DOI: 10.21917/ijct.2014.0139

ANALYTICAL CHARACTERIZATION OF WLANS FOR QUALITY-OF-SERVICE WITH ACTIVE QUEUE MANAGEMENT

M. Usha[1] and K. Madhana[2]
[1]*SonaNET Network Computing Research Center, Sona College of Technology, India*
E-mail: usha@sonatech.ac.in
[2]*Department of Computer Science, Sona College of Technology, India*
E-mail: ksmadhana11@gmail.com

Abstract

Design of an Active Queue Management scheme at the Access Point to address the problem of congestion control, packet delay variation and packet loss rate is discussed. The proposed mechanism calculates and adjusts redundancy rate adaptively at the access point by considering both network traffic load and wireless channel condition. Real-time applications such as Mobile learning and smart learning need the special treatment and require differentiated QoS to satisfy the client who is ready to pay more than others. Maintaining the jitter value of the multimedia packets below the threshold is essential to guarantee the desirable quality of the video at the receiver. The work initially concentrates on minimizing the packet loss of such priority flows and they have to be given place in the queue even at the time of buffer overflow. Thus the proposed work uses push-out policy to provide differentiated services to the multimedia flow which achieves considerable improvement in the video quality at the receiver. The considerable decrease in packet loss rate and special treatment in the queue of the access point lowers the packet delay variation of the multimedia flow. The results show that the AQM used at the access point effectively achieves low packet loss, low jitter using differentiated FEC rate calculation without generating congestion in the wireless network.

Keywords:

Active Queue Management, Wireless Access Point, Quality of Service, Push-Out Policy

1. INTRODUCTION

As Technology is attaining vast growth, numbers of users prefer to connect to the Internet through wireless devices. A variety of multimedia services like video conferencing, On-demand learning, mobile learning are in high demand and thus face diverse challenges such as attenuation, fading, interference from active sources during transmission in wireless network environments. Such challenges lead to packet losses which are recovered by two techniques: Automatic Repeat Request (ARQ) and Forward Error Correction (FEC). ARQ recovers packets by retransmitting the lost packets during timeouts or responding to the explicit requests from the receiver whereas FEC [9], [10] acts in advance by sending redundant packets along with original packets to recover successfully at the receiver in case of packet losses. Out of the two approaches, FEC guarantees the low retransmission delay and variation in delay (also known as 'jitter') and thus suitable for real-time data transmission.

Multimedia data finds the way difficult to transmit through wireless network because of low bandwidth, packet losses and delay. With the rapid increase of data transmission including data, voice, video and mobility supported by a common IP platform, quality-of-service provisioning has grabbed primary focus from the problem of congestion control. Since various types of traffic are carried through the Internet, multimedia flow requires differentiated services from other network services. The Active Queue Management (AQM) mechanisms are helpful for providing differentiated services to reduce delay variation or jitter, packet loss and bandwidth depending on mutually agreed upon service level agreements (SLA). Real-time applications require QoS guarantees such as high throughput, bandwidth [1]-[3], low delay and jitter to transmit the audio or video with better quality. A guarantee, in the sense, fixes a maximum threshold above which degradation in quality must not increase.

In the multimedia transmission used in applications such as Learning-on-Demand, packet delay variation is not acceptable. Jitter is defined as an end-to-end delay between the selected packets in a particular flow. The inter-packet arrival time plays an important role in providing QoS for real-time applications. The differentiated services are needed to reduce packet delay and variation to improve multimedia quality.

Congestion control is again a big challenge in wireless networks. Congestion affects the QoS characteristics and so it should be effectively managed by the queue management mechanisms. Tail drop is one such mechanism to control congestion [4]. It drops the packets at the tail of the queue during buffer overflow. It is not suitable for real-time data transmission. AQM [5]-[7] is proposed to control end-to-end congestion by adapting Random Early Detection (RED) [8] which drops the packets earlier to avoid congestion to eliminate buffer overflow. Thus it achieves high system utilization and low packet delay and suits for real-time applications.

The fundamental contributions of this work are:

- Reducing packet loss rate by using FEC mechanism.
- Controlling congestion using AQM mechanism.
- Reducing packet delay and jitter to achieve multimedia quality.

