A Feedback and Active Congestion Control Mechanism for Multimedia Transmission in MANETs

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Abstract : Multimedia transmission over MANETs with high Quality of Service is a challenging task mainly due to the mobility of the nodes. Multimedia applications should deliver the packets in sequence and order, and delay should be within the tolerable limits. State-of-the-art works mainly use packet loss information at source station to determine whether the transmission window size is increased/decreased based on the reception/non-reception of acknowledgement. This leads to poor utilization of resources. In this paper, we present a feedback and active congestion control protocol to control the packet loss. We model the active stations as a Poisson random process to compute the packet loss, idle period and busy period. The packet loss estimation is used for the precise specification of buffer size resulting in improved Quality of Service. The proposed congestion control mechanism produces high packet delivery ratio and improved link utilization.

IndexTerms - MANET, congestion control protocol, microcontroller, logic circuit, Bandwidth

I. INTRODUCTION

Multimedia uses multiple forms of information content (such as animation, audio, video, text, and data) and finds applications in various domains, spanning from entertainment to education. There is a continued increase in the demand for multimedia content, and it has become an unavoidable part of the presentation of information. Multimedia streaming over the internet is well-established service that resulted in successful applications such as video conferencing and surveillance system. Extending multimedia application on Mobile Ad-hoc Networks (MANETs) is the need of the hour to utilize the full potential of these networks as well as realize the benefits of multimedia. The problems associated with multimedia transmission are a loss of packets/frames and delays in receiving the packets at the destination. This necessitates estimating the loss apriority as per the actual scenario of the link, rather than predefining it. Also, a control system is needed to reduce the packet loss and thereby increases the utilization of the link.

MANETs introduce additional challenges for maintaining better Quality of Service (QoS) throughout a multimedia session due to less resource availability such as bandwidth, storage capacity, and energy at each node. To provide an acceptable QoS in multimedia applications, the delay is the prime factor which needs to be controlled. That is, there should be a continuous flow of data packets from source to destination, and delay, if any, should be averaged out among all the communication connections. This is achievable by controlling the flow of packets from each source in a fair manner, and hence each connection shall experience more or less equal transmission delay. On the other hand, multimedia applications are loss-tolerant. So, if some information in a packet gets corrupted, that packet can be still accepted. However, one has to make sure that packet as a whole should not be lost frequently. The loss of a packet in MANETS is due to (i) Transmission errors, (ii) Route failure and (iii) Congestion at routers. The delay is mainly due to congestion within the network.

Packets loss may occur if the sequence number of a packet gets corrupted due to transmission errors. This can be overcome to a large extent by using Forward Error Correction (FEC) schemes to guard the packet sequence number. Route failure due to node mobility can be controlled by either planning in advance the handoff or preplanning a parallel route to which a source-destination pair can shift. There are several works such as TCP Reno, Vegas, and TCP Illinois in the literature for controlling congestion, packet loss and delay on the internet. In these works, source station uses packet loss information to determine whether the transmission window size should be increased or decreased. In this way, data flow rate is controlled over the internet and subsequently, congestion is kept in check. If packet loss is within the acceptable limits, then no control is required. Otherwise, packet loss control strategy is needed. In the existing approaches (ex., [1], [2], [3]), the source station takes a decision about the increment or decrement of window size based on whether the acknowledgement (ACK) is received or not. The magnitude of increment/decrement in window size is predefined but do not compute the accurate window size that should be decreased or increased depending on the congestion. This results in poor utilization of the link. Normally, the threshold for voice/video Frame Erasure Rate (FER), Voice/Video messaging FER and Streaming Audio/Video FER is less than 3 % [4].

Packet loss due to congestion at router nodes can be primarily avoided by reserving sufficient bandwidth and buffering facility, particularly at the front end router. Packet loss cannot be reduced to zero and it may still occur with non-zero probability. However, new approaches can be developed to reduce the effect of these causes. Determining the packet loss precisely helps in defining the window size based on the amount of congestion during different intervals of time at the router node. Since multimedia is a loss tolerant up to a certain threshold limit, the necessary control action is needed for its successful transmission, otherwise, QoS degrades. However, before any control action, it is very necessary to estimate the packet loss at the front end of the router.

