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Hussein Eissa
Computers & Systems Department., Electronics Research Institute.

Tarek Kamel
Computers & Systems Department., Electronics Research Institute.

A NEW IP MULTICAST QoS MODEL FOR REAL-TIME AUDIO/ VIDEO TRAFFICS ON THE IP BASED NETWORKS

نموذج جديد لتحقيق جوده الخدمة في نقل الصوت و الصورة في الزمن الحقيقي عبر الشبكات التي تعتمد علي بروتوكول الانترنيت

Hussein Sherif Eissa, and Tarek Kamel
Computers & Systems Dept., Electronics Research Institute.

خلاصة :
يناقش هذا البحث النقاط الخاصة بمتطلبات تطبيقات الزمن الحقيقي وتعارضها مع متطلبات مصادر الشبكات. أيضا سوف يتم استعراض نماذج جودة الخدمة المقترحة و سيتم إجراء مقارنه بينهم. وفي النهاية سيتم توصيف النموذج الجديد المقترح تفصيليا و الذي سيقوم بتقديم ضمانات كافية للأداء بين المرسل و المستقبل وذلك لخدمة معلومات الصوت و الصورة الخاصة بالزمن الحقيقي عبر الشبكات التي تعتمد علي بروتوكول الانترنيت.

ABSTRACT

On congested links/routers, Best-Effort delivery will seriously degrade the service provided to the real-time traffic. Real-time audio/video traffics could not depend on the Best-Effort delivery of datagrams, and therefore, like the Differentiated-services (Diffserv) Model, there is a need for a model that can give a per-flow service assurances. This paper discusses issues related to real-time audio/video applications requirements, the trade off between them and the network resources requirements. This paper also states the IP QoS proposed models and presents a comparison between them. Finally, it describes in details the new proposed model which provides end-to-end performance guarantees to support the real-time audio/video traffics on the IP based networks. The new proposed model relies on having an acceptable delay, packet loss, and MTU (Maximum Transfer Unit). It also has an acceptable consumed bandwidth and buffers specially at the core routers in which the congestion usually occurs.

Keywords: Real-time Audio/Video traffic, IP Multicast Mechanism, IP QoS Mechanisms, IS, Diffsev, Best-Effort.

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1. INTRODUCTION

Currently, there is a great need and a big trend for gaining access to real-time audio/video applications on the Internet. The Internet is a web of different, intercommunicating networks funded by both commercial and governmental organizations. The Internet also has spread overseas to connect the networks all over the world.

Real-time traffic is the road to the future; it is the new wave of information technology development. It creates the environment of sensation and integration where text, images, audio and video integrate and act together to give a high degree of visualization.

In this paper, the importance of the Real-time Audio/Video traffic across the Internet will be highlighted in Sections 1 and 2. The basic IP QoS proposed models will be outlined and a comparison between them will be presented in Section 3. And then, in Section 4, a new proposed IP Multicast QoS model would be described in details.

2. REAL-TIME TRAFFIC REQUIREMENTS

2.1 Overview

There are two types of traffic from the reality point of view; the first type is Real-time traffic in which the delay between sending and receiving the packets must be under certain time constraints.

The second type is Non-real-time traffic in which the delay between sending and receiving the packets is not a matter of discussion.

The present paper is concerned with the first type, because of its increasing importance across the Internet and the serious problems that face the multimedia communications. The following sections describe these problems in details.

2.2 Performance Parameters

The following sections highlight the main performance parameters that affect the real-time traffics across the Internet. Sections 2.2.1, 2.2.2, and 2.2.3 discuss the application requirements parameters. Section 2.2.4 discusses the network requirements parameters.

2.2.1 Delay

The end-to-end delay is always a critical factor in the usability of a real-time audio/video applications and should be kept below certain limits. There are various types of delay such as processing, propagation, queuing and Jitter. The most important ones to be analyzed are the queuing delay and the Jitter, because they may affect the quality of the received audio/video streams and delay the packets to the limit that they will be dropped because they are too late to be played back.

2.2.2 Packet Loss

Packet Loss can occur for a number of reasons; the first is the congestion of Links/Routers in which the packets may be ignored to solve the problem of congestions. The second is because of the delay, in which the packets may arrive to the destinations but too late to be played back. So, it will be ignored at the destinations.

2.2.3 Maximum Transfer Unit (MTU)

MTU could be maximized to decrease the overhead percentage, but this will decrease the real-time stream quality when the packet loss occurs. For example, in case of Ethernet frames, the overhead percentage equals 5.1% in case of 512 byte MTU data, and 1.7% in case of 1500 bytes MTU data. So, from this example, it is clear that loosing one packet from the first case means loosing less data than the second one, but the second is much less overhead percentage than the first.