The mechanism called priority-based FEC (PFEC) is proposed to attain above achievements. To differ from the previous study [35], we include Jitter analysis at the Access Point (AP) level for real-time data transmission.

2. RELATED WORK

There are many researches which have been done on AQM mechanisms and FEC calculation to provide QoS. Many sender based FEC calculations takes finite duration and degrades the received data performance. Many traditional AQM mechanisms adapt RED which randomly drops packets when the buffer overflows and not suitable for real time multimedia applications.

2.1 AQM MECHANISM

Floyd proposed RED algorithm [8] which is a simple mechanism and controls congestion by accompanying the transport layer congestion control protocol. The RED gateways can detect congestion prior by continuously monitoring the average queue size. RED works in contrast to other traditional queue mechanisms. RED drops the packets with a probability according to the queue size. When the average queue size grows, the dropping probability also increases and reaches 1 when the buffer is full. The disadvantage of RED algorithm is that it does not provide differentiated QOS [15]. Since a single set of RED parameters are not available it does not react well to different congestion scenarios [19].

The integration of the RED mechanism with the priority scheduling paves solution for congestion control and service differentiation in [12]-[14]. The scheme uses partitioned buffer with corresponding thresholds. If the size of a particular class is filled then the packets with higher priority than that of the filled class will enter into the queue and those with lower priority will be dropped. But this scheme uses Short-Range-Dependent (SRD) arrival processes and so SRD cannot capture multimedia traffic nature [15].

To remove this complexity an analytical model [15] is introduced, which achieves service differentiation for different types of traffic flowing through the traffic. The buffer $[X_0 - X_k]$ is virtually divided into many partitions such as $[X_0\text{-}X_1,\ldots,X_i\text{-}X_{i+1},\ldots,X_{k-1}\text{-}X_k]$ as in buffer partition [16]. The packets of each flow occupy the corresponding partition. If any particular partition gets filled then the packets of corresponding flow will be dropped and only the packets of higher priority flow will be allowed. The issue presented here is that the scheme follows normal FIFO technique for virtual buffers which drops the incoming higher priority flows when the capacity of the respective partition overflows.

The buffer partition scheme [16] is introduced to manage multi-class buffer and admission control for the buffer. It extends the bandwidth concept to utilize the bandwidth efficiently by the Markovian traffic. The thresholds of the partitioned buffer are adaptively changed to tolerate the loss probabilities of the incoming traffic and to manage the input traffic load into the network.

An algorithm called Loss Ratio-based RED (LRED) [6] is developed to control congestion, achieve link utilization and low delay. It maintains stability of the network by adaptive adjustment of the network parameters. It monitors the average packet loss ratio to adjust the dropping probability of the packet adaptively to maintain the queue length stable. It achieves fast response time and good robustness.

2.2 PUSH-OUT POLICIES

There have been many push-out policies developed to utilize the buffer efficiently and to maintain a constant ratio of packet loss rate among different traffic flows.

The concept of maintaining multiple classes for different flows leads to high computing complexity in multi-class push-out policy. To overcome this problem, Partial Buffer Sharing [16] has been designed. Yet they drop the incoming higher priority packets when the buffer overflows though there are lower priority packets available in the buffer.

An efficient buffer sharing scheme [17] in ATM switches is also discussed to eliminate the same above problem. It replaces the less important packets in the queue by the incoming high priority packets when the average queue size exceeds the maximum threshold. This scheme attracts the researchers for its simple implementation and high performance.

The dropping of higher priority packets is eliminated by the push-out policy [18] with AQM which is developed in IP routers. It works by letting the higher priority packets to enter the queue by discarding lower priority packets already present in the queue even when the buffer overflows. It reduces packet loss rate compared to RED.

Another mechanism called Flow-based Priority Queuing (FPQ) [32] is introduced to give preference to real-time UDP flows at intermediate routers. Thus it reduces packet loss rate, delay and jitter and at the same time it maintains fairness among all the incoming traffic flows.

2.3 FEC MECHANISM

Many FEC mechanisms provide dynamic QOS control of real-time multimedia applications. Sender-based mechanisms [19], [20] calculate the redundancy rate at the sender which takes finite duration because of receiving loss reports from the receiver. Also it does not ensure that the sender predicts the current network condition. It causes variation in the delay between receiving packets at the receiver which greatly affects the multimedia quality. During congestion it increases the rate of redundant packets which may further increase the congestion and degrade the network performance.

