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# DSCP and other packet markings for WebRTC QoS
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## 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

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## 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 WebRTC 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 also multiplex data channel [I-D.ietf-rtcweb-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] says that all data channel traffic would be marked with a single value that is typically 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). This is expanded upon further in the next section.

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 WebRTC 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", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" 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:

- 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.
- 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 WebRTC. 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 (8) | DF (0) | EF (46) | EF (46) |
| 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) | AF11 | AF21 |

Table 1: Recommended DSCP Values for WebRTC Applications

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 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. It is common for CS1 to be treated the same as DF so anyone using CS1 cannot assume that CS1 will be treated differently than DF. 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.

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.

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.

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. With the exception of data channel traffic, one may select different 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 different drop precedences.

For reasons discussed in Section 6 of [I-D.ietf-dart-dscp-rtp], if multiple media flows are multiplexed using a reliable transport (e.g., TCP) then all of the packets for all media flows multiplexed over that transport flow MUST be marked using the same DSCP value. Likewise, all WebRTC data channel packets transmitted over an SCTP association MUST be marked using the same DSCP value, regardless of how many data channels (streams) exist or what kind of traffic is carried over the various SCTP streams. In the event that the browser wishes to change the DSCP value in use for an SCTP association, it MUST reset the SCTP congestion controller after changing values. Frequent changes in the DSCP value used for an SCTP association are discouraged, though, as this would defeat any attempts at effectively managing congestion. It should also be noted that any change in DSCP value that results in a reset of the congestion controller puts the SCTP association back into slow start, which may have undesirable effects on application performance.

For the data channel traffic multiplexed over an SCTP association, it is RECOMMENDED that the DSCP value selected be the one associated with the highest priority requested for all data channels multiplexed over the SCTP association. Likewise, when multiplexing multiple media flows over a TCP connection, the DCSP value selected should be

the one associated with the highest priority requested for all multiplexed flows.

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.

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, Erik Nordmark, and Michael Tuexen for their help.

## 10. Dedication

This document is dedicated to the memory of James Polk, a long-time friend and colleague. James made important contributions to this

specification, including being one of its primary authors. The IETF global community mourns his loss and he will be missed dearly.

## 11. 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.

## 12. References

### 12.1. Normative References

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### 12.2. Informative References

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