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CHOKeR: Active Queue Management for Multimedia Applications

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Abstract: For multimedia applications processing in MANETs, even the small packet loss can cause very much degradation in quality. Hence there is a need to design an efficient Active Queue Management to reduce the packet loss. This paper proposed a CHOKeR-M algorithm to reduce the packet loss in MANETs. This AQM considers the feature information of videos and network environment to increase the Quality of Service (QoS). The complete packets were divided into five prior classes based on the information carrying by them. For an arrival of a packet with particular priority, the router will check the priority and network congestion level. The complete buffer occupancy was compared at four levels. At each and every level, the router will decide the number of packets to be buffered and also to be dropped through a parameter called drawing factor. The simulation analysis was carried out over various videos. For each and every video, the subjective measurements were carried to show the effectiveness of proposed approach. Packet Delivery Ratio, End-to-End delay, Average Throughput and Peak Signal to Noise Ratio (PSNR) were evaluated at each and every stage and compared with earlier CHOKeW, RED, BLUE and RIO approaches.

Keywords: MANETs, AQM, MPEG, Drawing Factor, PSNR, PDR, Throughput.

I. INTRODUCTION

In today's world, with wide enhancement in the applications of wireless devices and multimedia technologies, Quality of Service (QoS) has become a major issue for multimedia services in MANETs. Hence, it is expected to provide QoS for various multimedia applications in MANETs. Due to the dynamic nature of MANETs, the QoS provision was very complex issue. QoS related issues have been studied from a number of years, but have not been solved completely. One of the main challenges of MANETs is to develop a cost effective and reliable approach that can provide better QoS for a network, supporting thousands of flows of multiple services at different priority levels. Diffserv architecture was one of the multiservice architecture which can support multiple and different flows in a single system. Based on the per-hop behavior (PHB) [1] of IP packets, it can support various types of services. Diffserv architecture has obtained more research interest due to its simplicity and scalability. In Diffserv architecture, Active Queue Management (AQM) approaches are the most efficient network control methodologies that can provide an adequate tradeoff between throughput and drop probability[2],[3]. Various AQM approaches lead to various QoS performance. The main objective of any AQM in Diffserv architecture is to provide an efficient QoS by reducing packet loss. However, the existing AQM algorithm's QoS provision is in the manner, such that it is not able to reach the requirements of real time video flow. Because, even a small packet loss will have a severe impact on the quality of video.

In the view of QoS, protection of data from packet loss is a widely discussed research topic [4]. But, it is very complex to control the packet loss to a minimum level using current packet dropping approaches. Furthermore, even a small packet loss may contribute to severed degradation in video quality [5].A main challenge for video stream delivery quality in MANETs is how to maintain the perceived video quality through the control of packet loss. In earlier, various AQM approaches were proposed which can provide better QoS by reducing packet loss. But, in such type of multimedia applications which were encoded by MPEG4, there is no a direct and simple relation between the video quality and packet loss. In MPEG encoded videos, there exists interdependency between the video frames. So, even if there is a small packet loss may cause many successive delivered packets with no use. This will give a serious impact to the video quality. Because the resource in MANETs is limited and varying, while large bandwidth is needed by video application, it is unavoidable that lots of video packets have to drop. To reduce the loss of video stream delivery quality and effectively drop video packets, this paper will introduce autonomic mechanism to queue management, dynamically adjusting the packet dropping operation according to the network congestion and video feature information.

In this paper, a CHOKeR-M algorithm mechanism was proposed by attributing autonomic features for AQM to reduce the packet loss in multimedia application. The proposed approach reduces the packet loss by considering the priority coding. The packets carrying low priority information will be dropped and the packets with high priority will be allowed. For priority coding, the proposed approach divided the total packets into five classes based on the information carrying by them. For a packet arrival with particular priority, the router will check the congestion statue of buffer and priority of packet. Depends on the congestion level, the router will buffer incoming packet into buffer or it draws packets from buffer to drop. The number of packets to drop depends on the drawing factor. Simulation was carried out through both visually and analytically. The rest of the paper is organized as follows: section II gives the details about the related work. Section III provides the details about the proposed AQM algorithm. Section IV illustrates the performance evaluation and section V concludes the paper.