Access-point based approaches have been proposed to eliminate the duration needed by the sender for redundancy rate calculation. The RED-FEC mechanism [21] does it and controls congestion by decreasing redundancy rate as queue size increases. It works by increasing redundant rate as the queue size decreases and decreasing the rate when the average queue size exceeds the maximum threshold. The limitation here is that it does not consider the packet loss rate.

The cross-layer based FEC mechanism – Adaptive Cross Layer FEC Mechanism [22] has been introduced to include the packet loss rate for redundancy rate calculation which is retrieved from the ARQ function of the MAC layer. In this mechanism, the redundant rate changes in proportion to the packet loss percentage. It increases the redundant rate when numbers of packets are lost and decreases when packet loss is decreasing. But it does not consider network traffic load for rate calculation.

ERED-FEC mechanism [23] has been developed to calculate the redundancy rate depending on both factors such as wireless channel condition and network traffic load to eliminate the problem of congestion due to excessive number of introductions of redundant packets which is found in the above two Access Point based approaches. The limitation found here is that it cannot provide differentiated QoS which is essential for multimedia traffic to achieve better QoS.

2.4 JITTER REDUCTION APPROACHES

A Jitter Detection method [24] has been proposed for gateway-based congestion control to transmit multimedia in packet switched networks. It introduces an AQM mechanism to improve QoS for multimedia transmission by jitter detection. It reduces jitter by detecting and discarding multimedia packets which accumulated more jitter to maintain high bandwidth for other good multimedia packets. The packets which accumulated jitter more than the jitter tolerance level become useless for clients to recover them and they degrades the QoS of the entire multimedia flow. Thus this method eliminates quality degrading and controls congestion by dropping unwanted packets. The jitter reduction not only provides high bandwidth but also achieves high throughput for multimedia flow.

It is more essential to evaluate the quality of multimedia using application-level evaluation metric like Decodable Frame Rate (DFR) than network-level metric such as packet delay, packet loss rate. The DFR evaluation [25] is used to analyze the effects of lossy wireless networks. A comparison of distribution packet loss and burst packet loss in wireless networks is done. The effect of the size of the play-out buffer and the size of the transmission are discussed because they affect the jitter quality of the received video. The packets which arrive at the receiver later than the needed time are dropped from the play-out buffer which affects the video quality.

It is now evident that PFEC mechanism results in controlling congestion and reducing jitter for multimedia transmission. Thus it maintains fairness among all the incoming flows and gives service differentiation to the preferred flow of multimedia by differentiated FEC redundancy rate calculation.

3. PROPOSED WORK

The current work focuses on an AQM scheme at the AP to improve Quality-of-Service of the multimedia data along with FEC redundancy rate calculation by prioritizing the incoming traffic. Initially the model of the system is explained and then a qualitative study is performed in this section.

3.1 SYSTEM MODEL

The topology assumed here is illustrated in Fig.1. The system classifies the incoming packets as TCP having variable bit rate and UDP having constant sending rate. When the sender is ready to send the video file they are encapsulated as video packets at the application layer. The identification of multimedia stream is based on encoding in the protocol field of the IPv4 header at the network layer. The priority of the particular flow is set by the sender in the Differentiated Service Code Point (DSCP) field of the IPv4 header. In the MAC layer, the ARQ function is responsible to control congestion by adjusting the transmission rate of the sender after receiving the response from the network.

At the access point, several functions will be performed as shown in Fig.2. FEC encoding is performed on the incoming packets while sending them out of the access point. The priority of incoming flow is checked in the DSCP field of the IP header. The packet loss monitor continuously monitors the wireless channel condition and calculates the packet loss of each flow. Since the evaluation metric called Decodable Frame Rate (DFR) is essential for application level users, the quality of the video is measured using DFR which will be explained later. The network traffic monitor analyses the network to find the load that is carried by the network at the given point of time. It is needed to adjust the redundancy rate explained as follows.

