Sizable Improvement for Reducing Packet Loss Rate in Video Transmission

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Abstract

Internet started as an experimental network for researchers and primarily was used as a medium for applications like file transfer and email. Some of the modern applications like video transmission over IP network require certain guarantees in terms of available network bandwidth. Packet loss directly affects the perceived quality of the video and packet loss can be caused by network congestion, which results in dropped packets.The QoS solutions for video transmission over IP network presently include integrated services (IntServ) [4], differentiated services [5], and multiprotocol label switching (MPLS) [6]. Among them, the DiffServ is regarded as a dominant protocol for its flexibility, scalability, and capability of QoS guarantee. An extensive simulation model has been developed to study the performance and viability for the video transmission over IP Network. The performance of video over DiffServ is studied with extensive simulation results and by comparing the performance of improved token bucket TCM algorithm with existing srTCM (single-rate three color marker) [2] and trTCM (two-rate three-color marker) [3] algorithms over DiffServ network. The paper suggests a mechanism (improved token bucket TCM) in order to improve the quality of video over Internet. The enhancement mechanism shows a sizable improvement in terms of reducing the total packet loss during the video communication.

Keywords - DiffServ , IntServ ,srTCM, trTCM, QoS

1. Introduction

The current Internet delivers one type of service, best effort, to all traffic. Traffic is processed as quickly as possible, but there is no guarantee of the Quality of Service (QoS) in terms of performance for individual or aggregated flows. QoS refers to network performance measures such as service availability, delay, delay-variation (jitter), throughput, and packet loss rate as seen by users and applications. Motivated by the changes in user expectations and Internet applications, there is a growing demand to replace the current best effort service with a model in which users, applications, or individual packets are differentiated based on their service needs. Two broad paradigms for quality of service in future Internet have emerged: Integrated services (IntServ), and Differentiated services (DiffServ). The IntServ model [4] provides quality of service to individual packet flows through resource reservation and admission control mechanisms. The Resource Reservation Protocol (RSVP) [16] is used as the signaling protocol in IntServ. The IntServ RSVP approach requires every router to maintain per-flow state information. For large networks with millions of simultaneous connections, maintaining the flow state information places a huge storage and processing overhead on the routers. To alleviate the difficulties associated with the IntServ model, Differentiated Services [5] was introduced. This

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approach focuses primarily on aggregates of flows in the core routers, with the intention of differentiating between service classes rather than providing absolute per-flow QoS measures. Though the access routers can still process packets on per-flow basis, the core routers do not maintain the per-flow state and process traffic based on a small number of Per Hop Behaviors (PHB) encoded in the packet header [7]. Premium service and Assured service [8] have been proposed within the DiffServ architecture. Premium service provides low-delay and low jitter service by guaranteeing the peak rate of the user flows [9, 10]. A potential disadvantage of Premium service is the weak support for bursts and the fact that users have to pay even if they are not using the bandwidth. The Assured service model tries to offer a service that may not guarantee bandwidth but provides assurance that high priority packets receive preferential treatment over lower priority packets [14]. During congestion periods, packets from the low-priority traffic have higher drop probability than those from the high-profile traffic [15]. However, the Assured service cannot provide any quantitative guarantee to user traffic. In this paper, Investigators introduce a new type of marker system within the IP DiffServ network. This marker is intended to provide a more predictable quality of service than srTCM and trTCM markers in terms of packet loss rate.

The hierarchical structure of MPEG encoding with possible error propagation through the frames impose a great difficulty on sending MPEG video streams over lossy networks. Small packet loss rates may translate into much higher frame error rates. For example, a 3% packet loss percentage could translate into a 30% frame error [12]. This situation may seriously degrade the user perceived quality at the video reproduction. Proposed scheme improve the throughput of MPEG4 video transmissions over IP network. In this scheme, I frames, P frames and B frames are marked as different service priorities according to transmission condition and their contribution to the perceived picture quality. Investigators used authentic MPEG4 video traffic traces to compare the performance of the proposed scheme with other existing RFCs for MPEG video transmission, such as srTCM and trTCM, in a DiffServ network. Results show that proposed scheme improves a sizable improvement in terms of reducing the total packet loss during the video communication.

