Transport Working Group
Internet-Draft
Updates: 4542,4594
(if approved)
Intended status: Standards Track
Expires: May 5, 2009

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November 1, 2008

DSCP for Capacity-Admitted Traffic
draft-ietf-tsvwg-admitted-realtime-dscp-05

Status of this Memo

Abstract

This document requests one Differentiated Services Code Point (DSCP) from the Internet Assigned Numbers Authority (IANA) for real-time traffic classes similar to voice conforming to the Expedited Forwarding Per Hop Behavior, and admitted using a call admission procedure involving authentication, authorization, and capacity admission.

This document also recommends that certain classes of video traffic described in RFC 4594 and which have similar requirements be changed to require admission using a Call Admission Control (CAC) procedure

involving authentication, authorization, and capacity admission.

Requirements Language

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

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1. Introduction

This document requests one Differentiated Services Code Point (DSCP) from the Internet Assigned Numbers Authority (IANA) for a class of real-time traffic. This class conforms to the Expedited Forwarding [RFC3246] [RFC3247] Per Hop Behavior. It is also admitted using a CAC procedure involving authentication, authorization, and capacity admission. This differs from a real-time traffic class conforming to the Expedited Forwarding Per Hop Behavior but not subject to capacity admission or subject to very coarse capacity admission.

It also recommends that certain classes of video described in [RFC4594] be treated as requiring capacity admission as well.

These applications have one or more potential congestion points between the endpoints. Reserving capacity for them is important to application performance. All of these applications have low tolerance to jitter (aka delay variation) and loss, as summarized in Section 2, and most (except for multimedia conferencing) have inelastic flow behavior from Figure 1 of [RFC4594]. Inelastic flow behavior and low jitter/loss tolerance are the service characteristics that define the need for admission control behavior.

One of the reasons behind this is the need for classes of traffic that are handled under special policies, such as the non-preemptive Emergency Telecommunication Service, the US Department of Defense's Assured Service (which is similar to Multi-Level Precedence and Preemption [ITU.MLPP.1990] procedure), or e-911, in addition to normal routine calls that use call admission. It is possible to use control plane protocols to generally restrict session admission such that admitted traffic should receive the desired service, and the policy (e.g. Routine, National Security or Emergency Preparedness [NS/EP] communications, e-911, etc) need not be signaled in a DSCP. However, service providers need to distinguish between special-policy traffic and other classes, particularly the existing VoIP services that perform no capacity admission or only very coarse capacity

admission and can exceed their allocated resources.

The requested DSCP applies to the Telephony Service Class described in [RFC4594].

Since video classes have not had the history of mixing admitted and non-admitted traffic in the same Per-Hop Behavior (PHB) as has occurred for EF, an additional DSCP code point is not recommended. Instead, the recommended "best practice" is to perform admission control for all traffic in three of [RFC4594]'s video classes: the

- Interactive Real-Time Traffic (CS4, used for Video conferencing and Interactive gaming),
- Broadcast TV (CS3) for use in a video on demand context, and
- AF4 Multimedia Conferencing (video conferencing).

Other video classes are believed to not have the current problem of confusion with unadmitted traffic and therefore would not benefit from the notion of a separate DSCP for admitted traffic. Within an ISP and on inter-ISP links (i.e. within networks whose internal paths are uniform at hundreds of megabits or faster), one would expect all of this traffic to be carried in the Real Time Traffic Class described in [RFC5127].

1.1. Definitions

The following terms and acronyms are used in this document.

PHB: A Per-Hop-Behavior (PHB) is the externally observable forwarding behavior applied at a Differentiated Services compliant node to a DS behavior aggregate [RFC2475]. It may be thought of as a program configured on the interface of an Internet host or router, specified drop probabilities, queuing priorities or rates, and other handling characteristics for the traffic class.

DSCP: The Differentiated Services Code Point (DSCP), as defined in [RFC2474], is a value which is encoded in the DS field, and which each DS Node MUST use to select the PHB which is to be experienced

by each packet it forwards [RFC3260]. It is a 6-bit number embedded into the 8-bit TOS field of an IPv4 datagram or the Traffic Class field of an IPv6 datagram.

CAC: Call Admission Control includes concepts of authorization and capacity admission. "Authorization" refers to any procedure that identifies a user, verifies the authenticity of the identification, and determines whether the user is authorized to use the service under the relevant policy. "Capacity Admission" refers to any procedure that determines whether capacity exists supporting a session's requirements under some policy.