The FEC redundancy rate calculation includes two phases for each priority flows. It first monitors the queue length and compares it with threshold values. If the queue length is less than the minimum threshold then maximum number of redundant packets will be generated. If the queue length is more than the maximum threshold then no redundant packets will be generated. Otherwise FEC packets are generated based on the data size fraction in the queue. It then adaptively adjusts the FEC rate according to packet retransmission time. If the packets of the particular flow need maximum number of retransmission then it will show that the flow is experiencing high loss or error. So the rate increases to overcome the loss.

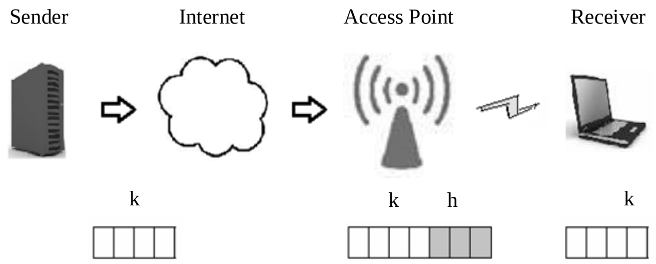

Fig.1. System Model

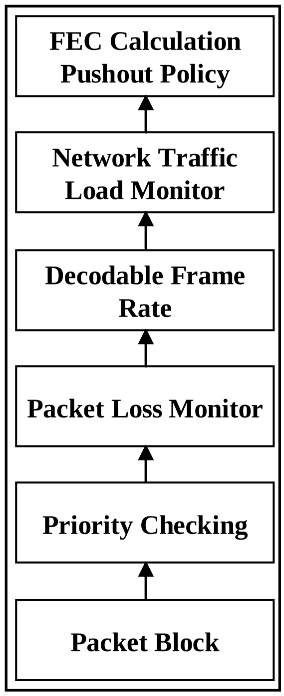

Fig.2. Functions of the Access Point

On the other hand, if the packets need zero or minimum number of retransmission then the AP will realize that the network condition is good. So there is no need to send the redundant packets and thus the rate decreases accordingly.

1) If the incoming packet is of high priority, packet retransmission time is low since the queue gives more preference to them by the preemption of low priority packets present in the queue during the buffer overflow; no or minimum number of FEC redundant packets have to be encoded.

2) If the incoming packet is of low priority, packet retransmission time is high due to the high loss rate of them when compared to high priority packets; FEC redundancy rate is high to balance the loss rate. This provides fairness to low priority flows.

The Access Point adjusts the redundancy rate for incoming traffic to provide differentiated service, based on wireless channel condition and network traffic load to avoid packet loss and control congestion, and to reduce the jitter.

3.2 QUALITATIVE ANALYSIS

Consider the sender transmits the video file with k packets per transmission block. The total number of video packets to be sent is N_{pkt}.

According to EAFEC, it is proposed that the effective packet loss rate is given by,

$$P_{FEC} = P_{block} - \frac{\sum_{i=0}^{k-1} C_i^{k+h} \times \left(1-P_{T_{\max}}\right)^{i} \times \left(P_{T_{\max}}\right)^{k+h-t} \times i}{k+h} \quad (1)$$

where, 'T_{max}' defines the maximum retransmission time, 'P_{block}' defines the probability of a block that has not been recovered at the receiver.

$$P_{FEC} = 1 - \sum_{i=k}^{k+h} C_i^{k+h} \times \left(1-P_{T_{\max}}\right)^{i} \times \left(P_{T_{\max}}\right)^{k+h-i} \quad (2)$$

'$P_{T\max}$' defines the probability of a packet that has not been received correctly.

$$P_{T_{\max}} = 1 - P_{correct} = \left(P_{pkt}\right)^{T_{\max}} \quad (3)$$

where, 'P_{pkt}' indicates the probability of a packet's failure when it is transmitted just one time, '$P_{correct}$' indicates the probability of a packet which is received correctly and 'C_i^{k+h}' indicates all possible combinations of 'i' packets received in a block successfully.