Packet Loss: Packet loss refers to the number of packets lost during the video transmission. Although, an important QoS parameter, loss is usually quantified in term of loss rate, which is equal to the total amount of lost traffic divided by the total amount of input traffic over the a certain period of time.

2. MPEG Frames

MPEG-4 serves as a standard for multimedia applications. MPEG-4 addresses various key issues such as ease of accessibility in heterogeneous and error prone network environments and compression efficiency. MPEG records only key frames and predicts what the missing frames look like by comparing differences between the key frames. MPEG works differently than other video compression formats. In addition to compressing individual frames, MPEG also compresses between individual frames of a video sequence. MPEG-4 encoders generate three types of frames.

I-frames (intra-coded frames) contain the information that results from encoding a still image. They can be decoded without the need for any other frames.

P-frames (predictively coded frames) require information from previous I-frames and/or P-frames for encoding and decoding.

B-frames (bidirectionally predictively coded frames) require information from the previous and following I- frames and/or P-frames for encoding and decoding. In otherworld B-frames contain differences with the previous and the next frames.

The compression rate, which determines video quality, is highest for B-frames while lowest for I-frames. Dependency relationship between I, P, B frames is illustrated in Fig.1. The encoding pattern of this stream is IBBPBBPBBPBBI.

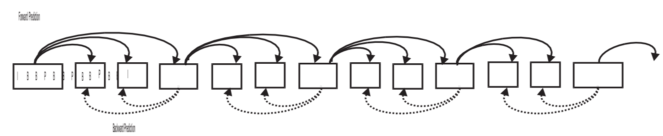

Fig 1 : Relation between I, P, B frames

The preceding I or P-frame serves for forward prediction as illustrated by the solid arrows in Fig.1. B-frames, on the other hand, are intercoded with reference to the preceding I or P frames, which serves for forward prediction, and the succeeding I or P frame, which serves for backward prediction.

Group of Pictures (GOP)

A typical coded video sequence uses all three types of frames. However, a fixed frame-type mixing pattern is usually repeated throughout the entire video stream. More specifically, the sub-sequence of frames from a given I frame up to the next I frame is referred to as a group of pictures (GOP). The pattern of I, P, and B frames that make up a GOP is commonly referred to as GOP structure. A typical group of picture IBBPBBPBBPBB pattern with two P frames in a GOP and two B frames before and after each P frame is illustrated in Fig. 1. MPEG encoder stores only the complete picture of the baseline frame (i.e., I-frame) and partial pictures of any subsequent frames. It does so by breaking the video sequence up into GOP (Group of Pictures). Each GOP generally contains 12 frames and has an I-frame at the beginning. Therefore, I-frames are composed of the first frame in a video sequence and numerous other “baseline” frames within the video stream. Frames following the I-frames are analyzed and only the differences between it and the I-frame are compressed. This increases the compression performance.

3. Legacy Packet Markers

Traditionally, policing or limiting function was implemented by using single-stage token bucket or leaky bucket. In DiffServ networks, it is implemented by using two-stage token bucket algorithms such as srTCM (single-rate three color marker) and trTCM (two-rate three-color marker) . Major differences between srTCM and trTCM are as follows:

The srTCM uses only CIR (comitted information rate) to update two token counters, Tc and Te, while the trTCM uses CIR and PIR (peak information rate) to update token counters Tc and Tp.

The srTCM limits the rate of incoming traffic based on the burst length, while the trTCM limits the rate of incoming traffic based on the peak rate as well as the burst length.