In this paper, we present a congestion control mechanism to reduce the loss by estimating the packet loss accurately during multimedia transmission. In this work, we present a control circuitry using digital logic to control the loss of packets, which not only increase the utilization of the link but also increase the throughput. Since the round trip time of packets is reduced by saving more number of packets, the control circuit helps in reducing power consumption when compared to window based techniques (e.g. TCP Reno and TCP Vegas). The proposed congestion control is two-fold: (i) uses *feedback* from the control circuit to the microcontroller to allow or disallow packet transmission from stations and (ii) *active* by early detection of possible congestion and adjust the packet transmission. We call the proposed mechanism as *Feedback and Active Congestion Control Mechanism* (FA-CCM).

The rest of the paper is organized as follows. In Section 2, we discuss the related work. In Section 3, we present an overview of the proposed work. Section 4 presents control to the packet loss and its applicability. The paper concludes with Section 5.

II. Background and Related Work

Transmission Control Protocol (TCP) is a reliable and connection-oriented end-to-end protocol. This protocol is mainly used to control the congestion of packet transmission and improve packet delivery ratio (PDR). At the application layer, this protocol divides the stream of bytes into TCP segments, and each segment limits the length to maximum segment length. TCP doubles the size of the window after every successive acknowledgement, called slow start (more specifically exponential start). If congestion occurs, it halved the size of the window. This protocol starts with an initial window size of 64KB. Congestion occurs if the linear growth of the window reaches the size of the receiver buffer. At this point, this protocol tries to avoid the congestion by reducing the size of the window to half for congestion avoidance.

The important congestion control schemes of TCP among several extensions are Regular TCP (also called **TCP *Tahoe***) [1] and ***TCP Reno*** [2]. TCP Reno is similar to TCP Tahoe and the only difference is that it works with fast recovery. At the arrival of three consecutive duplicate packets (DUPACKs), the TCP Reno starts recovery by sending the last packet back to the sender in order to reduce the threshold of the slow start method and reduce the current window size to half. Then increment the window to maximum segment size for each duplicate acknowledgement. If new positive ACK received (not DUPACK), the congestion window resets with the slow start threshold and enter into congestion avoidance phase (CAP) which is similar to TCP Tahoe. Hoe [5] presented an approach called *new-Reno*, in which TCP dispatcher does not exit the rapid improvement state, once the latest ACK is established.

TCP Vegas, proposed in [3], improves the congestion avoidance phase by controlling buffer space and modify the slow start mechanism of TCP Reno. The RTT value is precisely computed in the TCP Vegas approach. Both TCP Vegas and TCP Reno cannot perform well in the networks with the high bandwidth-delay product (BDP). The technique used in both the algorithms is Additive Increase Multiplicative Decrease (AIMD) which is conservative and is not designed for large window size to recover after a backoff, and also the bandwidth is not effectively utilized [6]. However, all traditional TCP approaches degrade the throughput in MANETs due to various reasons such as misinterpretation of packet loss and congestion window, frequent path breaks, the effect of path length, asymmetric link behavior and multipath routing.

Chandran et al [7] presented a ***feedback based TCP***, where the source node gets information about the route failure due to the mobility of nodes. When a link between the source and destination breaks, the node which detects the route failure, sends the route failure notification (RFN) information to the source node, so that source node establishes another route for faithful transmission. However, this method cannot reproduce the faithful transmission in a new route because of the congestion window.

Holland and Vaidya [8] presented a ***TCP with Explicit Link Failure Notification*** (TCP- ELFN) that improves the quality of traditional TCP by handling the link failure notification and TCP probe packet for the establishment of the route. In this approach, once the source receives failure notification, it stops transmission and put in a standby mode. In such scenario, it regularly generates probe packets to check whether the new route is reestablished or not. In case the ACK is received by the TCP receiver, it resumes the retransmission and starts functioning normally. However, this approach does not perform well if the network is provisionally divided into parts in which the failure of the path may remain for a longer time and thereby lost the path resulting in high bandwidth usage as well as more power consumption.

Several alternatives have been suggested to current TCP in high speed networks such as Extended Copy Protection(XCP) router algorithm[9], Scalable TCP[10], TCP Westwood[11], High Speed Transmitted Control Protocol (H-TCP)[12], Binary Increase Congestion TCP (BIC-TCP)[13], Fast TCP[14] and Compound TCP[15]. Though these algorithms have certain advantages over standard TCP, however, they do not significantly perform better than the standard TCP.