The Pushout Policy [18] proposed that

$$E_{R_1}\left(\sum_{t=0}^{\infty}\left[C_h M(t) + C_1 N(t)\right]\right) \le E_R\left(\sum_{t=0}^{\infty}\left[C_h M(t) + C_1 N(t)\right]\right) \quad (4)$$

where, Ch = cost of a lost higher priority packet,

C_l = cost of a lost lower priority packet,

$M(t)$ and $N(t)$ are number of higher and lower priority packets those are dropped and $E(.)$ denotes expectation of cost of dropped packet.

R denotes the policy in which the buffer is considered as virtually partitioned as two portions Q_1 and Q_2. The incoming packets will be dropped when the whole buffer overflows. If Q_1 is not filled then all the packets will be allowed to enter the queue. If Q_1 is filled and Q_2 is not filled the incoming packets will be dropped with some dropping probability.

R_1 is the subset of R and denotes push-out policy. If Q_1 is not filled then all the flows will enter the queue. If Q_1 overflows and Q_2 is not filled then higher priority flows will append at the tail of the queue and lower priority flows will be dropped with some dropping probability. If the whole buffer overflows then the incoming higher priority packets will replace the already occupied lower priority packets in the queue.

The previous study proves that

$$P_{FEC_h} < P_{FEC_l} \quad (5)$$

$$P_{block} - \frac{\sum_{i=0}^{k-1} C_i^{k+h_1} X \left(1-P_{T\max_h}\right)^{i} X \left(1-P_{T\max_h}\right)^{k+h_1-t} Xi}{k+h_1} < P_{block} - \frac{\sum_{i=0}^{k-1} C_i^{k+h_2} X \left(1-P_{T\max_l}\right)^{i} X \left(P_{T\max_l}\right)^{k+h_2-t} Xi}{k+h_2} \quad (6)$$

where, P_{FEC_h} and P_{FEC_l} indicate the effective packet loss rate of higher priority and lower priority flows respectively.

$P_{T\max_h}$ and $P_{T\max_l}$ denote the maximum retransmission rate of high and low priority flows respectively.

Since preemption technique followed by the queue gives more preference to higher priority flows, they face minimum loss. So the retransmission rate of such packets is becoming low and the number of redundant packets to be appended is less. In contrast, since lower priority flows face high loss when compared to higher priority flows, retransmission rate and the number of redundant packets generated for such flows is high. This makes the whole left hand side term to be lesser than right hand side term in Eq.(6).

3.2.1 Standard Deviation Calculation:

To calculate the standard deviation, the variation in packet delay should be known. Delay variation or jitter is defined as deviation from the ideal timing of an event. In other words, jitter of a packet denotes the difference between receiving time and sending time of the packet. The Eqs.(7-9) present these.

$$Send\_Time\_Diff = send\_time[i] - send\_time[i-1] \quad (7)$$

$$Recv\_time\_Diff = recv\_time[i] - recv\_time[i-1] \quad (8)$$

$$j_i = Recv\_Time\_Diff[i] - Send\_Time\_Diff[i] \quad (9)$$

where, 'j_i' indicates the jitter for each transmitted packet.

The term receiving time of a packet at the receiver depends on the factors like propagation delay, queuing delay, retransmission time if it has been lost. The terms such as queuing delay and retransmission time grabbed the major concentration in this work.

Standard deviation is calculated as follows:

$$\sigma = \sqrt{\frac{1}{N_{pkt}} \sum_{i=1}^{N_{pkt}} \left(j_i - \mu\right)^2} \quad (10)$$

where, 'μ' indicates mean of jitter values in Eq.(10).

3.2.2 Decodable Frame Rate:

The video file is transmitted through the network as segment of small packets. A frame is said to be decodable frame if and only if at least enough number of packets in the frame reached the receiver so that the whole frame can be recovered at the receiver successfully even if one or more packets in the frame have been lost.

The MPEG literature [28] defines a standard in which there are three frame types – I, P, B frames in the compressive video streams. The 'I' frames are encoded independently. The P frames are encoded depending on preceding ‘I’ or ‘P’ frames in the video sequence. The ‘B’ frames are encoded depending on the proceeding and succeeding ‘I’ or ‘P’ frames in the video sequence.

It is customary to calculate the application level metric, DFR, denoted as (Q), as shown in Eq.(11).

$$Q = \frac{N_{dec}}{N_{total_I} + N_{total_P} + N_{total_B}} \tag{11}$$

where, N_{dec} = Total number of decodable frames in the flow,

N_{total_I} = Total number of I frames,

N_{total_P} = Total number of P frames and

N_{total_B} = Total number of B frames.

