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F. Baker  
J. Polk  
Cisco Systems  
M. Dolly  
AT&T Labs  
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**DSCPs for Capacity-Admitted Traffic  
draft-ietf-tsvwg-admitted-realtime-dscp-01**

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Abstract

This document requests two DSCPs from the IANA for real-time traffic classes similar to voice and video, 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 separate them from each other, but they need a traffic class that separates them from the traffic not subject to admission control, from which they can deterministically obtain their service requirements, including SLA matters.

#### 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 two DSCPs from the IANA for two classes 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, 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.

One of these DSCPs applies to the Telephony Service Class described in [[RFC4594](#)]. The other applies to the Multimedia Conferencing Service Class described in the same document. Other video classes are not believed to be required by the targeted services and to not have the current problem of confusion with unadmitted 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 [[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".





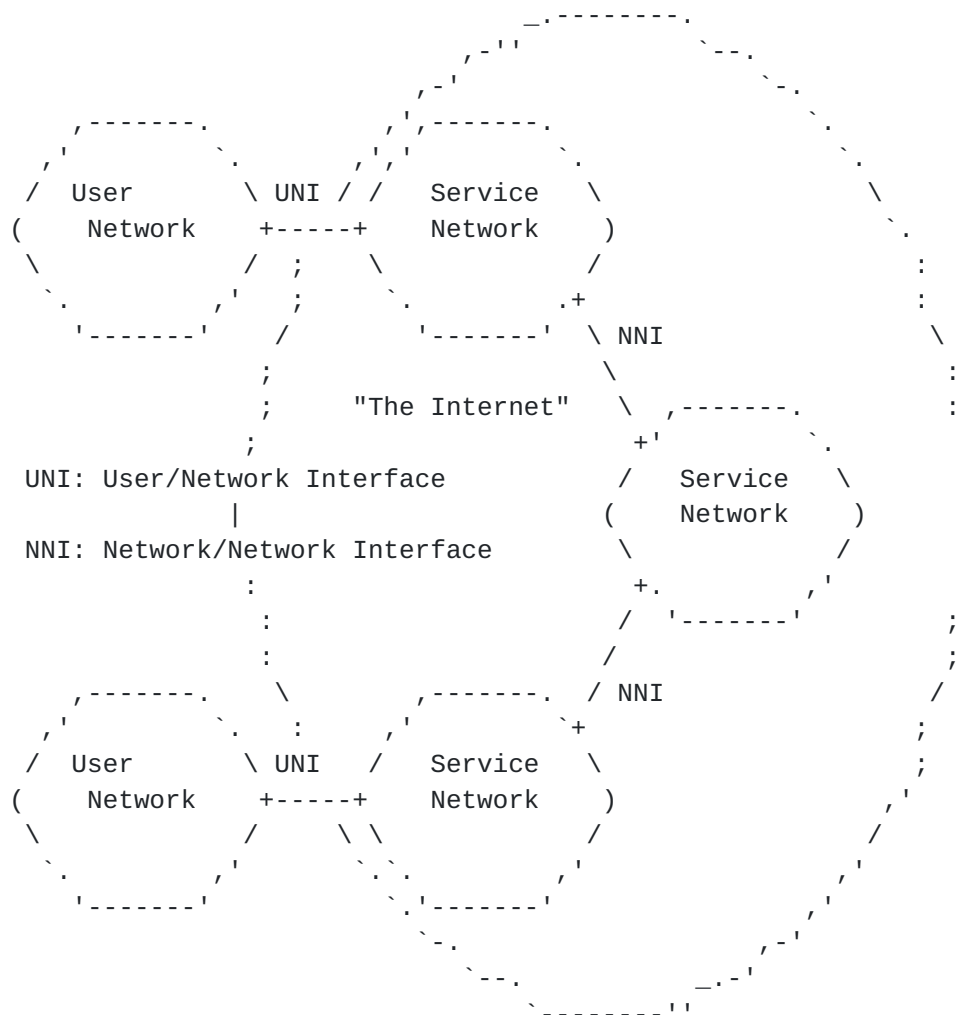


Figure 1: UNI and NNI interfaces

### 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 or Video-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 Service Classes**

### **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.

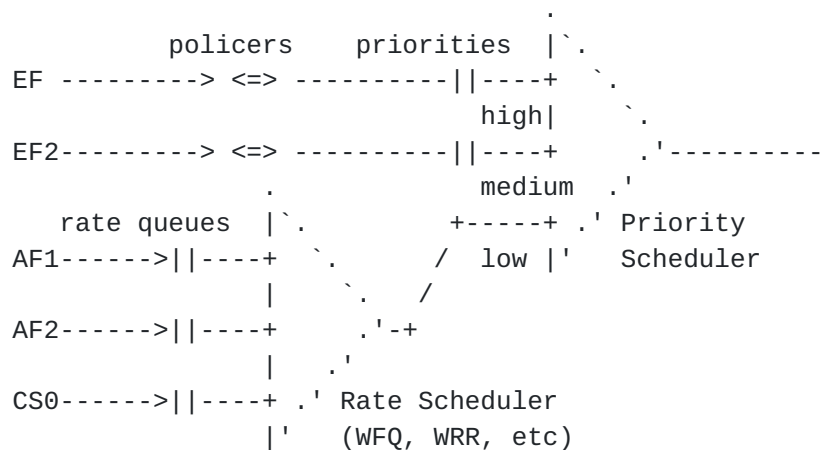


Figure 2: Implementation as a data plane priority

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.