In the Internet, these are separate functions, while in the PSTN they and call routing are carried out together.

UNI: A User/Network Interface (UNI) is the interface (often a physical link or its virtual equivalent) that connects two entities that do not trust each other, and in which one (the user) purchases connectivity services from the other (the network).

Figure 1 shows two user networks connected by what appears to each of them to be a single network ("The Internet", access to which is provided by their service provider) that provides connectivity services to other users.

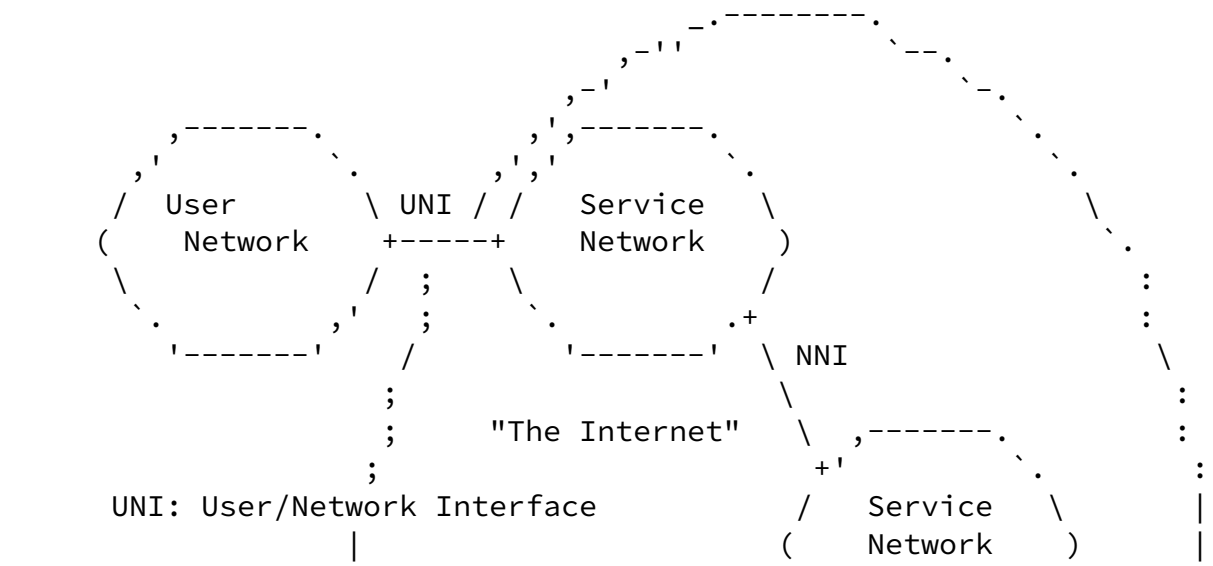
UNIs tend to be the bottlenecks in the Internet, where users purchase relatively low amounts of bandwidth for cost or service reasons, and as a result are most subject to congestion issues and therefore issues requiring traffic conditioning and service prioritization.

NNI: A Network/Network Interface (NNI) is the interface (often a physical link or its virtual equivalent) that connects two entities that trust each other within limits, and in which the two are seen as trading services for value. Figure 1 shows three service networks that together provide the connectivity services that we call "the Internet". They are different administrations and are very probably in competition, but exchange contracts for connectivity and capacity that enable them to offer specific services to their customers.

NNIs may not be bottlenecks in the Internet if service providers

contractually agree to provision excess capacity at them, as they commonly do. However, NNI performance may differ by ISP, and the performance guarantee interval may range from a month to a much shorter period. Furthermore, a peering point NNI may not have contractual performance guarantees or may become overloaded under certain conditions. They are also policy-controlled interfaces, especially in BGP. As a result, they may require traffic prioritization policy.

Queue: There are multiple ways to build a multi-queue scheduler. Weighted Round Robin (WRR) literally builds multiple lists and visits them in a specified order, while a calendar queue (often used to implement Weighted Fair Queuing, or WFQ) builds a list for each time interval and enqueues at most a stated amount of data in each such list for transmission during that time interval. While these differ dramatically in implementation, the external difference in behavior is generally negligible when they are properly configured. Consistent with the definitions used in the Differentiated Services Architecture [RFC2475], these are treated as equivalent in this document, and the lists of WRR and the classes of a calendar queue will be referred to uniformly as "queues".