The value of Q ranges between 0 and 1.0. If Q is 1 then the frame will be completely sensible that loss of even one packet will make the frame undecodable. If Q is 0.6 then the frame will be considered as decodable and at most of 50% packets in the frame can be lost.

The calculation of DFR implies that the mechanism is alert about the loss of the most important frames in the frame. Though the loss probability of all packets in a particular flow is same, the loss of most important frame leads the other dependent frame on it as undecodable one. Hence the loss probability of I frames is highly sensitive.

4. RESULTS AND DISCUSSIONS

The proposed system is validated with the performance evaluation. The video file is transmitted as many blocks each of which contains ‘k’ (assume k = 8) video packets. It is assumed to be transmitted using unicast transmission. The maximum retransmission time needed for each lost packet is assumed as T_{max} = 4. Consider ‘n’ traffic flows be entering the wireless network through the Access Point among which ‘m’ flows are Video-on-Demand traffic. The ‘m’ multimedia traffic are using UDP as a transport protocol and remaining ‘n-m’ traffic are considered as TCP flows.

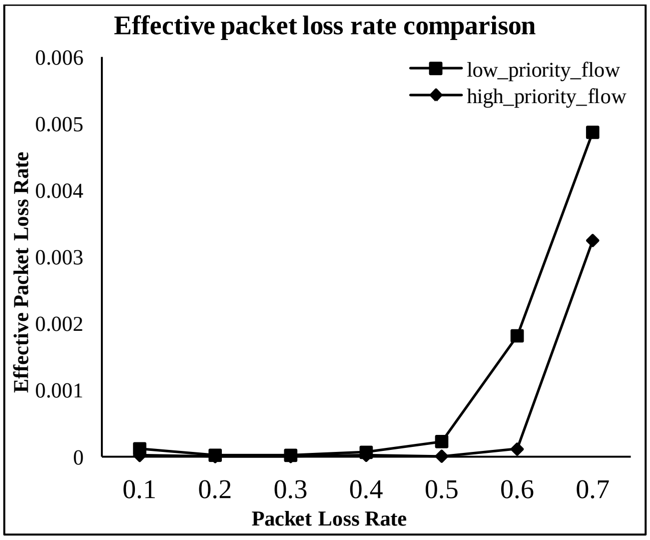

Fig.3. Comparison of effective packet loss rate of higher and lower priority flows

The maximum number of redundant packets to be generated for each block of 8 packets is 8. The redundant rates for lower priority flow is assumed intuitively as h_2 = 0,1,2,3,4,5,6 and that for high priority flow as h_1 = 2,3,4,5,6,7,8. Hence, when the packet loss is low the number of redundant packets generated for the low priority flow is 6 and that for QoS required flow is 8. No redundant packets would be generated for low priority packets and minimum number of redundant packets (say 2) would be generated for high priority packets even at the time of high packet loss. From the qualitative analysis of the previous study [35], a graph is shown to compare the effective packet loss rate of normal TCP flow and user demanded multimedia flow in Fig.3. The graph shows that the effective packet loss rate of multimedia flow is considerably less that of normal lower priority flow.

4.1 JITTER COMPARISON WITH AP-BASED FEC APPROACHES

From the extension of the previous study [35], analysis is done additionally on the evaluation metric i.e., jitter. The performance comparison of jitter values obtained for the multimedia flow using this mechanism are compared with that of EAFEC, one of the Access Point based FEC mechanisms. As we have analyzed in the section 3.2, the queue at the access point provides special treatment to higher priority flow. As a result, the retransmission time for such flow is less because they face less packet loss in the wireless network. Also the lower priority packets are getting replaced by the higher priority one during the buffer overflow. So the high priority flows do not suffer from high delay while waiting in the queue i.e., queuing delay and being processed in the queue. On the whole the preferred packets reach the receiver quickly when compared to other flows. This, in turn, lowers the jitter ‘j_i’ for the preferred multimedia flow.