2.2.4 Networks Resources

The real-time applications usually consume a lot of bandwidth and buffers. This means that there is a trade off between the networks requirements and real-time applications requirements.

From the previous sections, to enhance the quality of received audio/video streams the delay, the packet loss, and the MTU should be minimized, but this needs a huge amount of bandwidth and buffers. Therefore, there is a trade off between the real-time applications requirements and the Networks resources requirements and should be compromised. The optimal solution for real-time traffic should be a model, which relies on an acceptable delay, packet loss, and MTU, which result in an acceptable consumed amount of the bandwidth and buffers specially at the core routers in which the congestion usually occurs.

3. IP QoS MODELS

3.1 Overview

The current Internet consists of networks built from various link-layer technologies and relies on the Internet Protocol (IP) to internetwork between them. IP makes no assumptions about the underlying protocol stacks and offers an unreliable, connectionless network-layer service that is subjected to packet loss, reordering and packet duplication, all of which, together with queuing delay in routers, will increase with the network loads. Because of the lack of any firm guarantee, the traditional IP delivery model is often referred as Best-Effort (BE).

For traditional non-real-time Internet traffic such as File Transfer Protocol (FTP), the Best-Effort delivery model of IP has not been a problem. However, as moving further into the age of multimedia communications, many real-time applications are being developed which are delay-sensitive to the point where the best-effort delivery model of IP can be inadequate. So, there is a great need for a model, in which, it provides many applications with additional classes offering enhanced quality of service (QoS) with regard to bandwidth, buffers, packet queuing delay, and packet loss.

3.2 Integrated Services (IS) Model

This Model was proposed to change the existing Best-Effort service [1]. It provides the ability for applications to choose among multiple, controlled levels of delivery service for their data packets. The IS Framework includes four components; the packet scheduler, the admission control, the classifier, and the reservation setup protocol [2,3].

- Packet Scheduler

It manages the forwarding of different packet streams using a set of queues and perhaps other meehanisms like timers. It should be implemented at the point where packets are queued.

- Classifier

It classifies each incoming packet to a particular service class, so as to be scheduled appropriately for transmission by the packet scheduler. The packets are classified based on certain information that is present in the packet header. All the packets belonging to the same class are treated similarly.

- Admission Control

It implements the decision algorithm that hosts/routers use to determine whether a new flow can be granted the requested QoS without impacting earlier guarantees. The decision to accept or reject a reservation request is made on the basis of the requested QoS and the available QoS on the outgoing link.

- Reservation Setup Protocol

It is responsible for creating the flow specific information in the end-hosts/routers all along the path from the source to the destination. The applications specify the required QoS and then the reservation protocol uses these data to reserve the resources in the routers along the path from source to the destination.

3.3 Differentiated Services (Diffserv) Model

The goal of the Diffserv architecture is the enhancement cf the IP Protocol for the support of quality of service in a scaleable manner, i.e. without relying on either signaling protocols or per flow state information [4]. This architecturc employs a small set of building blocks from which a variety of services can be built. The Diffserv architecture supports both end-end and intra-domain reservations and it has been designed for both parameters and class differentiation based requirements [5].

The Diffserv architecture is based on a simple model where traffic entering a network is classified and possibly conditioned at the boundaries of the network, and assigned to different behavior aggregates. Each behavior aggregate is identified by a single Diffserv codepoint. Within the core of the network, packets are forwarded according to the Per-Hop Behavior (PHB) associated with the Diffserv codepoint. In the following sections, the key components within a Diffserv region [6] will be described.

- Classifier

It classifies the packets based on the content of some portion of the packet header. There are two types of classifiers. The BA (Behavior Aggregate) classifier, which classifies the packets, based on the Diffserv codepoint only. The second type is the MF (Multi-Field) classifier, which classifies the packets based on the values of combination of one or more header fields, such as source address, destination address, Diffserv codepoint, protocol ID, source port and destination port numbers.

- Meter

It measures the temporal properties of the stream of packets selected by the classifier against a traffic profile specified in a TCA (Traffic Conditional Agreement). Then, it passes the information to other conditioning functions to trigger a particular action for each packet, which is either In- or Out-of-profile.

- Marker/Re-marker

It sets the Diffserv field of the packet to a particular codepoint. Each codepoint is related to a selected PHB. Also, when the marker changes the codepoint of the packet, it is said to have ***re-marked*** the packet.

- Shaper

It delays some or all the packets in order to bring the stream into compliance with a traffic profile. A shaper usually has a finite-size buffer, and packets may be discarded if there is not a sufficient buffer space to hold the delayed packets.