The metering and marking alternatives considered are: single rate Three Color Marker (srTCM), two rate Three Color Marker (trTCM) and token bucket Three Color Marker (tbTCM)

A. Algorithm of srTCM Scheme

stTCM algorithm marks the incoming packets green, yellow, or red according to the metering result as shown in Fig.2.

The single srTCM can be used as component in a DiffServ traffic conditioner. Basically, it Meters a traffic stream and mark its packets according to three traffic parameters, CIR (Committed Information Rate), CBS (Committed Burst Size), and EBS (Excess Burst Size), to be either green, yellow, or red.

A packet is marked green if it does not exceed CBS, yellow if it does exceed the CBS but not the EBS, and red otherwise, The packet colour can be used by a congestion control to implement discarding policies, Also, a combined set of values for the three traffic parameters can be used to control the bandwidth usage for each data packet streams.

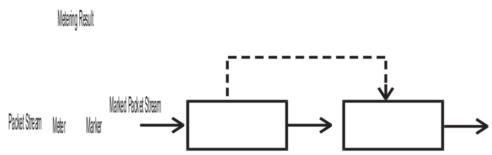

Fig. 2. srTCM: Traffic Conditioner

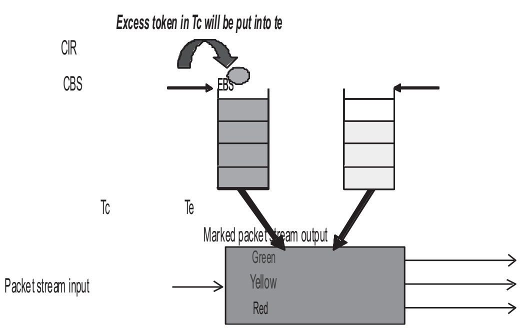

Fig 3 : Analytical model of srTCM algorithm

Fig.3 represents the srTCM as two buckets on line filled with tokens at a constant rate. Each token allows the transmission of one previous agreed of packets. When the packet of size 'B' arrives at a time't'. Then the size of the token in bytes is calculated by the product of CIR and the time interval between the arrival of two packets (Current packets and previous packets).

If the currently filled tokens in the current bucket plus the size of this token exceed bucket size remaining bytes are transfer to the excess buckets. Else, the token is enqueued in the current bucket. Tc denotes the space occupied by the tokens in current bucket in terms of bytes. Te denotes the space occupied by the tokens in the excess bucket in terms of bytes.

When token is transferred in current bucket Tc is incremented by its size. If this value exceeds bucket size then Tc is updated to full bucket size and remaining token bytes are transferred to excess bucket.

Hence, Te gets incremented. If its value exceeds excess bucket size, Te is updated to EBS.

Now the packets are checked for their colours. If the packet is not marked or has been marked green then it is marked green depending that its size is less that the total bytes filled in current bucket. Else, if the size is larger or the packet is yellow then its size is compared with the total bytes in excess bucket. I frames size less it is marked yellow else it is marked red.

When packet is finally marked as green the packet size is deducted from current bucket. If it is marked yellow its size is deducted from excess bucket.

The behavior of the meter is specified in terms of its mode and two token buckets, Tc and Te, which both share the CIR. The maximum size of the bucket Tc is CBS and the maximum size of the bucket Te is EBS.

After the meter the marker should reflects the metering result by setting the DS field of the packet to a particular codepoint. In case of AF PHB, the color can be coded as the drop precedence of the packet [2].