On analyzing Eq.(10), the terms ‘j_i’ and ‘μ’ become substantially less for the client preferred flow which considerably minimizes the standard deviation (σ) of the flow. Thus the PFEC mechanism achieves lower packet delay variation for the multimedia flow as shown in Fig.4. The graph compares the jitter values in some frames of a transmitted video using EAFEC with PFEC mechanism. It proves that the jitter values in proposed approach are maintaining less value throughout the transmission of the video file when compared to that in the existing approach.

5. CONCLUSION

Adaptive FEC mechanism PFEC is proposed to improve the QoS of the multimedia traffic with congestion control. The qualitative analysis for the metrics packet loss rate and jitter are presented. The results show low jitter and low packet loss due to effective management of congestion using adaptive FEC with effective AQM. The redundant rate calculation depending on the priority of the flows enables the mechanism to provide service differentiation to the multimedia packets. Inclusion of adaptive bandwidth adjustment based on wireless channel condition will be considered as a future work.

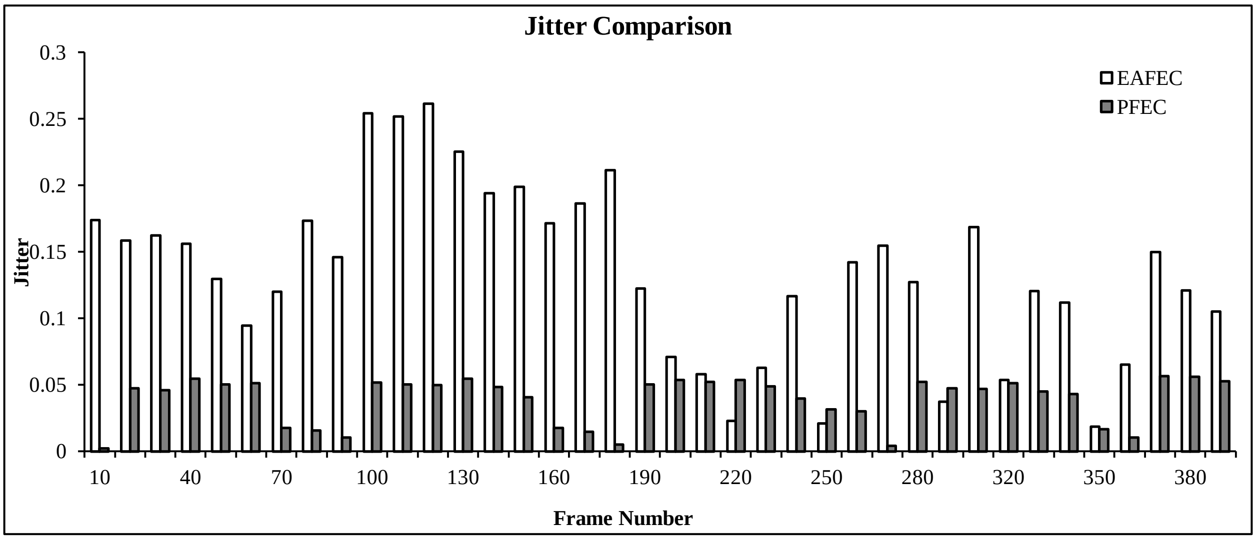

Fig.4. Comparison of Jitter in Proposed Approach PFEC with Existing Approach EAFEC

REFERENCES

[1] Liu Ping, S. C. Evans and Weerakoon Ishan, "Multimedia QoS through Content Aware Triage: An Integrated DiffServ Framework", *IEEE Military Communications Conference*, pp. 1-6, 2007.

[2] T. J. F. Omnes, T. Franzetti and F. Catthoor, "Interactive co-design of high throughput embedded multimedia", *Proceedings of the IEEE Design Automation Conference*, pp. 328-331, 2000.

[3] C. Oliveira, Jaime Bae Kim and T. Suda, "An adaptive bandwidth reservation scheme for high-speed multimedia wireless networks", *IEEE Journal on Selected Areas in Communications*, Vol. 16, No. 6, pp. 858-874, 1998.

[4] Shu Gang Liu, "Simulation and evaluation of random early detection in congestion control", *Proceedings of the International Conference on Machine Learning and Cybernetics*, Vol. 4, pp. 2363-2368, 2008.