II. RELATED WORK

In earlier, various approaches were proposed on Active Queue Management. Earlier AQM algorithms such as Random Early Detection (RED) [6] and BLUE [7] are not able to provide better QoS, because, they didn't considered any priority coding for multimedia data, depends upon the congestion status of buffer the route will drop the packets. In [8], an enhancement to conventional RED was proposed as RED with preferential dropping (RED-PD). RED-PD requires reserved parameters to drop the packets. This increases memory requirement. One more disadvantage, all these algorithms drop the packets without considering any priority. There is no vast research on AQM exploiting it towards the multimedia flow delivery quality. Most of the earlier proposed AQMs drop the packets randomly without considering the importance of packets. For multimedia data, there are two types of frames, intra coded frames and inter coded frames. There exists a relation between these frames. If any one of the frames was dropped unnecessarily, it will affect the entire video quality. RIO [13], Weighted RED (WRED) [14], Autonomic Active Queue Management (AAQM) [16] and WRED with threshold (WRT) [15] are some of the earlier proposed AQM approaches which consider the differentiated packet priorities. But the priority was not given in the view of packet importance. In [10], a rate based RIO, (RB-RIO) was proposed, classifies the packets into I frame, P frame and B frameto three priority classes. According to RB-RIO, compared with other packets, the packets carrying I frame data has low drop probability.

In [11], an enhanced WRED, REDN3 was proposed which mapped I frame, P frame and B frame packets of MPEG4 video flow system with different priorities of WRED. Some scheduling schemes, such as weighted fair queuing (WFQ) may also support differentiated bandwidth allocation [19], [20]. The main drawback of these scheduling schemes, such as WFQ, is that they require constant per-flow state maintenance, which is not scalable in core networks. An autonomic active queue management (AAQM) was proposed in [16] was also considered the packet priorities through the interdependency relation of video frames in a MPEG4 encoded video. However, the queuing mechanism proposed in this approach considered only two levels for the evaluation of congestion status of buffer. If the max buffer limit was exceeded by maximum threshold, it evaluates the packet dropping probability and simply drops the all entire packets with high probability, but didn't discuss about the number of packets to be dropped at each and every stage. This will increase the packet loss and then proceeds to degrade the Quality of video at receiver.

III. CHOKeR-M

The proposed approach applies the priority based dropping criterion for congestion control and also to provide better QoS for Multimedia applications. For a given video input, the proposed approach provides better Quality of Service by dropping packets based on their priority. This approach also considers the network environment. Network environment will change dynamically. The length of node's buffer will also change according to traffic. The network router will collect this network context information to judge the network congestion condition. Network context information is monitored by wireless nodes and the service context information is recorded in the packet header according to their video compression character by source end. When the packet travels in network, nodes are able to abstract the service context information directly from packet header. The service context information, for example video compress character information, such as frame type, frame situation and frame size are abstracted from the IP packet header, and it will determine the configuration of the queue management mechanism parameters. During the packet processing, the network context information and service context information were collected dynamically and are used to judge the network condition and configuration. This adapts self-configuration and self-adaption to the network. To obtain eservice context information, the proposed approach uses complete packet feature information of video encode process to optimize the queue management.

A. Priority Indexing

In MPEG4 encoded video stream, each and every frame in Group of Pictures (GOP) encoded with different methods have different influence on the video at the time of reconstruction at the receiver to maintain video quality. The complete video was encoded as three frames by MPEG4 encoder. They are I frame, P frame and B frame. I frame is an independent frame. Because it was encoded in an independent manner or intra frame manner, so it can be reconstructed independently after decoding. Whereas, P and B frames are encoded in dependent manner or inter frame manner, so they require relative frames for reconstruction after decoding. We can obtain the frame importance from the Group of Pictures. Due to the independent nature of I frame, it can be considered as a reference frame and can give high importance. In P frame, the front part will have more importance compared to rear part in each GOP. Finally, the B frame was encoded through prediction encoding, its amount of data almost in similar with prediction frame, hence the B frame with larger size have high importance compared to B frame with smaller size.