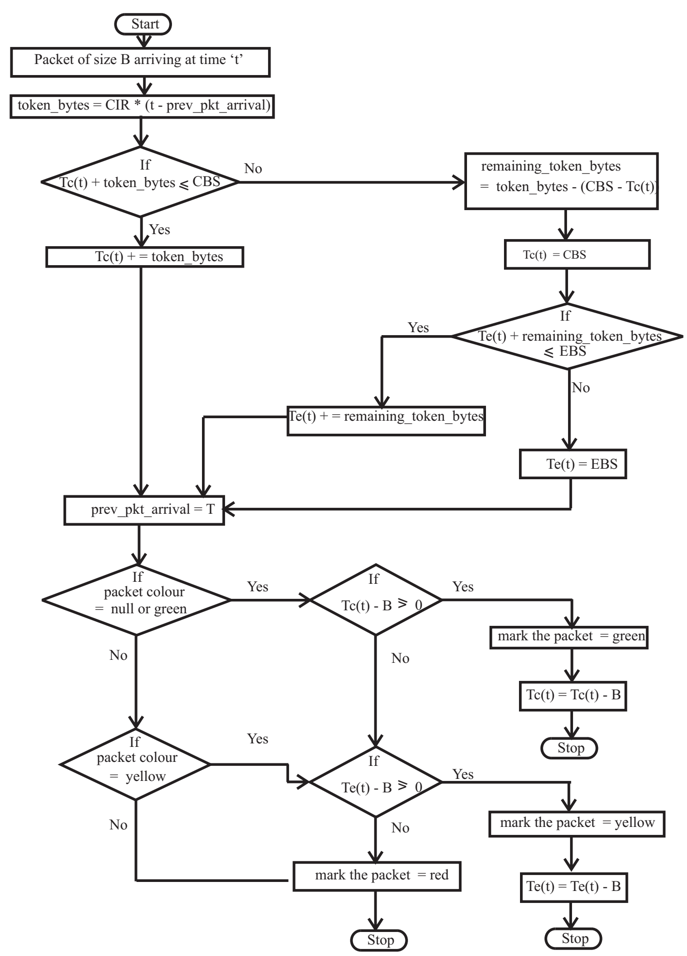

Fig 4 : The color-aware srTCM algorithm

The color-aware srTCM algorithm is shown in Fig.4. In this algorithm, B (packet size) is the size of the arrival packet, prev_pkt_arrival is the arrival time of the previous packet, Tc(t) is the token counter for bucket Tc, and Te(t) is the token counter for bucket Te.

The algorithm of the color-aware SRTCM is described as follows:
When a packet of size B arrives at time t, the following happens if the srTCM is used
if (Tc(t) - B >= 0)
the packet is Green and Tc is decremented by B
else
if (Te(t) - B >= 0)
the packets is Yellow and Te is decremented by B
else
the packet is Red and neither Tc nor Te is decremented