Avrachenkov and Antipolis [16] presented an analytical method for the optimal choice of the router buffer size. However, their method corresponds to a linear combination of average sending rate and average delay in the queue. It does not compute the loss of packets and utilization of the link. David et al [17] describes congestion control algorithm for high-speed networks and presents the difficulties faced in current TCP algorithm. Sarker and Johansson [18] achieved a good result for calculation of loss and delay by using Long Term Evolution (LTE) simulator based on Explicit Congestion Notification (ECN) on packet marking and adaptation, from real-time video communication. Bauer et al [19] demonstrated that the development of LTE networks result in an increase in the development of ECN. On the other hand, there is no effect (or least effect) on buffering the packets when an initial window size increases [20]. Gettys [21] presented problems in Buffering bloat, which is a phenomenon used in a store and forward switching networks, particularly in packet-switched computer networks. The problem related to this technique is that excess of buffering of packets results in latency and jitter and at the same time reducing the network throughput. Jarvinen et al [22] showed the limitation of Random Early Deduction (RED) algorithm that it does not handle fast changes due to TCP slow start when the traffic is limited. All the above works either suggest the size of the buffer or increment /decrement the size of the window. However, to the best of our knowledge, there is no (or less) work on the precise computation of the packet loss, which is very necessary for multimedia transmission. This is because the allowed tolerance for multimedia transmission is up to a certain limit, which is mainly depending on the multimedia application and the delay. However, unlike multimedia transmission, the loss and in-turn the delay does not significantly affect the normal data transmission.

Dorri and Kamel proposed [23] a control system based on fuzzy logic to determine and balance the traffic in the wireless sensor network (WSN). In this approach, the source node selects one node in each grid called monitoring node, which continuously monitors the node to control and balance the traffic. However, this approach consumes more energy because of its continuous~~ly~~ monitoring. Uthra and Raja [24] proposed an approach to control the packet loss based on congestion. However, congestion has an adverse effect, i.e., when packet loss occurs node has to retransmit the packet again resulting in energy wastage and also reduced network lifetime. Karakus et al [25] proposed an approach to save the energy of a node. However, they did not consider the transmission of neighboring nodes of the destination, which increases packet loss probability and energy consumption of the node near the destination.

Visweswaraiya and Gurumurthy [26] presented an approach using medium access control to coordinate the nodes based on their priority to provide the congestion-less routing. However, this method uses queue buffer length to estimate the congestion and decide the route. This may not always be true as large queue length does not always indicate the occurrence of loss. However, if output link capacity is more, the length of the queue can be reduced quickly in comparison with the node that has less output link capacity and small queue length. That is, the resources of the link management (e.g. traffic intensity, bandwidth, queuing delay, energy, and distance) decide about the congestion control mechanism, rather than queue length alone. Antoniou et al [27] proposed an approach to control the loss of packets based on birds' behavior. However, this approach is implemented at the individual node, which results in a reduction in packet delivery ratio. Jaiswal and Yadav [28] presented a fuzzy control approach to estimate the congestion, buffer occupancy, participants, and traffic rate in order to balance the system.

Chia-Hsu et al [29] proposed a protocol for controlling the congestion to improve efficiency in sensor networks. However, the proposed protocol may not be accurate as it is not reliable in determining the packet loss which is to be controlled. Yao et al [30] developed a congestion control mechanism by employing software define technique which optimizes the channel. Najme et al [31] presented a scheme for congestion control by considering various parameters and they also proposed congestion prevention method to improve the efficiency of the network. Mudassar et al [32] made use of TCP CUBIC congestion control protocol for far distance and high spectrum networks to improve the infrastructure performance in the medical field. In this work, we first compute packets loss by modeling the number of active stations as a Poisson random process and then provide a logic circuitry to control the packet loss at the front end of the router buffer in order to improve the overall QoS.

III. PROPOSED APPROACH

Fig. 1 shows the block diagram of our approach. Microcontroller stores the addresses of all the active input mobile users (also called input mobile nodes). These input station addresses are used to switch off the appropriate input station when congestion occurs at the router. Usually, it is difficult to recognize such input stations as the position of the active nodes may change over a period of time. Microcontroller recognizes an input station that has to be controlled at a particular instance of time. Control of stations in this way reduces loss and thereby increases the utilization of the link