Finally, the order of importance from high to low of frames in each GOP can be summarized as, I frame, P frame with front part, P frame with rear part, B frame with larger size and b frame with smaller size. The service context information includes the video frame type, frame size, frame number and so on. This was generated by video compression. All these character information generated by video codec will be inserted into the packet header. Along with these character information, several fields such as Number of P frame in GOP, maximum P frame number, B frame size, maximum B frame size and frame type, named as f_seq, PN, f_size, BS and f_type respectively. For te filed f_type, 0 indicates I frame, 1 indicates P frame and 2 indicates B frame. The number of P frame is the P frame predicted sequence number in GOP, which is between 1 and PN. And the f_size field records the B framesize. The complete packets are also categorized into five categories, PCI (Packet Carrying I frame data in GOP), PCFP(packet carrying the former part P frame data in GOP), PCLP (packet carrying the latter part P frame data in GOP), PCLB (packet carrying the latter part P frame data in GOP), and PCSB (packet carrying smaller size B frame data). The source end divides the video packets into five priorities according to the video compression character information. Additionally we add another filed into IP header, named PI(k). The priority dividing table is shown in table I. The packet importance decreases from class1 to class5.

TABLE I: Packer Priority Indexing

Class	Packet Type	Priority Index (k)
Class 1	PCI	001
Class 2	PCFP	010
Class 3	PCLP	011
Class 4	PCLB	100
Class 5	PCSB	101

B. Queue Management Operation

The proposed active queue management mechanism defined a new parameter called drawing factor (d_0) to define the number of packet needs to be dropped. The drawing factor will update automatically depends on the congestion level of queue. In this approach, the drawing factor will increase in multistep and will decrease in single step. The collecting operation of proposed approach monitors the queue length to estimate the network congestion situation. Here, the complete queue length was divided into five thresholds ,T_{th}, T_l, T_h and T_{lim}. The complete queue management operation was shown in fig.1. Parameters in the RED and WRED queue management algorithms are static. The parameter Np is set as a fixed value. But in our proposed approach, the Np is able to adjust according to the packet's service context information. The original Np for I, P and B frame areNp_i, Np_p and Np_b respectively. But the $Np_p$is able to be adjusted according to the P frame number. The Np is adjusted according to P number as

$$Np = Np_i + (Np_p - Np_i) * a * (No./N) \qquad (1)$$

Where a is a small constant. And the Np_b is adjusted according to the B frame size, as

$$Np = Np_i + (Np_p - Np_b) * b * ps/size \qquad (2)$$

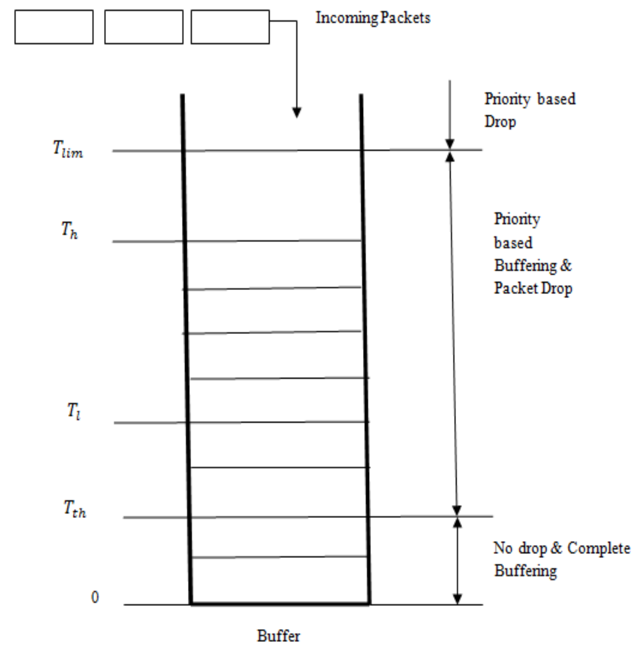

Fig.1.Queue Management.

In which b is a small constant. When a packet with priority k arrives, the router checks the buffer occupancy c_k with the set of predefined thresholds. If the buffer occupancy was less than the minimum threshold T_{th}, then there should be no drop and the drawing factor will be set to zero. When the buffer occupancy c_k is greater than the minimum threshold T_{th} and less than the lower threshold T_l, the drawing factor will be decreased by a single step d^-.

$$d_0 = |d_0 - d^-|^+ \qquad (3)$$

Where $|x|^+ = max(0, x)$. If the buffer occupancy is in between lower and higher thresholds, $T_l < c_k < T_h$, the drawing factor d_0 was kept unchanged. When the buffer occupancy is above higher threshold, the router will increase the drawing factor in multiple steps aggressively with

$$d_0 = d_0 + w * d^+ \qquad (4)$$

Where, the number of steps calculated as

$$w = \frac{[C_k - T_h]}{[T_h - T_l]} \qquad (5)$$

When the buffer occupancy is above the threshold limit T_{lim}, the complete incoming packets were buffered into five virtual queues and the packet with low priority will be dropped. The variation of drawing factor for varying traffic was shown in fig.2. The complete idea of proposed approach was depends on the drawing factor that can be used as signal to inform the congestion status. Fig.2 illustrates the variation of drawing factor with traffic flow. The graph shown in fig.2 gives the relationship between traffic flow and drawing factor. As the traffic increases, the number of packets to be dropped also increases, thus the variations of drawing factor will increases exponentially and for maximum flow, the drawing factor will reaches a high value to block the complete video flow.