B. Algorithm of trTCM Scheme

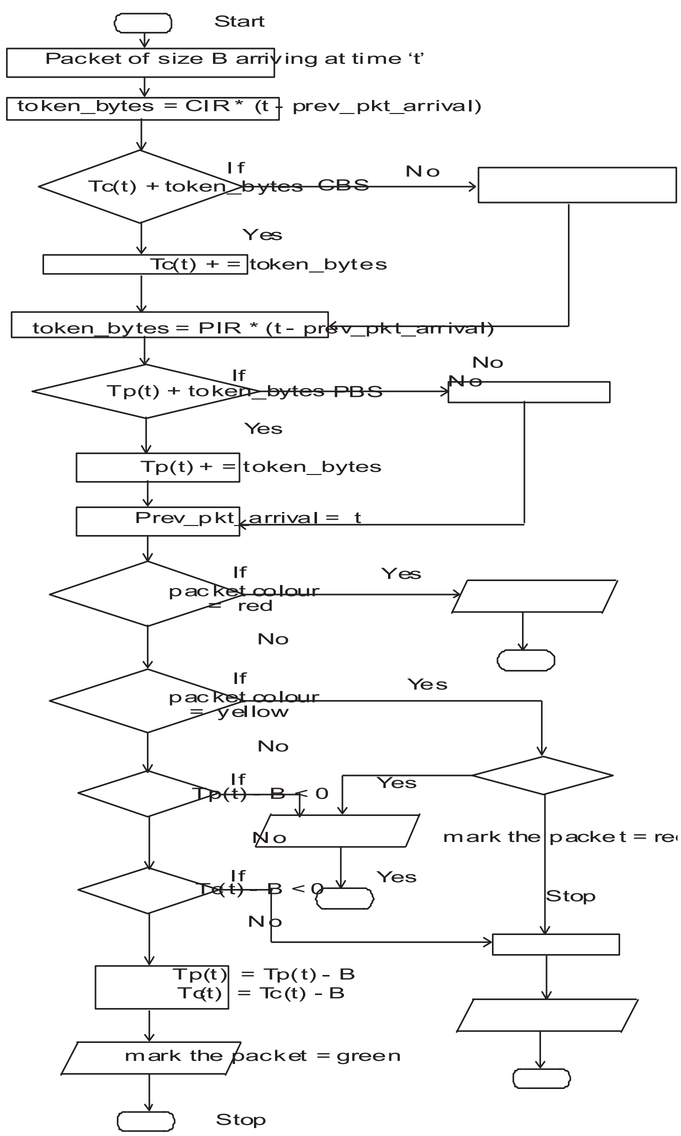

Fig 5 : The color-aware trTCM algorithm

The algorithm of the color-aware trTCM is described as follows:

When a packet of size B bytes arrives at time t, the following happens if the trTCM is used.
if (Tp(t)-B < 0)
the packet is red
else
if (Tc(t)-B < 0)
the packet is yellow and Tp is decremented by B
else
the packet is green and both Tp and Tc are decremented by B

C. Proposed Algorithm

The fundamental principle of proposed improved token bucket three color marker (ItbTCMproposed algorithm is that all frame packets are marked as green provided that sufficient tokens are available in the bucket. When there are enough tokens, packets are marked as green (lowest drop probability) and then forwarded. If there are no enough tokens, important packets, i.e., I or P frames are marked as yellow; B frame packets are marked as red (highest drop probability). So when congestion occurs, B frames are dropped first. Therefore, in the event of network congestion, less important frame packets can be dropped before more important ones. The algorithm of the color-aware ItbTCM is described as follows:

When a packet of size B bytes arrives at time t, the following happens if the trTCM is used.
if (Tp(t)-B $\geq$ 0)
the packet is green
else
if (significance bit = set)
the I or P frame packet is yellow
else
the B frame packet is red
if (the significance bit = not set)
mark the packet as red
endif

4. Simulation and Analysis

This Section describes the Simulation model, which includes the network model and the traffic model, and the experimental setting used in our simulation. To achieve improved video over IP, the proposed marker system can be deployed on edge routers and core routers along the data path. To simulate and analysis the mechanisms, a number of experiments have been performed on the basis of the network simulator2 (ns2) which has been developed at the Lawrence Berkeley National Laboratory (LBNL) of the University of California, Berkeley (UCB) [13]. The performance of video over IP DiffServ is studied with extensive simulation results and by comparing the performance of improved token bucket TCM algorithm with existing srTCM (single-rate three color marker) [2] and trTCM (two-rate three-color marker) [3] algorithms over DiffServ network. The reason of the selection of srTCM and trTCM because they are the only token bucket based three-color maker proposed by IETF.

A. IP Diffserv Network Model

The performance evaluations of video transmission over IP network, investigators uses simulation topology of IP DiffServ network model as depicted Fig.6 and implemented the simulation model in the ns-2 simulator. The performance evaluation is carried out through simulations at the network scenario depicted in IP DiffServ network models (Fig.6). The link between core router (r2) and edge router (r3) is the only bottleneck link in this topology and has a bandwidth of 1mbps with 10ms propagation delay. Other links are all l0mbps with lms propagation delay. It consists of a MPEG-4 video server, a FTP& on-off background traffic flow server and a client which are connected respectively with the IP core network by edge router. The FTP & on-off background traffic flow servers generate best-effort (BE) background traffic. The traffic model used in the simulation is the ON/OFF model with exponentially distributed ON and OFF times. During each ON period, an exponentially distributed random number of packets with average burst length (BL), are generated at fixed rate "p" packet/sec. Investigators used the foreman QCIF movie sequences, encoded with MPEG-4 encoding. Each frame is fragmented into packets of 1024 octets. The video packets are transmitted from the video server S1 to Video client D1.The MPEG-4 video flow competed with one on-off background traffic flow, which has an exponential distribution with mean packet size of 1500 bytes, burst time of 500 ms, idle time of 0 ms, and rate of 1mbps, and one FTP traffic flow. The differentiated service code point (DSCP) value is marked in the IP header.