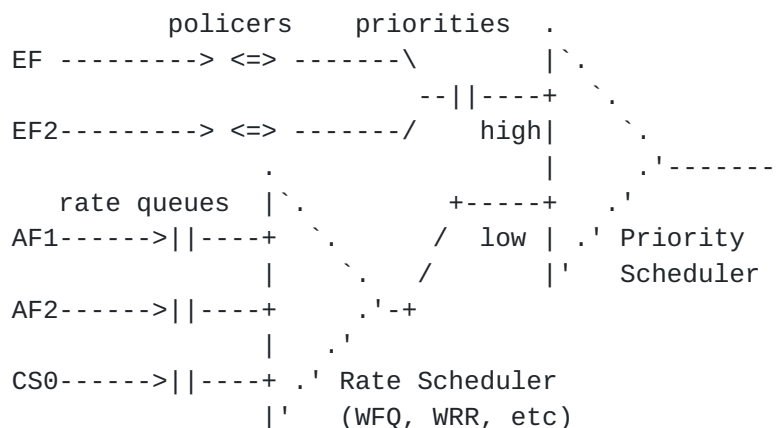


Figure 3: Implementation as a data plane policer



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 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 real-time 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





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**

The IETF started looking into the use of Pre-Congestion Notification mechanism to full fill the need of admission control for real-time traffic. The main contribution of this work for admission control is to allow the network to provide the network's pre-congestion information using encoding of a field in the IP header. This network pre-congestion information is then used for making admission control decisions. With the decision influenced by this network pre-congestion notification information and any applicable policy information with possible user credentials and situational information. The pre-congestion notification mechanism does not



limit the placement of the admission control decision point or the signaling protocol used.

The overview of one of the current proposals is provided by [[I-D.chan-pcn-problem-statement](#)]. With the pre-congestion notification encoding described in [[I-D.briscoe-tsvwg-cl-phb](#)]. An initial deployment model provided by [[I-D.briscoe-tsvwg-cl-architecture](#)]. Another proposal is embodied in [[I-D.charny-pcn-single-marking](#)]. Similar approaches have been proposed in [[I-D.morita-tsvwg-pps](#)] and its related drafts, by Ivars and Karlsson in their PBAC work, and many others.

Current simulation work have shown the pre-congestion notification mechanism works well with large number of audio and video flows on high speed lines. This work is ongoing.

#### **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 been working on a next



generation signaling protocol called NSIS that can be used for capacity admission protocol, 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 and an Admitted Multimedia Conferencing Service Class. If preemption is in view, 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 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 requests that IANA assign a DSCP value to a second EF traffic class consistent with [[RFC3246](#)] and [[RFC3247](#)]. It implements the Telephony Service Class described in [[RFC4594](#)] at lower speeds and is included in the Real Time Treatment Aggregate [[I-D.ietf-tsvwg-diffserv-class-aggr](#)] at higher speeds. The recommended value for the code point 101101, paralleling the EF code point, which is 101110, and is allocated from the pool set aside for experimental use but potentially available for standards use as well in [[RFC2474](#)]. The code point should be referred to as VOICE-ADMIT.

Additionally, it requests that IANA assign a DSCP value to a traffic class consistent with [[RFC2597](#)]. It implements the Multimedia Conferencing Service Class described in [[RFC4594](#)] at lower speeds and is included in the Real Time Treatment Aggregate [[I-D.ietf-tsvwg-diffserv-class-aggr](#)] at higher speeds. The recommended value for the code point 100101, paralleling the AF42 code point, which is 100100, and is allocated from the pool set aside for experimental use but potentially available for standards use as well in [[RFC2474](#)]. The code point should be referred to as VIDEO-ADMIT.

Both of these new traffic classes require 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 offered some textual comments and rewrote [Section 2.2.3](#). Georgios Karagiannis offered additional comments on the same section. The impetus for including Video in the discussion, which initially only targetted voice, is from Dave McDysan.

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#### Authors' Addresses

Fred Baker  
Cisco Systems  
Santa Barbara, California 93117  
USA

Phone: +1-408-526-4257  
Email: fred@cisco.com

James Polk  
Cisco Systems  
Richardson, Texas 75082  
USA

Phone: +1-817-271-3552  
Email: jmpolk@cisco.com





Martin Dolly  
AT&T Labs  
Middletown Township, New Jersey 07748  
USA

Phone: +1-732-420-4574

Email: [mdolly@att.com](mailto:mdolly@att.com)

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