- Policer

It discards or re-marks some or all the packets in order to bring the stream into compliance with a traffic profile. This process is known as policing the stream [5].

3.4 Comparative Analysis among BE, IS and Diffserv Models

From the previous sections, it is clear that the main difference between both IS and Diffserv models is that the IS model relies on the signaling protocol or per-flow state information. This causes a lot of problems in the routers along the path (specially the core routers) such as congestions and overloading because of state information keeping process all the path long during the session.

Therefore, a brief comparison is shown in Table 1. Both IS and Diffserv guarantee the quality of the services, and both BE and Diffserv are scaleable. Finally, both IS and Diffserv have a setup between the source and the destination, but it is a dynamic per session for IS model and it is statie for long term for Diffserv model.

Table 1: Comparative Analysis among BE, IS and Diffserv Models

	BE	IS	Diffserv
QoS Guarantees	No	Yes	Yes
Scaleability	Yes	No	Yes
Setup	No	Dynamic per Session	Static for long term

4. A NEW PROPOSED MODEL

4.1 Overview

From the previous seetions, it is clear that the Diffserv model is sealeable and guarantees the required QoS. However, it does have neither assumption regarding minimal latency, nor assumption regarding the jitter effects [7]. Moreover, it has no fixed architecture in which the locations of the traffic conditioners and the classifiers are determined. So, there is a big need to have such a model that have a simple, scaleable, delay-sensitive, and networks resources saver to provide the necessary level of services to Real-time Audio/Video streams across the IP based networks in order to maintain an expected quality level. The new proposed model guarantees the required quality by using some of the Diffserv Simple blocks at edge devices. It also, uses the IP Multicast mechanism to save more networks resources [8]. This new model relies on shaping the out-of-profile streams, dropping them in case of the real-time marked packets, and re-marking them in case of the non real-time marked packets. By implementing these simple blocks in the edge devices only, the overload on the core routers will be decreased and minimized

4.2 General Model Architecture

In this section, the overall structure of the new model will be described (Fig. 1). In this model, three types of traffic are proposed; the Real-time traffic, Non-real-time traffic, and Best-effort traffic.

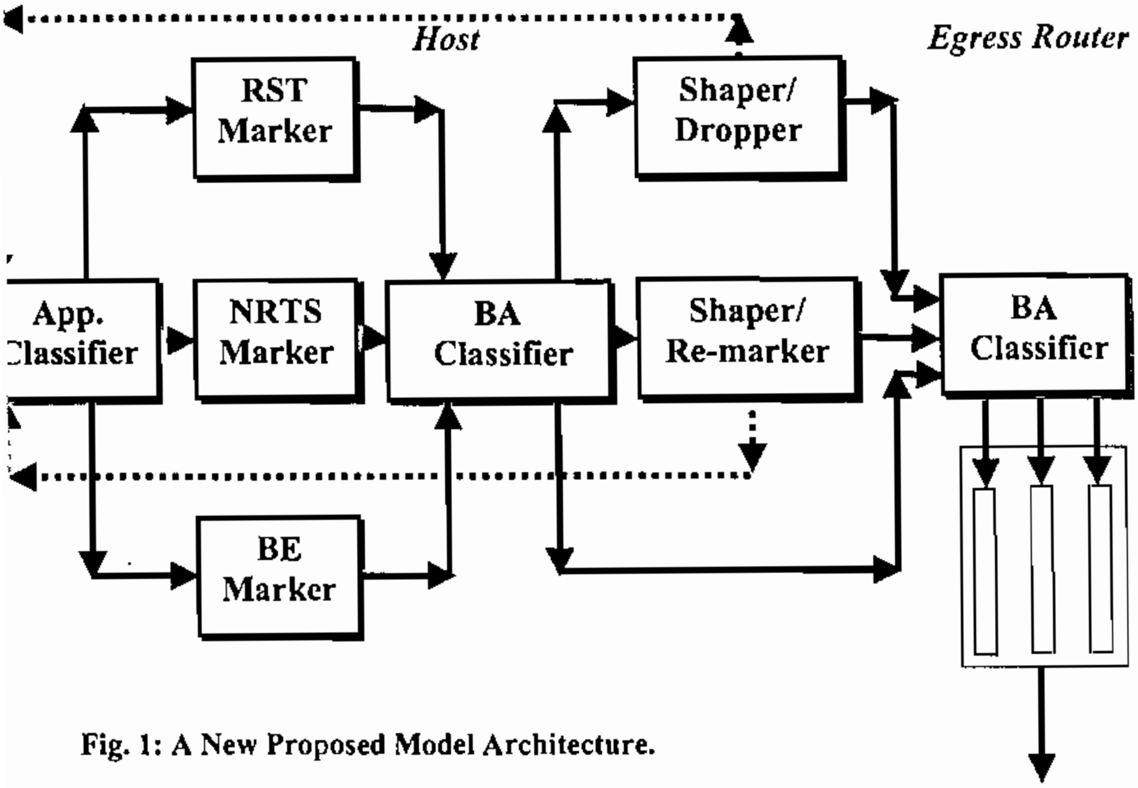

Fig. 1: A New Proposed Model Architecture.