When packets are transmitted to the DiffServ edge router, they will be classified, conditioned and scheduled. The core router implemented a RED mechanism with three levels called as weighted random early detection (WRED) mechanism for active queue management (AQM) and forward packets to next core or edge router. The threshold parameters of the WRED mechanism for the three levels are configured as percentages of the total queue length (qlen), which is kept equivalent to 50 packets. Investigators used the WRED parameters standardized by the previous study for enhanced delivery of video over IP network [1]. The WRED parameters used in the simulations are (0.6*qlen, 0.8qlen, 0.025), (0.4*qlen, 0.6qlen, 0.05), (0.2*qlen, 0.4qlen, 0.1) for green, yellow, and red packets respectively. This set discriminates red packets, progressively avoiding the discard of yellow and green packets. The packets of the TCP flows are considered as red packets. Finally packets are routed to edge router 2 (r2) and then to the client. The client will decode each frame.

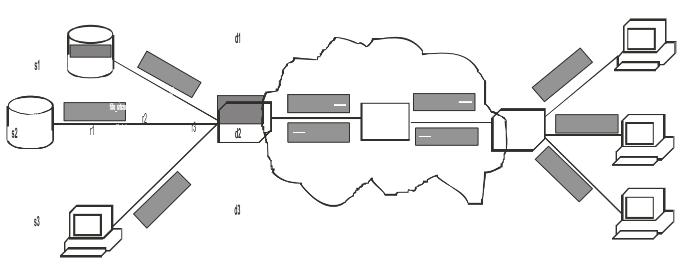

Fig 6 : Simulation topology of IP DiffServ network model

A single video server generates packets based on a public available MPEG-4 frame trace file to a remote video client. There are 3 servers (s1-s3), 3 clients (d1-d3) and three routers (2 edge routers and 1core router) in the network. Traffic from the server will send to the client with corresponding number, for example, from s1 to d1, s2 to d2, etc. Each source has a responsive aggregate separately. Investigators used the "Foreman" movie sequence, encoded with high quality MPEG-4 encoding, which has a mean bit rate of 0.77 mbps and a peak rate of 3.3 mbps. Each frame was fragmented into packets. To evaluate the performance of the Video over DiffServ network researcher used MPEG-4 video traffic traces to compare the performance of the proposed mechanism with those of legacy packet markers using the NS2 simulator. The test video trace was a Foreman QCIF format sequence, which has 400 frames. The encoded stream has 30 frames, mean bit rate of 257 kbps. Each frame was fragmented into packets of 1024 bytes before transmission. The srTCM, trTCM, and ItbTCM (Improved Token Bucket TCM), schemes were all implemented on the ingress router for performance comparison purposes. The WRED parameters at core router include a minimum threshold, a maximum threshold, and a maximum drop probability. In the current simulations, these parameters were specified respectively as 30, 40, and 0.025 for green packets, 20, 30, 0.05 for yellow packets, and 10, 20, 0.10 for red packets. During the simulation, the significance bit of I frame and P frame packets was set at the video server prior to transmission.