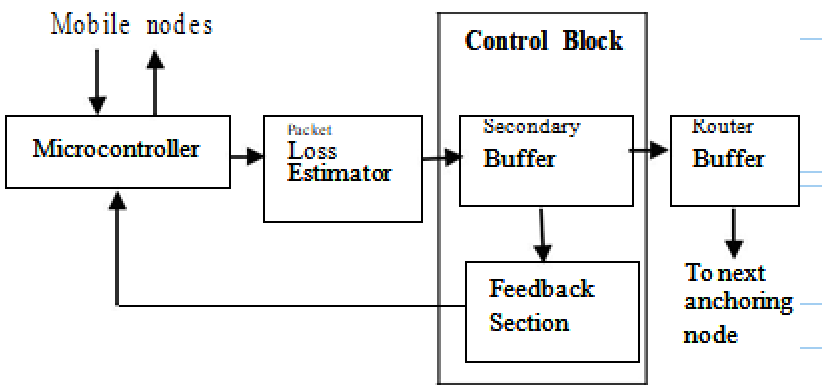

Fig. 1 High-level overview for controlling Packet Loss

Packet Loss Estimator (PLE) computes the loss by modeling the active stations as a *Poisson random process* as per the actual scenario [33]. This estimator computes the packet loss due to packet overflow in the router, active input stations and the packets arriving time from the active stations at the router. Also, the estimator calculates the idle and busy periods of the outgoing link from the router to the next anchoring node within the network. The output (i.e. active stations, loss of packets and the arriving time of the packets) of the estimator is given to the router through control block that encompasses a secondary buffer and a feedback circuit.

The secondary buffer is used to store a bit, which is set based on the PLE estimated value. The bit is set to zero in case the estimator estimates packet transmission towards router is less than or equal to a certain value, say α, indicating that there is no packet loss and thus no information is passed to the feedback circuit. If the estimator sets the secondary buffer bit to one, indicating there is an occurrence of packet loss and thus signal information is passed through *feedback circuit* (see section 4) in order to control the loss. In our framework, PLE and control block works in tandem to control the packet loss so that the utilization of the outgoing link capacity of the router can be increased. This also results in an increase in the QoS of multimedia transmission. On the other hand, due to additional hardware, there will be a slight delay when compared to existing approaches [1, 2]. However, the increase in the delay may, in many cases, be ameliorated by the gain towards controlling packet loss by our approach. That is, the overall delay can be reduced by packet loss prevention in our approach. Also by the use of high-speed processors, the delay due to additional hardware can be further minimized so that the overall QoS can be improved.

IV. PROPOSED FA-CCM

In this section, we first present the operation of the control circuit in order to control the loss of packets. Table 1 shows the response of control signals in our work. When D0 is 0, (that is no information is passed through D0), it stops transmission from station one (i.e. S1) for one RTT. When D1 is zero, it stops transmission of two input stations for '1' RTT (i.e., S2 & S3). This process continues for D2, D3, and so on till secondary buffer bit is zero.

TABLE 1. THE RESPONSE OF CONTROL SIGNALS

S.No.	Control signal	Stations control
1	$-_{0}$	One station (S_1) only
2	$\cdot_{1}$	Two stations (S_2 & S_3)
3	$\cdot_{2}$	Three stations (S_4, S_5 & S_6)
4	$\cdot_{3}$	Four stations (S_7, S_8, S_9& S_{10})
5	$\cdot_{4}$	Five stations (S_{11}, S_{12}, S_{13}, S_{14} & S_{15})
6	$\cdot_{5}$	Six stations (S_{16}, S_{17}, S_{18} ,S_{19}, S_{20}& S_{21})
7	$\cdot_{6}$	Seven stations (S_{22}, S_{23}, S_{24}, S_{25} $S_{26,}$ S_{27}& S_{28})
8	$\cdot_{7}$	Eight stations (S_{29}, S_{30}, S_{31}, S_{32} $S_{33,}$ S_{34}, S_{35} & S_{36})

Fig. 2 represents the circuit diagram for FA-CCM (see Fig 1). All the active input nodes have to transmit through port one (i.e. port P1) of the microcontroller to store the addresses of all the input stations, then all the stations transmit data packets to the secondary buffer through port two (i.e. port P2) of the microcontroller. If secondary buffer bit is one, it indicates the existence of packet loss, otherwise no packet loss at the router. So, no control action takes place when the buffer bit set at zero value, and in the case of loss, the control action takes place through port 3 of the microcontroller (see Fig. 2). The circuit consists of 3 bit asynchronous counter, 3 to 8 active low De-multiplexer, AND gate, OR gate and an inverter. The 3 to 8 de-multiplexer is used to control the 2^3 output signal lines. Note that one can use higher order de-multiplexer if input stations to be controlled are more than 25. In the given circuit diagram, the congestion information is recognized with the help of secondary buffer bit.