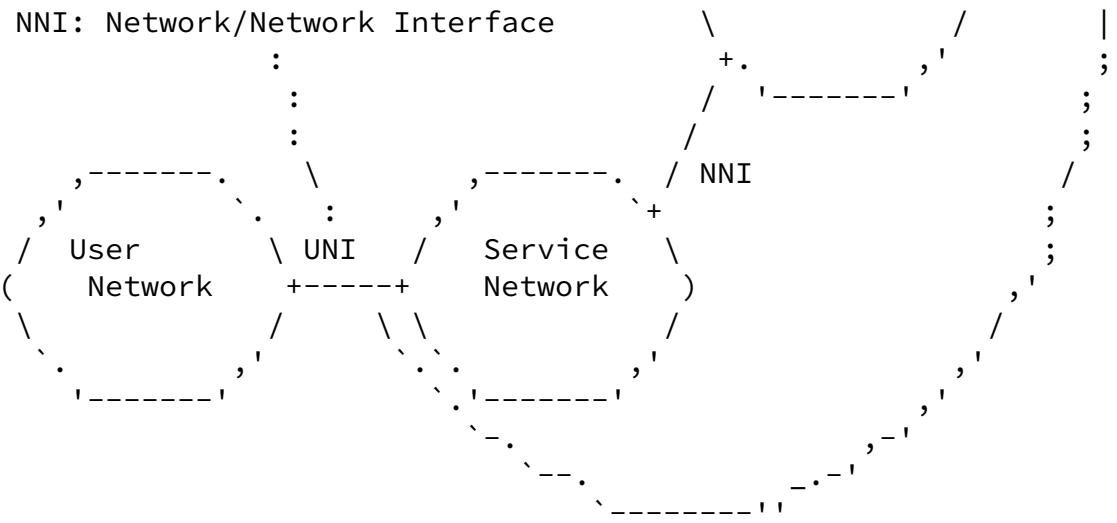


Figure 1: UNI and NNI interfaces

1.2. Problem

In short, the Telephony Service Class described in [RFC4594] permits the use of capacity admission in implementing the service, but present implementations either provide no capacity admission services or do so in a manner that depends on specific traffic engineering. In the context of the Internet backbone, the two are essentially equivalent; the edge network depends on specific engineering by the service provider that may not be present, especially in a mobile environment.

However, services are being requested of the network that would specifically make use of capacity admission, and would distinguish among users or the uses of available Voice-over-IP or Video-over-IP capacity in various ways. Various agencies would like to provide services as described in section 2.6 of [RFC4504] or in [RFC4190].

This requires the use of capacity admission to differentiate among users (which might be 911 call centers, other offices with preferential service contracts, or individual users gaining access with special credentials) to provide services to them that are not afforded to non-capacity admitted customer-to-customer IP telephony sessions.

2. Candidate Implementations of the Admitted Telephony Service Class

2.1. Potential implementations of EF in this model

There are at least two possible ways to implement isolation between the Capacity Admitted PHB and the Expedited Forwarding PHB in this model. They are to implement separate classes as a set of

- Multiple data plane traffic classes, each consisting of a policer and a queue, and the queues enjoying different priorities, or

- Multiple data plane traffic classes, each consisting of a policer but feeding into a common queue or multiple queues at the same priority.

We will explain the difference, and describe in what way they differ in operation. The reason this is necessary is that there is current confusion in the industry.

The multi-priority model is shown in Figure 2. In this model, traffic from each service class is placed into a separate priority queue. If data is present in both queues, traffic from one of them will always be selected for transmission. This has the effect of transferring jitter from the higher priority queue to the lower priority queue, and reordering traffic in a way that gives the higher priority traffic a smaller average queuing delay. Each queue must have its own policer, however, to protect the network from errors and attacks; if a traffic class thinks it is carrying a certain data rate but an abuse sends significantly more, the effect of simple prioritization would not preserve the lower priorities of traffic, which could cause routing to fail or otherwise impact an SLA.