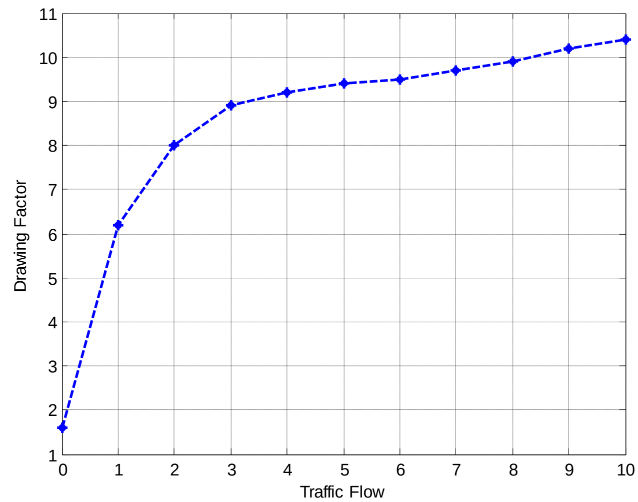

Fig.2. Drawing factor variations for varying traffic flow.

The complete details of the proposed approach were shown as,

Algorithm

Initialization $d_o = 0$; // *drawing factor*
For each incoming packet with priority k
If (k=001) //PCI[1]
$Np = Np_i$ // *I-frame data*
Else if (k=010||011) //PCP[2]
$Np = Np_i + (Np_p - Np_i) * a * (No./N)$
Else if (k=100||101) //PCB[3]
$Np = Np_p + (Np_b - Np_p) * b * ps/size$
// *calculate no. of packets to be dropped for each and every incoming packet through drawing factor*//
if $C_k < T_{th}$
$d_o = 0$; // *No drop*
$C_k = Np$
Else if $T_{th} < C_k < T_l$ *Then*
$d_o = |d_o - d^-|^+$
Else if $T_l < C_k < T_h$ *then*
$d_o = d_o$ || *No change in drawing factor*
Else if $T_h < C_k < T_{lim}$ *then*
$$w = \frac{[C_k - T_h]}{[T_h - T_l]}$$
$d_o = d_o + w * d^+$
Else if $C_k \geq T_{lim}$
// *create five virtual queues and drop the packet with lower priority index*
drop Np

PCI[1] packet carrying I frame data
PCP[2] packet carrying P frame data
PCB[3] Packet carrying B frame data

IV. SIMULATION RESULTS

The simulation analysis of the proposed approach was shown in this section. For simulation, a randomly moving 30 nodes were considered in the area of 500X500 m^2. The proposed approach was applied on the created network by selecting one source node and one destination node. Initially the proposed approach is applied on the Constant bit rate (CBR) information. The buffer limit is 500 packets, and the average packet size is1000 bytes. TCP flows are driven by FTP applications, and UDP flows are driven by constant bit rate (CBR) traffic at the speed 10 Mb/s. All TCPs are TCP SACK. Each simulation runs for 500s. We use another parameter W to denote w eight for priority in simulations. The weights were varied as 1.0, 1.5, 2.0 and 2.5. Based on this weights four priority levels are assigned for information and then processed in the queue.

Case 1: The following table gives the details about the simulation parameters for case 1. Table.2 represents the details about the simulation parameters used.

TABLE II: Simulation Parameters

Parameter	Value
Number of nodes	30
Bandwidth	11Mb
Area	500mX500m
Radio transmission range	300m
Routing protocol	DSDV
QoS Metrics	Avg TCP Goodput, queue length, Frame index
weights	1.0, 1.5, 2.0 and 2.5
Data	CBR

The obtained Average TCP good put in Mb/s for three priority levels and four priority levels are shown below.

