                An EF DSCP for Capacity-Admitted Traffic
                draft-baker-tsvwg-admitted-voice-dscp-01

Status of this Memo

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Abstract

   This document requests a DSCP from the IANA for a class of real-time
   traffic conforming to the Expedited Forwarding Per Hop Behavior and
   admitted using a CAC procedure involving authentication,
   authorization, and capacity admission, as compared to a class of
   real-time traffic conforming to the Expedited Forwarding Per Hop
   Behavior but not subject to capacity admission or subject to very

coarse capacity admission.

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 DoD's Assured Service
(which is similar to MLPP), or e-911. These do not need separate
DSCPs or separate PHBs that are separate from each other, but they
need a traffic class from which they can deterministically obtain
their service requirements from including SLA matters.

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1. Introduction

This document requests a DSCP from the IANA for a class of real-time traffic conforming to the Expedited Forwarding [RFC3246][RFC3247] Per Hop Behavior and admitted using a CAC procedure involving authentication, authorization, and capacity admission, as compared to a class of real-time traffic conforming to the Expedited Forwarding Per Hop Behavior but not subject to capacity admission or subject to very coarse capacity admission.

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 DoD's Assured Service (which is similar to MLPP and uses preemption), 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, NS/EP, 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.

This DSCP applies to the Telephony Service Class described in [RFC4594]. 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 this traffic to be carried in the Real Time Traffic Class described in [I-D.ietf-tsvwg-diffserv-class-aggr].

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 DS-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 Codepoint (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, which includes concepts of
   authorization (an identified and authenticated user is determined
   to also be authorized to use the service) and capacity admission
   (at the present time, under some stated policy, capacity exists to
   support the call). 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) which provides connectivity
   services to other users.

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.

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".

```
                                 _.--------.
                            ,-''            `--.
                         ,-'                    `-.
                       ,-'                         .
           ,-------.  ,',-------.                    .
         ,'         . ,','        `.                   .
        /  User      \ UNI / /  Service   \             \
       (    Network    +-----+    Network    )           `.
        \             /  ;    \            /               :
         `.          ,'   ;     `.        .+                :
          '-------'    /       '-------'   \ NNI            \
                    ;                        \                :
                   ;        "The Internet"    \  ,-------.     :
                   ;                          +'         `.     :
       UNI: User/Network Interface          /   Service   \     |
                   |                        (     Network    )    |
       NNI: Network/Network Interface        \             /     |
                   :                          +.          ,'      ;
                   :                          /  '-------'        ;
                   :                         /                    ;
           ,-------.     \          ,-------. / NNI               /
          ,'         `.    :       ,'          `+                ;
         /  User       \ UNI   /  Service   \                   ;
        (    Network    +-----+    Network    )               ,'
         \             /   \ \              /                /
          `.         ,'     `.`.          ,'              ,'
           '-------'          `.'-------'               ,'
                       `-.                          ,-'
                          `--.                   _.-'
                             `--------''
```
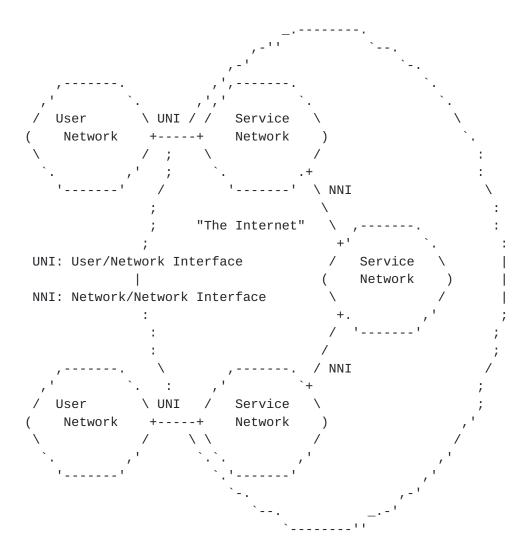
                    Figure 1: UNI and NNI interfaces

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 is depending on specific engineering by
   the service provider that may not be present.

