the stream are more important than the

P-frames, then the I-frames can be marked with AF42 and the P-frames marked with AF43.

If a packet enters a QoS domain that has no support for the above defined data types/application priority (service class), then the network node at the edge will remark the DSCP value based on policies. This could result in the media flow not getting the network treatment it expects based on the original DSCP value in the packet. Subsequently, if the packet enters a QoS domain that supports a larger number of service classes, there may not be sufficient information in the packet to restore the original markings. Mechanisms for restoring such original DSCP is outside the scope of this document.

The above table assumes that packets marked with CS1 are treated as "less than best effort". However, the treatment of CS1 is implementation dependent. If an implementation treats CS1 as other than "less than best effort", then the actual priority (or, more precisely, the per-hop-behavior) of the packets may be changed from what is intended. Implementers should also note that the excess EF traffic is dropped. This could mean that a packet marked as EF may not get through as opposed to a packet marked with a different DSCP value.

In summary, there are no guarantees or promised level of service with the use of DSCP. The service provided to a packet is dependent upon the network design along the path, as well as the congestion levels at every hop.

6. Security Considerations

This specification does not add any additional security implication other than the normal application use of DSCP. For security

implications on use of DSCP, please refer to [Section 6 of RFC 4594](#). Please also see [[I-D.ietf-rtcweb-security](#)] as an additional reference.

7. IANA Considerations

This specification does not require any actions from IANA.

8. Downward References

This specification contains a downwards reference to [[RFC4594](#)]. However, the parts of that RFC used by this specification are sufficiently stable for this downward reference.

9. Acknowledgements

Thanks To David Black, Magnus Westerland, Paolo Severini, Jim Hasselbrook, Joe Marcus, and Erik Nordmark for their help.

10. Document History

Note to RFC Editor: Please remove this section.

This document was originally an individual submission in RTCWeb WG. The RTCWeb working group selected it to be become a WG document. Later the transport ADs requested that this be moved to the TSVWG WG as that seemed to be a better match. This document is now being submitted as individual submission to the TSVWG with the hope that WG will select it as a WG draft and move it forward to an RFC.

11. References

11.1. Normative References

[I-D.ietf-dart-dscp-rtp]

Black, D. and P. Jones, "Differentiated Services (DiffServ) and Real-time Communication", [draft-ietf-dart-dscp-rtp-10](#) (work in progress), November 2014.

[I-D.ietf-rtcweb-security]

Rescorla, E., "Security Considerations for WebRTC", [draft-ietf-rtcweb-security-07](#) (work in progress), July 2014.

[I-D.ietf-rtcweb-transports]

Alvestrand, H., "Transports for WebRTC", [draft-ietf-rtcweb-transports-08](#) (work in progress), February 2015.

Requirement Levels", [BCP 14](#), [RFC 2119](#), March 1997.

[RFC4594] Babiarz, J., Chan, K., and F. Baker, "Configuration Guidelines for DiffServ Service Classes", [RFC 4594](#), August 2006.

11.2. Informative References

[RFC2474] Nichols, K., Blake, S., Baker, F., and D. Black, "Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers", [RFC 2474](#), December 1998.

[RFC2597] Heinanen, J., Baker, F., Weiss, W., and J. Wroclawski, "Assured Forwarding PHB Group", [RFC 2597](#), June 1999.

[RFC3246] Davie, B., Charny, A., Bennet, J., Benson, K., Le Boudec, J., Courtney, W., Davari, S., Firoiu, V., and D. Stiliadis, "An Expedited Forwarding PHB (Per-Hop Behavior)", [RFC 3246](#), March 2002.

[RFC3550] Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications", STD 64, [RFC 3550](#), July 2003.

[W3C.REC-html5-20141028]
Hickson, I., Berjon, R., Faulkner, S., Leithead, T., Navara, E., O'Connor, E., and S. Pfeiffer, "HTML5", World Wide Web Consortium Recommendation REC-html5-20141028, October 2014,
<<http://www.w3.org/TR/2014/REC-html5-20141028>>.

Authors' Addresses

Subha Dhesikan
Cisco Systems

Email: sdhesika@cisco.com

Cullen Jennings
Cisco Systems

Email: fluffy@cisco.com

Dan Druta (editor)
AT&T

Email: dd5826@att.com

Paul E. Jones
Cisco Systems

Email: paulej@packetizer.com

James Polk
Cisco Systems

Email: jmpolk@cisco.com