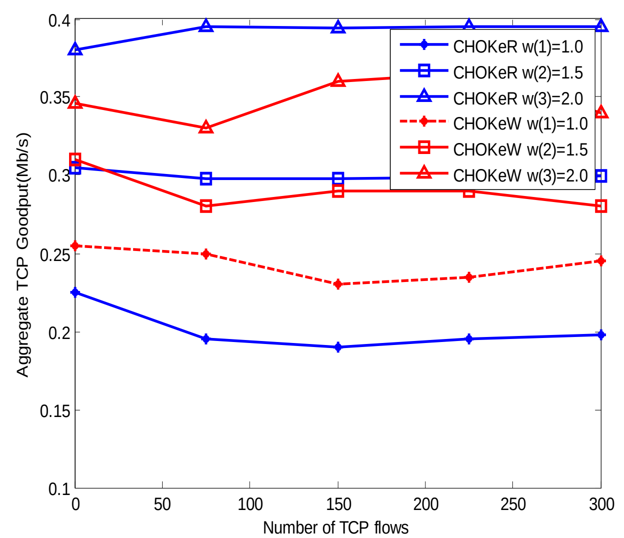

Fig.3. Aggregate good put versus the number of TCP flows under a scenario of three priority levels.

Figs.3 and 4 demonstrate the aggregate TCP good put for each priority level versus the number of TCP flows for three priority levels and four priority levels of CHOKeW and CHOKeR, respectively. In Fig.3, the number of TCP flows ranges from 30 to 300 at each level. When the number of TCP flows changes, the ratios of allocated bandwidth for each priority level in CHOKeR are still stable, while CHOKeW cannot support the bandwidth allocation. Moreover, as the number of priorities increases, CHOKeR is still stable, while CHOKeW becomes worse in terms of bandwidth allocation.

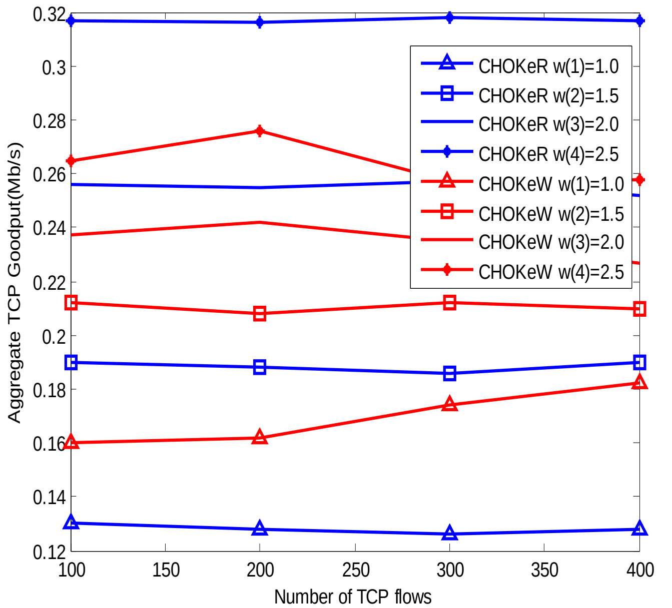

Fig.4. Aggregate good put versus the number of TCP flows under a scenario of four priority levels.

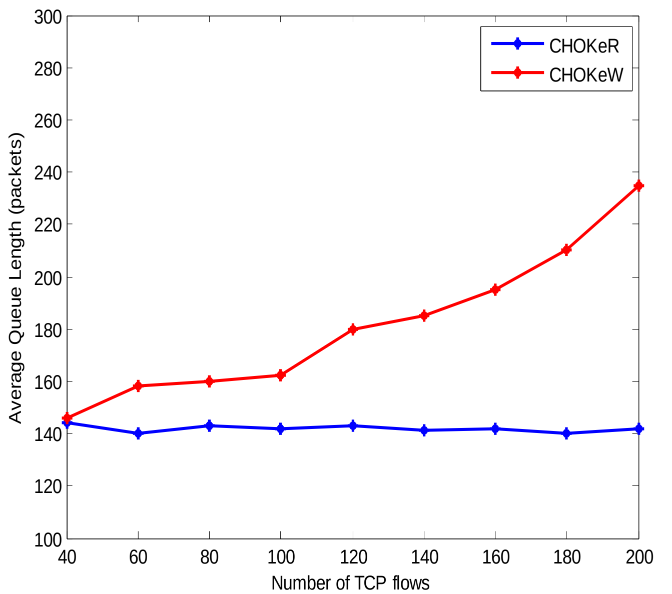

Fig.5. Average queue length of CHOKeR and CHOKeW.