In this circuit diagram, if the secondary buffer bit is '1' and clk is also '1', it sets the output of AND gate to '1', which indicates is also '1'. As the clear signal is given to NOT gate from buffer bit, this is '1' in this case. So the utput. The counter output is now '000', means it sets D0 to '1'. As the de-multiplexer is active low so $\overline{0}$ will be is also '0', means off station one (i.e. S1). If again secondary buffer bit is one '0'. So the output of OR gate i.e. 0

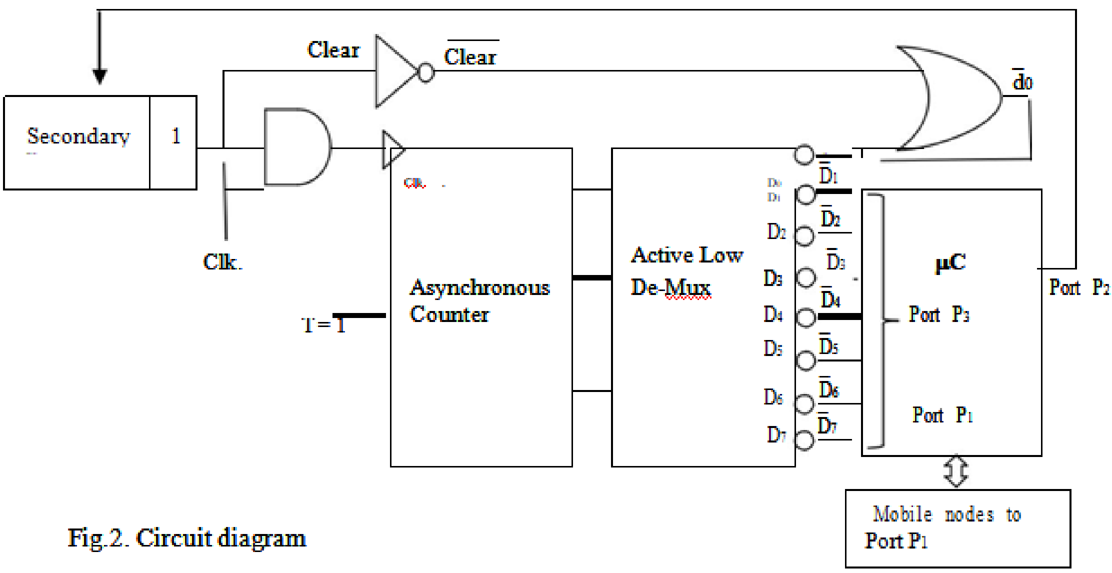

Fig.2. Circuit diagram

Here, the output of this de-multiplexer is used to turn off the input stations through port P3 (see truth table shown in Table 2), so that the accurate station has to stop for a period of time (i.e., One round trip time (RTT)). Because the time to live of a packet is always more than a normal RTT (normal RTT means the RTT when there is no congestion). Next time, if secondary buffer output bit is again 1 the controller stops the packet transmission of next three stations but not the previous one which was already stopped earlier (see fig. 3) so that the previous station cannot face synchronization problem. This control process always increments one more station to stop packet transmission if secondary buffer bit is again 1. But it cannot stop the transmission of those stations which was already stopped earlier. This process continues till the secondary buffer is empty. If the secondary buffer bit is 0 thenthere is no packet loss so ̄ becomes high and counter clears its output, means output is '000'indicates I[st] count. So de-multiplexer selects □D0 which is active low and hence □d0 becomes high since is active high (i.e. 1+0=1) and hence no station turned off as shown in Table 1. Fig. 3 shows the proposed FA-CCM protocol. When the router buffer is overloaded, the secondary buffer bit is set to one indicating the congestion. It sends the negative acknowledgement to the active input stations by generating a control signal $\bar{}_0$to stop transmission of packets from active source node S_1 for a period of one RTT. At this stage, the transmission of remaining active input stations (i.e., S_2, S_3, S_4,..., S_N) do not effect.