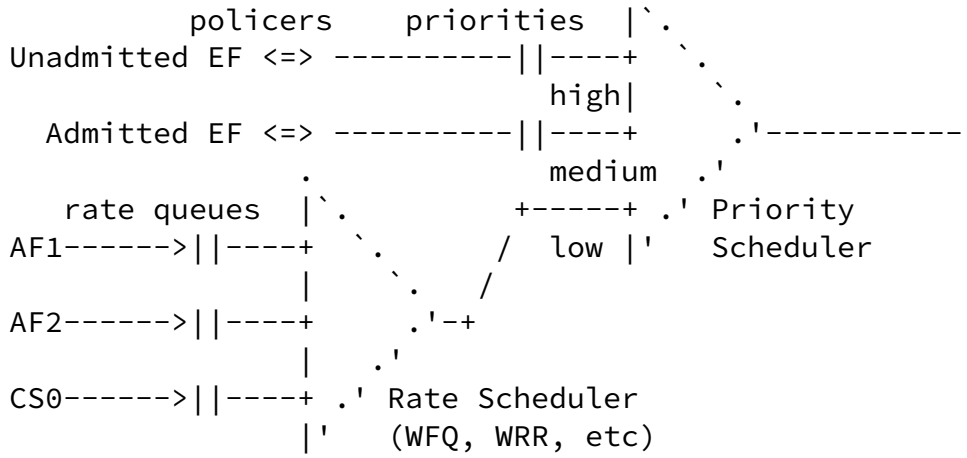


Figure 2: Implementation as a data plane priority

The multi-policer model is shown in Figure 3. In this model, traffic from each service class is policed according to its SLA requirements, and then placed into a common priority queue. Unlike the multi-priority model, the jitter experienced by the traffic classes in this case is the same, as there is only one queue, but the sum of the traffic in this higher priority queue experiences less average jitter than the elastic traffic in the lower priority.

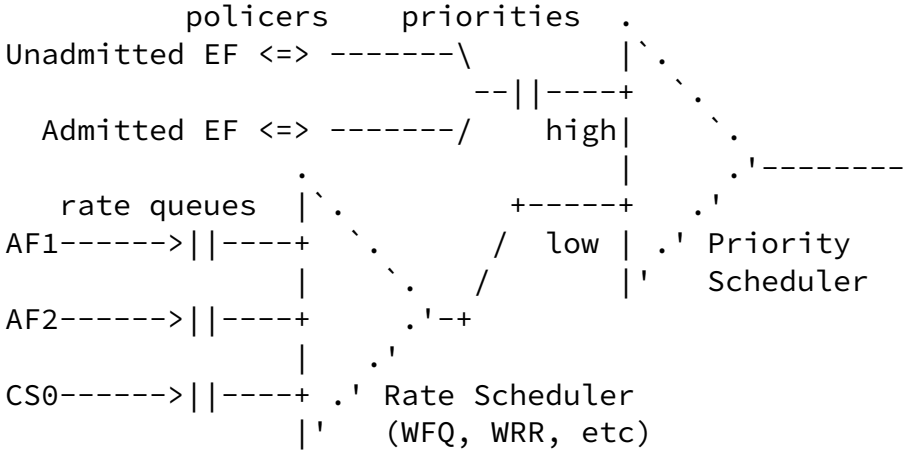


Figure 3: Implementation as a data plane policer

The difference between the two operationally is, as stated, the issues of loss due to policing and distribution of jitter.

If the two traffic classes are, for example, voice and video, datagrams conaining video data can be relatively large (often of variable sizes up to the path MTU) while datagrams containing voice are relatively small, on the order of only 40 to 200 bytes, depending on the codec. On lower speed links (less than 10 MBPS), the jitter introduced by video to voice can be disruptive, while at higher

speeds the jitter is nominal compared to the jitter requirements of voice. At access network speeds, therefore, [RFC4594] recommends separation of video and voice into separate queues, while at optical speeds [RFC5127] recommends that they use a common queue.

If, on the other hand, the two traffic classes are carrying the same type of application with the same jitter requirements, then giving one preference in this sense does not benefit the higher priority traffic and may harm the lower priority traffic. In such a case, using separate policers and a common queue is a superior approach.

2.2. Capacity admission control

There are at least six major ways that capacity admission is done or has been proposed to be done for real-time applications. Each will be described below, then Section 3 will judge which ones are likely to meet the requirements of the Admitted Telephony service class. These include:

- Drop Precedence used to force sessions to voluntarily exit,
- Capacity admission control by assumption or engineering,
- Capacity admission control by call counting,
- End-point capacity admission performed by probing the network,
- Centralized capacity admission control via bandwidth broker, and
- Distributed capacity admission control using protocols such as RSVP or NSIS.