In order to compare the average queue length of CHOKeR and CHOKeW, all flows in the simulations are assigned the same priority. In Fig.5, we can see that the average queue length for CHOKeW increases as the number of TCP flows increases. In contrast, the average queue length of CHOKeR can be maintained at a steady range. That is to say, when the number of TCP flows increases, the queue length of the router does not increase. The reason that CHOKeR can maintain a steady average queue is that it uses a MISD drawing factor.

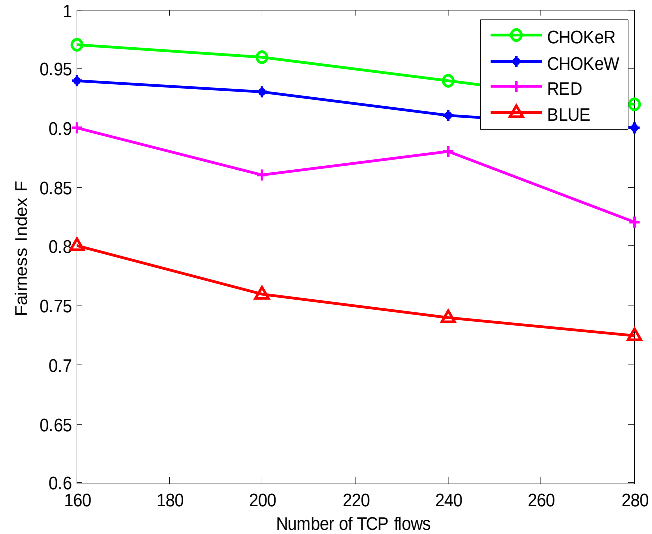

Fig.6. Fairness index versus the number of flows for CHOKeR, RED, andBLUE.

Fig.6 shows the fairness index of CHOKeW, CHOKeR, RED, and BLUE versus the number of TCP flows ranging from 160 to 280. It can be seen that while the fairness indexes of RED and BLUE decease as the number of flows increases, CHOKeW and CHOKeR provide much better fairness. The second match drop action i.e., the priority match drop in Algorithm, enables proportional bandwidth allocation in CHOKeR, which is a significant improvement over CHOKeW.

Case 2: Then the proposed CHOKeR is applied for multimedia information, where as in multimedia data, the frames having individual priority, thus there is a need to allocate bandwidth by considering that priority also. The following table gives the details about the simulation parameters for case 2. Table.3 represents the details about the simulation parameters used.

TABLE III: Simulation Parameters

Parameter	Value
Number of nodes	30
Bandwidth	11Mb
Area	500mX500m
Radio transmission range	300m
Routing protocol	DSDV
Input Video	.AVI format
QoS Metrics	Avg TCP Goodput, queue length, PSNR
weights	1.0, 1.5, 2.0 and 2.5
Data	Videos

The videos are encoded with MPEG formatted file and transmitted through the above created network. Finally, the received video was compared with original video and evaluates the PSNR (Peak Signal to Noise Ratio). PSNR is one of the most widespread objective metrics to assess the application-level QoS of video transmissions. The wireless bandwidth is set as 11Mb. Node communication radius is set as 300m. Routing protocol uses DSDV. The length of interface queue is set 100.The QoS performance of the CHOKeR-M was verified through the evaluation of QoS metrics such as Packet Delivery Ratio, End-to-End delay and Average TCP good put. Fig7 represents the video file which was used to evaluate the proposed approach. The video file has been transmitted from source node to the destination node. Here the video file is of MPEG4 formatted file and transmitted through the wireless mobile ad hoc network, MPEG encodes video as a sequence of frames. The video sequence may be decomposed into smaller units which are coded together. Such units are called GOPs (Group of pictures). Each GOP holds a set of frames or pictures that are in a continuous display order.

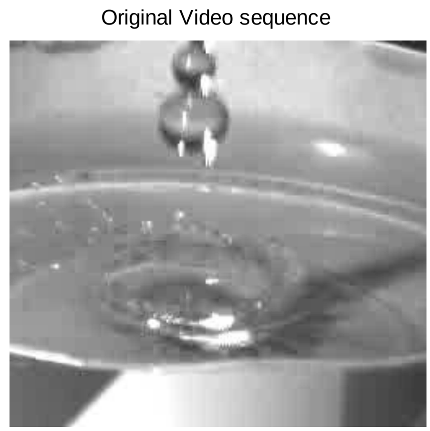

Fig.7. Original video sequence.