   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-on-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 routine customer-to-customer IP telephony sessions.

1.3. Proposed Solution

The IETF is asked to differentiate, in the Telephony Service, between sessions that are originated without capacity admission or using traffic engineering and sessions that are originated using more robust capacity admission procedures. Sessions of the first type use a traffic class in which they compete without network-originated control as described in Section 2.2.1 or Section 2.2.2, and in the worst case lose traffic due to policing. Sessions of the second type cooperate with network control, and may be given different levels of preference depending on the policies that the network applies. In order to provide this differentiation, the IETF requests that the IANA assign a separate DSCP value to admitted sessions using the Telephony service (see Section 4).

2. Implementation of the Admitted Telephony Service Class

2.1. Potential implementations of EF in this model

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

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

o 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, including a widely reported test for NS/EP services that implemented the policing model and described it as an implementation of the multi-priority model, and discussion in other environments of the intermixing of voice and video traffic at relatively low bandwidths in the policing model.

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.

```
                                          .
                 policers     priorities  |`.
          EF ---------> <=> ----------||----+  `.
                                    high|    `.
          EF2---------> <=> ----------||----+     .'-----------
                              .            medium   .'
            rate queues  |`.            +-----+ .' Priority
          AF1------>||----+  `.        /  low |'   Scheduler
                         |    `.    /
          AF2------>||----+      .'-+
                         |   .'
          CS0------>||----+ .' Rate Scheduler
                         |'   (WFQ, WRR, etc)
```
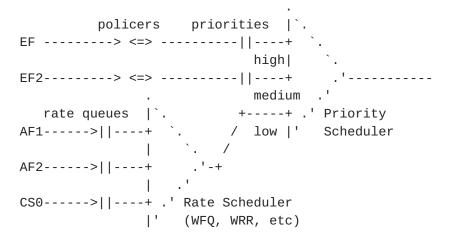
              Figure 2: Implementation as a data plane priority

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.

```
                   policers     priorities  .
          EF ---------> <=> -------\         |`.
                                    --||----+  `.
          EF2---------> <=> -------/    high|    `.
                          .                 |    .'--------
            rate queues  |`.            +-----+   .'
          AF1------>||----+  `.        /  low | .' Priority
                         |    `.    /         |'   Scheduler
          AF2------>||----+      .'-+
                         |   .'
          CS0------>||----+ .' Rate Scheduler
                         |'   (WFQ, WRR, etc)
```

              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 containing video data are relatively large (generally the
size of 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
[I-D.ietf-tsvwg-diffserv-class-aggr] 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 five major ways that capacity admission is done or has been
proposed to be done in Internet Voice applications:

o Capacity admission control by assumption,

o Capacity admission control by call counting,

o End-point capacity admission performed by probing the network,

o Centralized capacity admission control, and

o Distributed capacity admission control

2.2.1. Capacity admission control by assumption

The first approach is to ignore the matter entirely. If one assumes
that the capacity available to the application is uniformly far in
excess of its requirements, it is perhaps overhead that can be
ignored. This assumption is currently made in Internet VoIP
offerings such as Skype and Vonage; the end user is responsible to
place his service on a LAN connected to the Internet backbone by a
high speed broadband connection and use capable ISPs to deliver the
service. There is an authorization step in the sense that the
service ensures that the user pays his bills, but no capacity
admission is considered because there is a clear separation from the

voice application service provider admitting the calls and the access
network provider admitting the traffic. The two have no way of
knowing about each other, except maybe in the abstract sense.