The problem with capacity admission by assumption, which is widely reployed in today's VoIP environment, is that it depends on the assumptions made. One can do careful traffic engineering to ensure needed bandwidth, but this can also be painful, and has to be revisited when the network is changed or network usage changes.

The problem with dropping traffic to force users to hang up is that it affects a broad class of users - if there is capacity for N calls and the N+1 calls are active, data is dropped randomly from all sessions to ensure that offered load doesn't exceed capacity. On very fast links, that is acceptable, but on lower speed links it can seriously affect call quality.

There are two fundamental problems with depending on the endpoint to

perform capacity admission; it may not be able to accurately measure the impact of the traffic it generates on the network, and it tends

to be greedy (eg., it doesn't care). If the network operator is providing a service, he must be able to guarantee the service, which means that he can't trust systems that are not controlled by his network.

The problem with mechanisms that don't enable the association of a policy with the request is that they don't allow for multi-policy services, which are becoming important.

The operator's choice of admission procedure must, for this DSCP, ensure the following:

- The actual links that a session uses have enough bandwidth to support it.
- New sessions are refused admission if there is inadequate bandwidth under the relevant policy.
- If multiple policies are in use in a network, that the user is identified and the correct policy applied.
- Under periods of network stress, the process of admission of new sessions does not disrupt existing sessions, unless the service explicitly allows for disruption of calls.

2.3. Recommendations on implementation of an Admitted Telephony Service Class

It is the belief of the authors that either PHB implementation described in Section 2.1, if coupled with adequate AAA and capacity admission procedures as described in Section 2.2, are sufficient to provide the services required for an Admitted Telephony service class. If preemption is required, as described in section 2.3.5.2 of [RFC4542], this provides the tools for carrying out the preemption. If preemption is not in view, or if used in addition to preemptive services, the application of different thresholds depending on call precedence has the effect of improving the probability of call completion by admitting preferred calls at a time that other calls are being refused. Routine and priority traffic can be admitted

using the same DSCP value, as the choice of which calls are admitted is handled in the admission procedure executed in the control plane, not the policing of the data plane.

On the point of what protocols and procedures are required for authentication, authorization, and capacity admission, we note that clear standards do not at this time exist for bandwidth brokers, NSIS has not at this time been finalized and in any event is limited to unicast sessions, and that RSVP has been standardized and has the

relevant services. We therefore recommend the use of RSVP at the UNI. Procedures at the NNI are business matters to be discussed between the relevant networks, and are recommended but not required.

3. Summary: changes from RFC 4594

To summarize, there are two changes to [RFC4594] discussed in this document:

Telephony class: The Telephony Service Class in RFC 4594 does not involve capacity admission, but depends on application layer admission that only estimates capacity, and that through static engineering. In addition to that class, a separate Admitted Telephony Class is added which performs capacity admission dynamically.

Video classes Capacity admission is added to three video classes. These are the Interactive Real-Time Traffic class, Broadcast TV class when used for video on demand, and the Multimedia Conferencing class.

4. IANA Considerations

This note requests that IANA assign a DSCP value to a second EF traffic class consistent with [RFC3246] and [RFC3247] in the "Differentiated Services Field Codepoints" registry. It implements the Telephony Service Class described in [RFC4594] at lower speeds and is included in the Real Time Treatment Aggregate [RFC5127] at higher speeds. The recommended value for the code point is 101100, paralleling the EF code point, which is 101110. The code point

should be referred to as VOICE-ADMIT.

This traffic class requires the use of capacity admission such as RSVP services together with AAA services at the User/Network Interface (UNI); the use of such services at the NNI is at the option of the interconnected networks.

5. Security Considerations

A major requirement of this service is effective use of a signaling protocol such as RSVP, with the capabilities to identify its user either as an individual or as a member of some corporate entity, and assert a policy such as "routine" or "priority".

This capability, one has to believe, will be abused by script kiddies

and others if the proof of identity is not adequately strong or if policies are written or implemented improperly by the carriers. This goes without saying, but this section is here for it to be said...

6. Acknowledgements

Kwok Ho Chan, Georgios Karagiannis, Dan Voce, and Bob Briscoe commented and offered text. The impetus for including Video in the discussion, which initially only targeted voice, is from Dave McDysan,

7. References

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