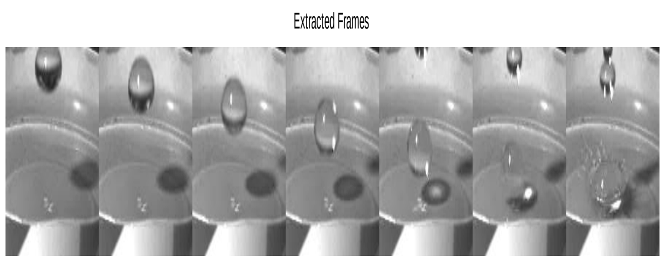

Fig.8. Extracted frames.

The GOP pattern specifies the number and temporal order of P and B frames between two successive I frames. Standard MPEG encoders generate three types of compressed frames (I, P, & B). The decomposed frames of original video sequence were shown in Fig.8. Fig.9 represents the recovered video sequences at destination node using proposed CHOKeR. The proposed approach gives frame priority along with TCP protection such that the video quality obtained through proposed CHOKeR is better compared with earlier CHOKeW and RED. From the above figure, it is clear that the video quality of proposed approach is better. Figs.10 and 11 demonstrate the aggregate TCP goodput for the given video input for each priority level versus the number of TCP flows for three priority levels and four priority levels of CHOKeW and CHOKeR, respectively. In Fig.10, the number of TCP flows ranges from 3000 to 30000 at each level. When the number of TCP flows changes, the ratios of allocated bandwidth for each priority level in CHOKeR are still stable, while CHOKeW cannot support the bandwidth allocation. Moreover, as the number of priorities increases, CHOKeR is still stable, while CHOKeW becomes worse in terms of bandwidth allocation.

Fig.9. Recovered Video Sequences.

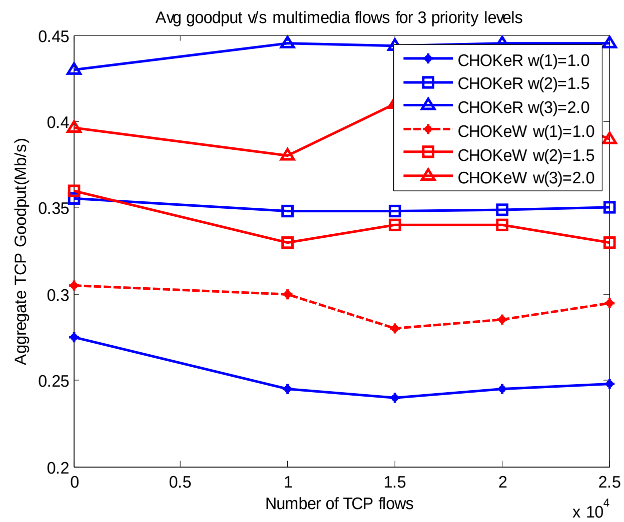

Fig.10. Aggregate goodput v/s the number of TCPflows for video under a scenario ofthree priority levels.

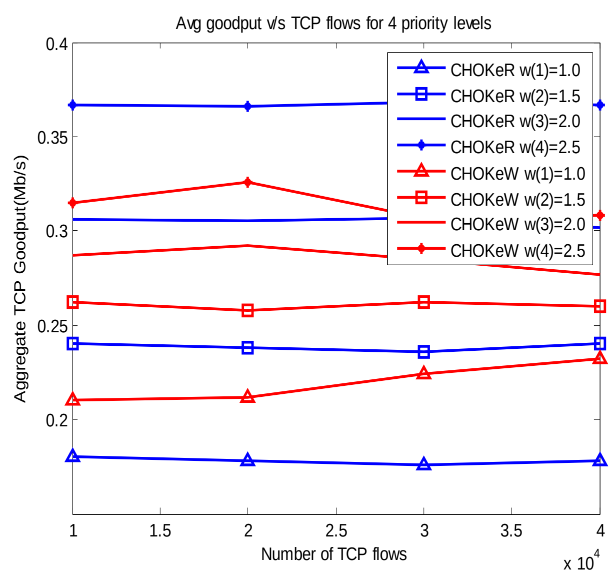

Fig.11. Aggregate goodput v/s the number of TCPflows for video under a scenario of four priority levels.