TABLE 2. TRUTH TABLE

Buffer Bit	Clk.	$\overline{\text{Clk}}$	$\overline{\text{Clear}}$	Counter Output	De-Mux. Output
1	1	1	0	Starts counting 000 = 0	D_0 =1, D_1 to D_7 = 0; $\bar{D}_0$ = 0, $\bar{D}_1$ to $\bar{D}_7$ = 1; $\bar{d}_0$ = 0
1	1	1	0	001 = 1	D_0 =0, D_1 = 1, D_2 to D_7 = 1; $\bar{D}_1$ = 0, $\bar{D}_0$ = 1, $\bar{D}_2$ to $\bar{D}_7$ = 1; $\bar{d}_0$ = 1
0	1	0	1	Clears O/P 000 = 0	D_0 =1, D_1 to D_7 = 0; $\bar{D}_0$ = 0, $\bar{D}_1$ to $\bar{D}_7$ = 1; $\bar{d}_0$ = 1
1	1	1	0	000 = 0	D_0 =1, D_1 to D_7 = 0; $\bar{D}_0$ = 0, $\bar{D}_1$ to $\bar{D}_7$ = 1; $\bar{d}_0$ = 0
1	1	1	0	001 = 1	D_0 =0, D_1 = 1, D_2 to D_7 = 1; $\bar{D}_0$ = 1, $\bar{D}_1$ = 0, $\bar{D}_2$ to $\bar{D}_7$ = 1; $\bar{d}_0$ = 1
1	1	1	0	010 = 2	D_0, D_1 = 0, D_2 = 1, D_3 to D_7 = 0; $\bar{D}_0$, $\bar{D}_1$ = 0, $\bar{D}_2$ = 0, $\bar{D}_3$ to $\bar{D}_7$ = 1; $\bar{d}_0$ = 1
•••	•••	•••	•••	•••	•••
1	1	1	0	111 = 7	$\bar{D}_7$ = 0, $\bar{D}_2$ to $\bar{D}_7$ = 1; $\bar{d}_0$ = 1

In case, the router buffer is overloaded again, the secondary buffer is set to one and it sends a negative acknowledgement to active input source stations by generating a control signal $\bar{}_1$ to stop transmission of packets from source nodes 2 and 3 for a period of one RTT. However, it will not stop transmission of data packets of remaining active source nodes (i.e., S1, S4, S5, · · ·SN). This process continues till secondary buffer bit is set to zero. Fig. 4 shows the Pseudo code of the proposed approach. Step 1 represents parameter initialization for a number of iterations n, controlled signal j, and overflow counter k. Initially, the addresses of input stations are stored in the microcontroller and control signal ̄[j] is set to zero (steps 2 and 3). Microcontroller transmits the data towards the router from its port 1 to port 2 (step 4). Step 5 shows the status of the secondary buffer bit *b*. If the buffer bit is set to zero, it will not disturb any input station, that is, there is no need to control any station. Otherwise, control action has to be taken place (i.e. step 6). Step 7 to step 11 indicates that if the secondary buffer bit is set at 1, then accordingly it generates the control signal (represented by step 12) according to FA-CCM, so that desirable station or stations(see Table 1) will stop transmission of packets during given amount of time (here, it is 1RTT).

Table 3 shows the values for *packets generated* and *loss of packets* at α =1, where τ is defined as the time interval during which an active station completes the packet transmission and same is received at the router station and active nodes are varied from 1 to 25. Fig. 5 shows the packet loss with respect to λ, where λ is the average random variable value per interval (λ ≤ α) for different values of α (*i.e.*,

α = *1, 2, 4).*From the figure, we can observe that the packet loss is drastically reduced by using the control logic circuitry in the router circuit when compared to existing approaches (i.e., window based approaches such as TCP Reno and TCP Vegas).

Window-based techniques are useful when the output capacity of a router is less in comparison with the packets coming from the mobile users to the input of the router (i.e. active input stations are more and output capacity of the router is very less), as these techniques divide the size of the window to half when there is a loss. However, the window-based technique takes a longer time to reach to steady-state value (i.e. actual value of the window at which loss occurs), and thereby reduces the QoS of multimedia applications in MANETs, as the multimedia applications cannot tolerate packets that arrive after TTL. That is, though these techniques overcome the packet loss by decreasing the window size, they show a reduction in the percentage of packet delivery ratio because the packets will not reach to a destination within time to live (TTL) of a packet.