- *Applications Classifier:*

It classifies the streams based on the used applications. For example, if the used application is a Videoconferencing application, then it passes the packets to the marker to mark it with real-time applications codepoint, and if the used application is a Web Browser, then it passes the packets to the marker to mark it with the Best-effort codepoint.

- *Marker: (inside the sender host only)*

Based on the used application, it marks the packets with 3 different values in the precedence field (3 bits) located inside the TOS (Type Of Service) field in the IPv4. The suggested values for precedence field is described in Table 2.

Table 2: Proposed Precedence Values

Precedence Value	*Description*
7	Network Control
6	Real-time Applications
5	Non-real-time Applications
4→1	Unused
0	Best-Effort Applications

Then the Marker passes the packets to the BA Classifier.

- BA Classifier: (inside the sender host only)
The Classification is based on the value of the precedence field only. So, it classifies the outgoing packets into 3 categories:
-RTS (Real-Time Stream)
-NRTS (Non-Real-Time Stream)
-BE (Best Effort Stream)

Then the Classifier, as shown in Figure 1, passes the RTS packets to the Shaper/Dropper block, and passes the NRTS to the Shaper/Re-marker block, and finally passes the BE packets to the Egress Router.

- Shaper/Dropper: (inside the sender host only)
The Shaper does not delay the incoming packets, but it checks if it conforms to some defined traffic profile. The Shaper here relies on single token bucket measurement mechanism. In this token bucket, as shown in the figure, the tokens are inserted into the bucket with token generation rate (r) in kb/s. The depth of the bucket is the burst size (b) in bytes. When RTS traffic arrives at the bucket, if sufficient tokens are available then the traffic is said to **conform** and the corresponding number of tokens are removed from the bucket. If insufficient tokens are available then the traffic is said to **exceed** then drop them immediately and send an Error_Drop message to the source Application. Finally, the shaper/dropper block passes the conform packets to the Egress Router.

-Shaper/Re-marker: (inside the sender host only)
This Shaper delays the incoming packets to conform it to some defined traffic profile. It also relies on single token bucket mechanism. When NRTS traffic arrives at the bucket, if sufficient tokens are available then the traffic is said to **conform** and the corresponding number of tokens are removed from the bucket. If insufficient tokens are available then the traffic is said to **exceed** then re-mark them with precedence value 0 (i.e. as a Best-Effort traffic) and send an Error_Re-mark message to the source Application. Finally, the shaper/dropper block passes the packets to the Egress Router.

- Egress Router: (inside the Domain)
The boundary node, which handles the leaving traffic, from the Domain. It classifies the incoming packets from the Precedence field inside the TOS field in the IPv4 header to three categories; RTS, NRTS, and BE (BE or **exceed** NRTS packets). Then it applies the three queues Priority Queuing, and sends the RTS packets to the highest priority Queue and sends the NRTS to the medium priority queue and finally sends the BE packets to the lowest priority Queue.

4.3 Simulations Modules Implementations for the New Proposed Model

For RTS generators, the real-time Audio/Video streams characteristics will be analyzed experimentally by using Videoconferencing applications (e.g. Microsoft Netmeeting) and capture the generated packets to draw the PDF curves (Packet Size PDF curve, and Packets Intervals PDF curve) for both the Audio and the Video generators. Also For NRTS and BE generators, the non-real-time streams characteristics will be analyzed experimentally (e.g. FTP, Telnet applications) and capture the generated packets to draw the PDF curves (Packet Size PDF curve, and Packets Intervals PDF curve). These PDF eurves are the new proposed model inputs (RTS, NRTS, and BE). To implement the new proposed model, the OPNET MODELER Network Simulator version 6.0 (network simulation software package, www.mil3.com) will be used. After having the PDF curves, the OPNET PDF Editor will be used to draw these curves. The Packet Generator in the Node Editor will be used to create the audio/video packet generators. The generator s generation rate attribute, generation distribution attribute, average packet size attribute, and packet size distribution attribute must be set in the generator module. The ***interarrival pdf attribute*** will be used to set the interarrival times of generated packets to be audio/video Interval distribution. The ***pk size pdf attribute*** will be used to set the size of generated packets to be audio/video Packet Size distribution. To design the Shaper/Dropper & the Shaper/Re-marker blocks, the Single token bucket will be used. As shown in Figure 2, the inputs will be the audio/video packets (the packets arrivals), entering the packet queue. This packet queue is infinite and based on FIFO as a service ordering discipline. At the same time the tokens (here are bytes) are generated by rate of r (tokens rate) to feed the bucket which have a maximum depth of b (bucket size). The server will pass the packet at the packet queue only if there are enough bytes inside the bucket, and then remove these bytes from the bucket. If there are insufficient bytes inside the bucket to pass the packet at the packet queue, the server will drop this packet or send it to the Re-marker block to remark it with BE mark.