2.2.2. Capacity admission control by call counting

The H.323 gatekeeper, originally specified in 1996, operates on the
model that the considerations of Section 2.2.1 generally apply, and
that it is therefore sufficient to count calls in order to ensure
that any bottlenecks in the network are never overloaded. . Which
phone is calling which phone is configured information into the
Gatekeeper, ensuring it doesn't admit too many calls across a low
speed link. The area of influence of a Gatekeeper is called a Zone,
and limits how far away a Gatekeeper can influence calls. This is
because call counting doesn't scale when more than one server is
admitting flows across the same limited speed links. This approach
is consistent with the original design of H.323, which in 1996 was a
mechanism for connecting H.320 media gateways across a LAN. VoIP has
come a long way since then, however, and the engineering trade-offs
this approach requires in complex networks are unsatisfactory.

SIP provides the option to go down another path, to admit its servers
at layer 7, have no awareness of lower layer connectivity, resulting
in a divorce from infrastructure knowledge - save for [RFC3312],
which binds the two, but only at the endpoints.

In short, if there is a bottleneck anywhere in the network that might
be used to connect two gatekeepers, SIP proxies that do not implement
or do not configure the use of [RFC3312], or other call management
systems, the amount of traffic between the two must be contained
below that bottleneck even if the normal path is of much higher
bandwidth. In addition, the multiplexing of traffic streams between
different pairs of gatekeepers over a common LAN infrastructure is
not handled by the application, and so must be managed in the
engineering of the network.

2.2.3. End-point capacity admission performed by probing the network

[I-D.briscoe-tsvwg-cl-architecture] is one of many proposals that
have looked at probing of the network by the end system to determine
its capacity to accept a new session. Such proposals have been made
a number of times by the likes of NTT Labs, UIUC researchers, Cisco
Systems (which used its Service Assurance Architecture to probe
capacity using pings and report when network delay variability
increased), and others. Many of the proposals tested in research
have fared reasonably well in high bandwidth environments where
actual network congestion is unusual, but have not scaled down to
slower access links.

The problem has been, in essence, that variable rate codecs can be on the quiet side of the average for lengthy periods of time and then become noisier. New sessions can be disrupted or disrupt existing sessions if they perform their capacity admission procedures at a quiet time and find themselves overrunning the allocated capacity during a noisy time. In addition, for a service in which the network must exercise control and differentiate among users, the users cannot be depended on to differentiate among themselves in the network's favor. The network must manage that service.

For this reason, [I-D.briscoe-tsvwg-cl-architecture] is only proposed as a solution within backbone networks, leaving access networks to provide other forms of capacity admission, and more generally such techniques are only recommended in high bandwidth contexts. What is not addressed, is when these quite times become not-at-all quite due to some event occurring, leading to great amounts of traffic. A means of maintaining existing critical calls is essential to retain a given service. Times of disaster can be such times of extreme bursts of the number of call attempts. Once a call is established, that call needs to be retained.

2.2.4. Centralized capacity admission control

The concept of a Bandwidth Broker was first discussed in the research world surrounding ESNET and Internet II in the late 1990's, and has been discussed in the literature pertaining to the Differentiated Services Architecture [RFC2475]. It is, in short, a central system that performs a variety of services on behalf of clients of a network including applying AAA services (as in [RFC2904])and authorizing them to use specified capacity at specified times. Its strength is that it is relatively simple, at least in concept, and can keep track of simple book-keeping functions apart from network elements such as routers. Its weakness is that it has no idea what the specific routing of any stated data flow is, or its capacity apart from services such as MPLS Traffic Engineering or engineering assumptions specified by the designers of a network, and obtaining that information from the network via SNMP GET or other network management action can impose a severe network overhead, and is obviously not in real-time.

For scaling reasons, operational Bandwidth Brokers generally take on a semi-distributed or fully distributed nature. They are implemented on a per-point-of-presence basis, and in satellite networks might be implemented in each terminal. At this point, they become difficult to operationally distinguish from distributed capacity admission services such as described in Section 2.2.5.

2.2.5. **Distributed capacity admission control**

   The IETF developed the Integrated Services Model [RFC1633]and the
   RSVP capacity admission protocol [RFC2205] in the early 1990's, and
   then integrated it with the Differentiated Services Architecture in
   [RFC2998]. Since then, the IETF has worked to describe a next
   generation capacity admission protocol, which is calls NSIS, and
   which is limited in scope to considering unicast sessions. [RFC4542]
   looked at the issue of providing preferential services in the
   Internet, and determined that RSVP with its defined extensions could
   provide those services to unicast and multicast sessions.