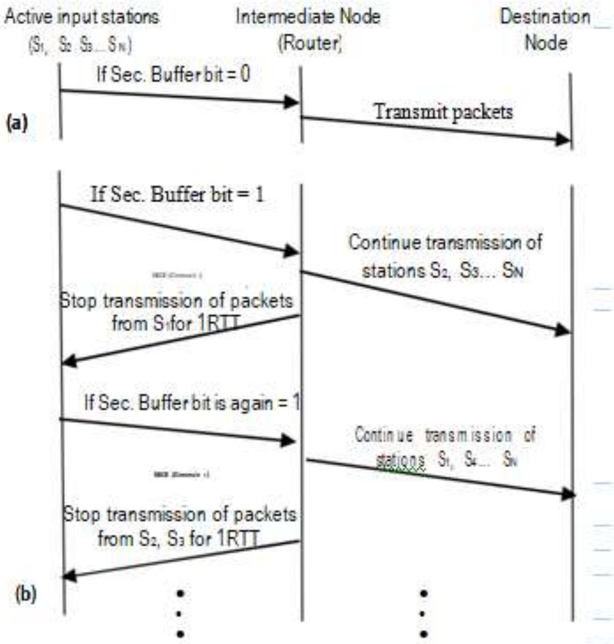

Fig.3. FA-CCM protocol (a) secondary buffer bit is zero and (b) secondary buffer bit is one