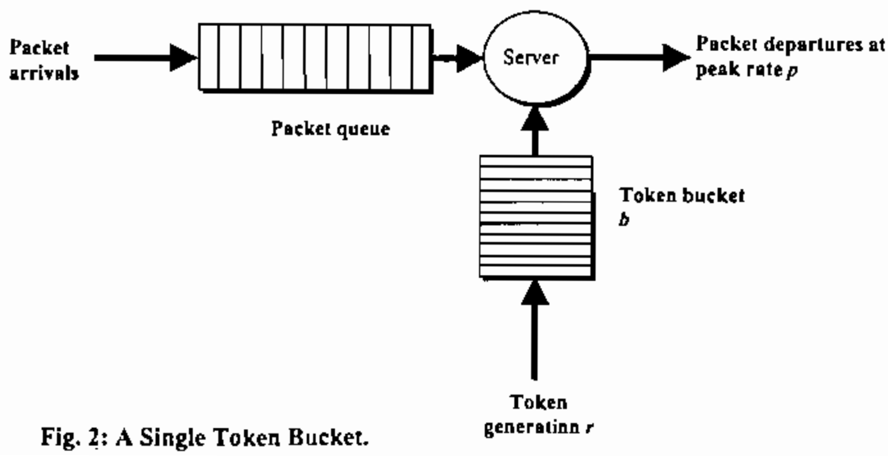

Fig. 2: A Single Token Bucket.

After Designing the Shaper/Dropper and the Shaper/Re-marker blocks, different scenarios will be applied to have the values of tokens rate (r) and the bucket size (b) for both of them. It is clear that by increasing the values of r and b, the packet loss will decrease, but this will consume more reserved bandwidth and buffers. So, the values of r, b, and packet loss should be compromised (i.e. to have minimum values of r and b, which give an acceptable packet loss and minimum MTU value).

To implement the Egress router, the Queue Node in the Node Editor will be used. The ***process model attribute*** will be set to ***Priority Queuing*** and the ***service_rate attribute*** will be sct to the value of reserved bandwidth to the RTS streams. At last, the ***subqueue attributes*** will be used to set two new queues, one for the NRTS and the other is for the BE streams. Then different scenarios will be applied on the proposed model to have general statistics information which are the packet queuing delay, Consumed buffers, Saved bandwidth, Jitter, IP Multicast mechanism pros over the IP Unicast mechanism, and the burstness effects due to the cross aggregations from different domains.

5. SUMMARY

There are two main IP QoS model. The IS model faces some problems such as the scaleability. The other one, Diffserv model, is more simpler and scaleable because it does not rely on either signaling protocols or per flow state information. So, the second model is more applicable in the scale of the Internet. But, the Diffserv model has no assumptions regarding the Queuing and Jitter delays. So, the ncw IP Multicast proposed model is designed to provide a guaranteed QoS for real-time Audio/Video traffics. In this new model, the marker, the BA classifier, and the Shaper/Policer blocks have been used inside the sender host only. Also, The sender host is relying on the IP Multicast mechanism to send its data to all members of the certain group. Finally, the Egress routers, elassify the incoming packets from the sender host (inside the Diffserv domain) and forward them according to their priorities.

6. FUTURE WORK

The performance analysis of the new proposed model has shown that it is recommended to make adds/modifications, to test the model stability by increasing the number of nodes, and to find out the required values of the networks recourses (reserved bandwidth, and buffers) by which the model will give an acceptable end-to-end delay, and Jitter.

The Routing mechanism is the one of the interesting issues that should be studied carefully in the arca of IP QoS. There is a great need to have a routing protocol that supports both the QoS parameters and the IP multicast mechanism. This protoeol should be designed to have the optimum path that is suitable for the real-time streams and other guaranteed services.

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