   As with the Bandwidth Broker model, there are concerns regarding
   scaling, mentioned in [RFC2208]. Present implementations that have
   been measured have been found to not display the scaling concerns,
   however, and in any event the use of RSVP Aggregation enables the
   backbone to handle such sessions in a manner similar to an ATM
   Virtual Path, bundling sessions together for capacity management
   purposes.

2.3. **Prioritized capacity admission control**

   Emergency Telecommunication Service, the US DoD's Assured Service,
   and e-911 each call for some form of prioritization of some calls
   over others. Prioritization of the use of bandwidth is fundamentally
   a matter of choices - at a point where one has multiple choices,
   applying a policy that selects among them. In the PSTN, GETS
   operates in favor of an authorized caller either by routing a call
   that would otherwise be refused by a path unavailable to the general
   public or by queuing the call until some existing call completes and
   bandwidth becomes available. e-911 is similar, but the policy is
   based on the called party, the emergency call center. MLPP operates
   by preempting an existing call to make way for the new one.

   In the Internet, routing is not performed on a per-call basis, so,
   apart from interconnections to the PSTN, re-routing isn't an option.
   On the other hand, in the Internet there are more classes of traffic
   than in the PSTN. In the PSTN, all calls are uses of circuits, while
   in the Internet some bandwidth is always reserved for elastic
   applications - at least, it must be available for routing, and there
   is generally significant consideration of the web, instant messaging,
   and other applications. In essence, any capacity admission policy
   that differentiates between calls has the option of temporarily
   borrowing bandwidth from the capacity reserved for elastic traffic by
   accepting new sessions under some prioritized policy while refusing
   sessions of lower priority because the threshold at that priority has
   been reached.

For example, regardless of the type of capacity admission that is used (apart from "no admission process"), one might admit prioritized sessions using a higher bandwidth threshold than one admits lower priority sessions.

If capacity admission as described in Section 2.2.2 is in use, the thresholds must be set low enough that bandwidth would be available anywhere in the network. This greatly limits the utility of such a service due to the level of bandwidth waste that results.

If capacity admission as described in Section 2.2.3 is in use, then either multiple thresholds must be applied in marking the traffic, multiple traffic marks must be applied, or there must be multiple ways to interpret the result. In any event, this is only applicable in domains in which the law of large numbers applies.

If capacity admission as described in Section 2.2.4 is in use, thresholds can be applied related to a general policy or SLA, or related to the network ingress and egress in use. It requires them to maintain state regarding network traffic routing separate from the network; to the extent that is variable, it requires direct monitoring in the OSS.

If capacity admission as described in Section 2.2.5 is in use, thresholds can be applied to the critical points of the path that the traffic in question actually takes because one is asking the equipment that the path traverses.

3. Recommendations on implementation of an Admitted Telephony Service Class

It is the belief of the authors that either data plane PHB described in Section 2.1, if coupled with adequate AAA and capacity admission procedures as described in Section 2.2.5, are sufficient to provide the services required for an Admitted Telephony service class. If preemption is in view, as described in section 2.3.5.2 or [RFC4542], this provides the tools for carrying out the preemption. If preemption is not in view, or 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.

4. IANA Considerations

This note, fundamentally, requests IANA the assign a DSCP value to a
second EF traffic class consistent with [RFC3246] and [RFC3247] and
implementing the Telephony Service Class described in [RFC4594] at
lower speeds and [I-D.ietf-tsvwg-diffserv-class-aggr] at higher
speeds. This new 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. The recommended value
for the code point 101100, paralleling the EF code point, which is
101110, and both of which are allocated from Pool 1 as described in
[RFC2474].

The code point should be referred to as EF-ADMIT.

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...

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