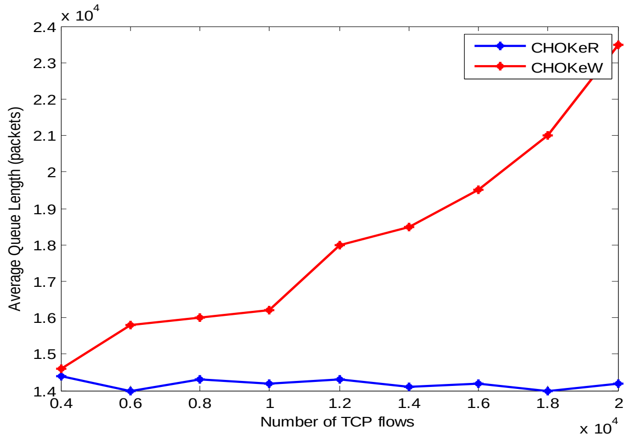

Fig.12. Average queue length of CHOKeR and CHOKeW.

In Fig. 12, we can see that the average queue length for CHOKeW increases as the number of TCP flows increases. In contrast, the average queue length of CHOKeR can be maintained at a steady range. That is to say, when the number of TCP flows increases, the queue length of the router does not increase. The reason that CHOKeR can maintain a steady average queue is that it uses a MISD drawing factor. Along with this performance evaluation, to show the enhancement of proposed approach the QoS of video is observed by measuring the PSNR. Fig.7 to Fig.9illustrates the visual analysis by displaying the given video input and the obtained video output. After receiving the video at destination node, it was compared with original video file and PSNR was evaluated using the following mathematical formula.

$$PSNR = 10.log\,(255^2/MSE) \qquad (6)$$

Where

$$MSE = \frac{1}{mn}\sum_{i=1}^{n}(\hat{y}_i - y_i)^2 \qquad (7)$$

Where
y_i=Original video file
$\hat{y}_i$=recovered video file

To show the enhancement of proposed approach, a subjective analysis was carried out by evaluating a numerical metric called peak signal to noise ratio (PSNR) for varying frame index and the respective plot was shown in fig.13.

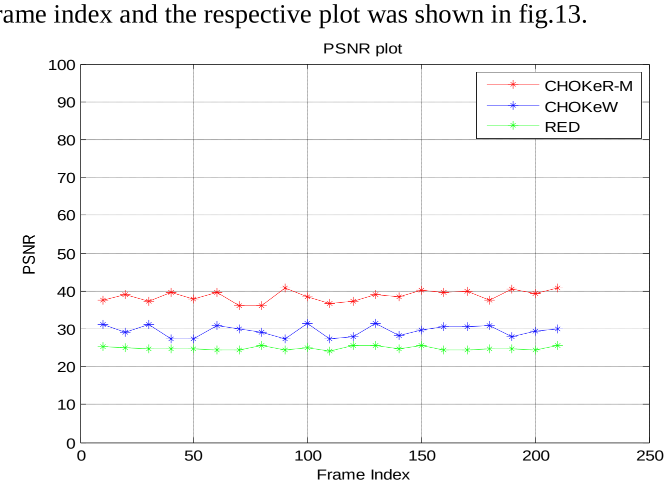

Fig.13. Peak Signal to Noise Ratio (PSNR).

TABLE IV: PSNR Comparison

Approach	PSNR
CHOKeR	40.2864
CHOKeW	30.5648
RED	26.2486

Fig.13 illustrates the PSNR details of proposed and earlier approaches. The PSNR is one of the subjective assessments metric to illustrate the QoS of video. More the PSNR, more the QoS. The PSNR gives the information about how much percentage of original video received at the receiver. As the PSNR value high, more the quality of the received video. From the above figure, as the frame index increases, the obtained PSNR for the proposed approach was better compared with CHOKeW and RED.

V. CONCLUSION

This paper proposed a new autonomic active queue management algorithm to reduce packet loss for multimedia application in MANETs. Even a small packet loss of multimedia information will degrade the quality very severely. Thus, this approach was designed very carefully to reduce the packet loss by considering multimedia feature information and the network environment. The network environment was defined through buffer occupancy and the multimedia feature information was defined through the frames of it. Both subjective and objective analysis was carried out on the proposed approach to show the effectiveness. From the simulation results, it is clear that for the proposed approach, the average TCP throughput and the PSNR are optimal, thus the proposed approach was efficient.

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