5. Results and Discussion

Results include a quantitative study on how much a DiffServ enabled network may improve the delivery quality of MPEG video streams over IP network. Comparing with the quality of received video of proposed ItbTCM marker with srTCM, and trTCM. Simulation results show that proposed scheme performs better than the other three schemes. Numerical experiments on video encoding, transmission and decoding have been done using foreman in YUV format of QCIF format video. The Three Colour Markers (TCM) viz., srTCM, trTCM, and ItbTCM have been used under DiffServ. Investigators compared the performance of the packet loss rate of reconstructed picture. Packet loss rate compares the video quality over IP network of various token bucket based algorithms. From Fig. 7 the frame packets I, P, and B for various markers are which video server is transmitting are respectively 237, 149, and 273. By implementing srTCM at edge router with no frame packet marking, the frame packets I, P, and B received at the video client are 191, 138, and 223 respectively. While implementing srTCM with marking of frame packets the packets I, P, and B received at the video client are 181, 132, and 184 respectively. The packet loss rate of I frame is 19.40 percent in srTCM without mark and 23.62 percent in srTCM with marking of frame packets. In P frame packets the packet loss rate is 7.38 and 11.40 percent in srTCM without any mark and with packet farmes marking respectively. Same trend is seen in B frame packets. 18.31 percent in srTCM with any marking and 32.60 in srTCM with marking. Therefore, the layered approach (base and enhancement layer) i.e., srTCM packet marking algorithms increases the I and P frame packets losses which are 4.22 and 4.02 percent respectively and hence the performance of received video quality is degraded.

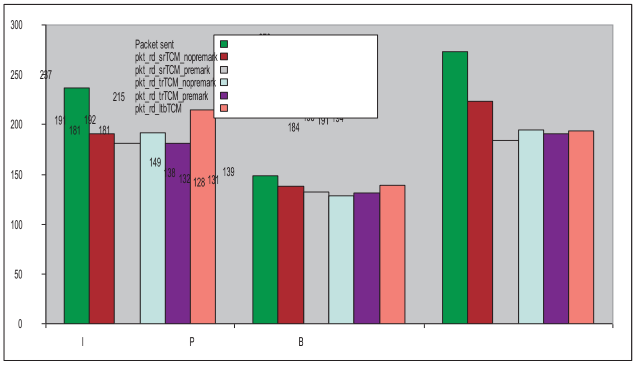

Fig 7 : PLR between various token bucket based TCM

By implementing trTCM_nopremark at edge router, the frame packets I, P, and B received at the video client are 192, 128, and 195 respectively and while implementing trTCM_premark of frame packets the packets I, P, and B received at the video client are 181, 131, and 191 respectively. The packet loss rate of I frame is 18.98 percent in trTCM_nopremark and 23.62 percent in trTCM_premark. In P frame packets the packet loss rate is 14.09 and 12.08 percent in trTCM_nopremark and trTCM_premark. Same trend is seen in B frame packets. 28.57 percent in trTCM with any marking and 30.03 in trTCM with marking. trTCM_premarking increases the I frame packet loss by 4.64 percent and decreases P frame packets losses by 2.01. The P frame packet is more important than P and B frames therefore this scheme also degraded the performance of received video quality. By implementing ItbTCM, the frame packets I, P, and B received at the video client are 215, 139, and 194 respectively. The packet loss rate is 09.28 percent (I frames), 06.71 percent (P frames), 21.93 percent (B frames) using ItbTCM with DiffServ. Therefore the performance of ItbTCM is clearly superior and enhances the received video quality over IP network.

6. Conclusions

Packet loss is used as a parameter to determine the quality of video during the video communication. There is packet loss due to congestion in the network. This is mainly due to the fact that, if a packet arrives at the router and the router queues are full, then the packet is dropped. Simulations are run and this performance metrics are evaluated to find the best algorithm to minimize the total packet loss. Researchers have implemented the MPEG-4 video transmission in a DiffServ network to compare the performance of the proposed scheme (ItbTCM) with other existing algorithms such as srTCM and trTCM. In this scheme, I frames, P frames and B frames are marked as different service priorities according to transmission condition and their contribution to the perceived picture quality. Results show that proposed scheme improves the sizable improvement in terms of reducing the total packet loss during the video communication over IP network.

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