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Input: Input stations S[i], Secondary buffer bit b;
Begin
 1.Int i, n = 0, j = 0, k = 0  // initialization
 2. Map input stations S[i] to port 'P1' of the
      microcontroller
 3. D̄ [j] = 0;
 4. P1 transmits data to router through port 'P2' of the
      microcontroller.
 5.Check the secondary buffer bit b status;
 6.While (b == 0) {
 7.          n++;
 8.          For (i = 0; i< n; i++) {
 10.              k++;
 11.          }
 12.          Off. [j]. // Stop packet transmission from stations corresponding to the
 controlled signal//
 13.         j++;
 14. }
End.
```

Fig.4. Pseudo code of the proposed approach

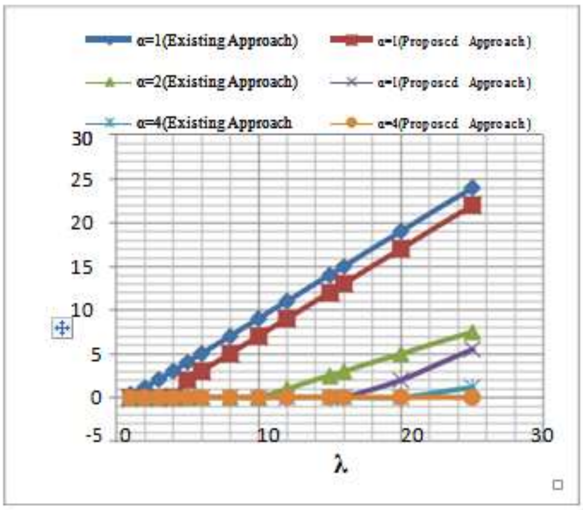


Fig.5. 'λ' versus Packets Loss

TABLE 3. SIMULATION RESULTS FOR G, L, D.

FOR α=1, τ = 5MS WITH CONTROL CIRCUITRY

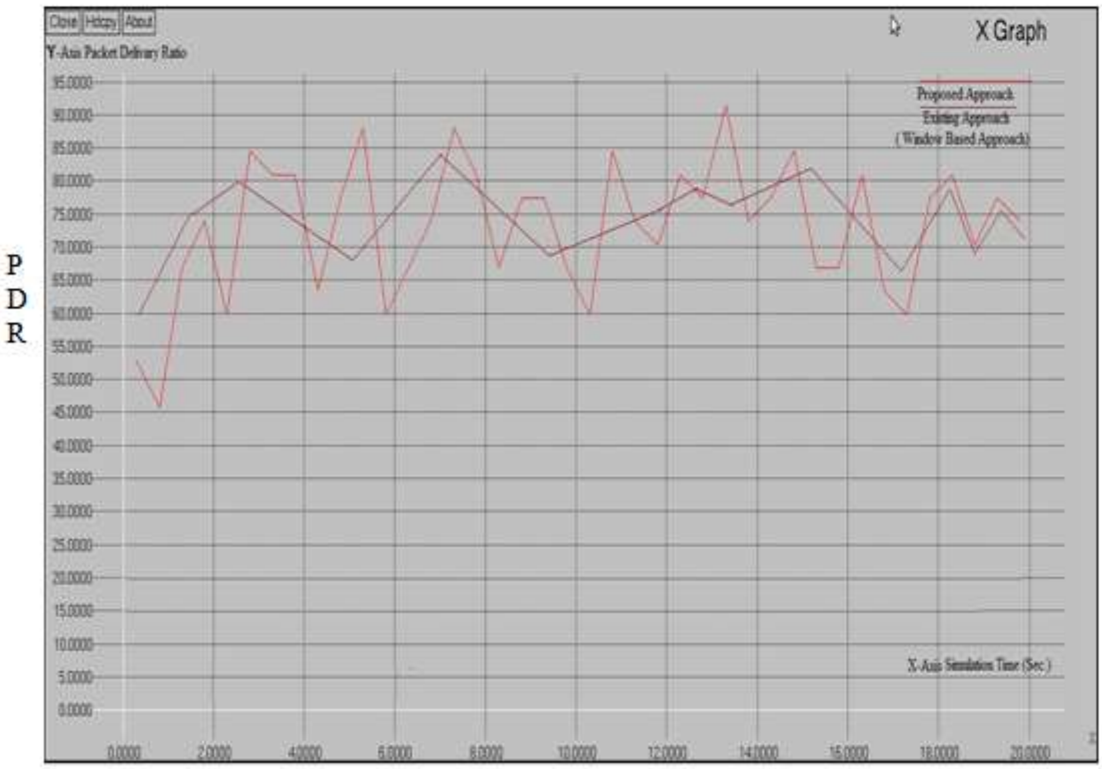
Active nodes in time interval τ (λ)	Packets generated (G)	% packets loss (Window based approaches)	% of Packet Loss (Proposed approach)
1	1.0	0	0
2	2.0	1	0
3	3.0	2	0
4	4.0	3	1
5	5.0	4	2
6	6.0	5	3
8	8.0	7	5
10	10	9	7
12	12	11	9
15	15.0	14	12
16	16.0	15	13
20	20.0	19	17
25	25.0	24	22

TABLE 4. SIMULATION PARAMETERS

Network Parameters	Values
Simulation Time	50 seconds
Number of nodes	2 to 50
Pause Time	30 sec.
MAC type	802.11
Radio Propagation Model	Two-Ray Ground
Queue Type	Drop-Tail
Antenna	Omni antenna
Routing	LAEERP
Network Area	1000m x 1000m
Channel capacity	1Mbps
Traffic	Video, Voice
Network Area	1000m x 1000m
Simulation Speed	2,5,8,10,12 m/sec
Pause Time	50 sec.
Packet Size	1Kbyte

Further, the link utilization is poor when a loss occurs at a very nearer value of window (e.g. if the window size is 64 and loss occurs at 63, so it has to come down to 32 window size). That is, even there is a loss of only one packet, window-based techniques reduces the window size to half and thereby reduces link utilization by 49.01% and also degrades the QoS in MANETs.

Table 4 shows the simulation environment along with the parameters considered. Total simulation time is 50 sec. Fig. 6 shows the variation of Packet Delivery Ratio (PDR) with respect to simulation time of the source-destination pair. For routing, we used the routing protocol defined for Long Distance MANETs [34]. Initially, PDR of window based approach is more than our approach. This is because there is no loss at the initial stage. However, after elapse of the certain time period, this approach doubles the window size (i.e. when no loss) in comparison with the previous window. So, when congestion occurs, there may be a chance of large packet loss. As per the window-based approach, it has to come down to half of the window and takes a large time to maintain a steady state, which results in ups and downs in the simulation curve. In our approach, we control the packet loss as per the loss estimation, which also results in overshoots, because of transmission of active input stations changes with respect to time. However, our approach still maintains higher PDR when compared to window based approaches, as our approach directly controls the loss of packets as per the actual scenario happened at the output of router.



Simulation Time

Fig.6. PDR Vs Simulation Time

V. CONCLUSION

Transmission control in MANETs is very important for transmission performance and link utilization, especially when a bunch of wireless devices communicates with each other in a crowded place. In this paper, we presented a feedback and active congestion control mechanism for successful multimedia transmission over MANETs. In multimedia transmission, improved QoS can be obtained by accurately computing the packet loss at each intermediate node which acts as a router in MANETs. Also, increased usage of bandwidth day-by-day necessitates the effective use of bandwidth without any wastage. If more bandwidth is reserved for the connection, then better QoS can be maintained for multimedia connections without any control strategy. This paper presents the congestion control protocol and a control circuit which not only avoids the loss of packets but also increases the utilization of link.

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