[5] K. Chavan, R. G. Kumar, M. N. Belur and A. Karandikar, "Robust active queue management for wireless networks", *IEEE Transactions on Control System Technology*, Vol. 19, No. 6, pp. 1630-1638, 2011.

[6] C. Wang, J. Liu, B. Li, K. Sohraby and Y. T. Hou, "LRED: a robust and responsive AQM algorithm using packet loss ratio measurement", *IEEE Transactions on Parallel and Distributed Systems*, Vol. 18, No. 1, pp. 29-43, 2007.

[7] Geyong Min and Xiaolong Jin, "Performance Modelling of Random Early Detection Based Congestion Control for Multi-Class Self-Similar Network Traffic", *IEEE International Conference on Communications*, pp. 5564-5568, 2008.

[8] S. Floyd and V. Jacobson, "Random Early Detection Gateways for Congestion Avoidance", *IEEE/ACM Transactions on Networking*, Vol. 1, No. 4, pp. 397-413, 1993.

[9] J. Dan, F. Pascal and J. Aleksandar, "Forward error correction for multipath media streaming", *IEEE/ACM Transactions on Circuits and Systems for Video Technology*, Vol. 19, No. 9, pp. 1315-1326, 2009.

[10] E. Maani and A. K. Katsaggelos, "Unequal error protection for robust streaming of scalable video over packet lossy networks", *IEEE Transactions on Circuits and Systems for Video Technology*, Vol. 20, No. 3, pp. 407-416, 2010.

[11] A. Nafaa, T. Ahmed and A. Mehaoua, "Unequal and interleaved FEC protocol for robust MPEG-4 multicasting over wireless LANs", *Proceedings of the IEEE Conference on Communications*, Vol. 3, pp. 1431-1435, 2004.

[12] Y. Cheng and W. Zhuang, "Effective bandwidth of multiclass Markovian traffic sources and admission control with dynamic buffer partitioning", *IEEE Transactions on Communications*, Vol. 51, No. 9, pp. 1524-1535, 2003.

[13] A. Ioannou and M. G. H. Katevenis, "Pipelined heap (priority queue) management for advanced scheduling in high-speed networks", *IEEE/ACM Transactions on Networking*, Vol. 15, No. 2, pp. 450-461, 2007.

[14] Y. Li and S.-H. Yang, "PC-RED for IPv6: algorithm and performance", *Proceedings of the IEEE International Conference on Communications*, pp. 5581-5585, 2008.

[15] Geyong Min and Xiaolong Jin, "Analytical Modelling and Optimization of Congestion control for prioritized Multi-Class Self-Similar traffic", *IEEE Transactions on Communications*, Vol. 61, No. 1, pp. 257-265, 2013.

[16] Chuang Lin and Yin Li, "Dynamic Partial Buffer Sharing scheme: Proportional Packet Loss Rate", *Proceedings of the International Conference on Communication Technology*, Vol. 1, pp. 255-258, 2003.

[17] D. W. Petr and V. S. Frost, "Nested threshold cell discarding for ATM overload control: Optimization under cell loss constraints", *Proceedings of the Tenth Annual Joint Conference of the IEEE/ACM Computer and Communications Societies*, Vol. 3, pp. 1403-1412, 1991.

[18] M. Usha and R. S. D. Wahida Banu, "Push-out Policy in Active Queue Management to support quality of service guaranties in IP Routers", *Tenth IEEE International Conference on Communication Systems*, pp. 1-5, 2006.

[19] K. Takahata, N. Uchida and Y. Shibata, "Packet error and frame rate controls for real time video stream over

wireless LANs", *Proceedings of the 23rd International Conference on Distributed Computing Systems Workshops*, pp. 594-599, 2003.

[20] I. V. Bajic, "Efficient cross-layer error control for wireless video multicast", *IEEE Transactions on Broadcasting*, Vol. 53, No. 1, pp. 276-285, 2007.

[21] C. H. Lin, C. K. Shieh, N. K. Chilamkurti, C. H. Ke and W. S. Hwang, "A RED-FEC mechanism for video transmission over WLANs", *IEEE Transactions on Broadcasting*, Vol. 54, No. 3, pp. 517-524, 2008.

[22] L. Han, S. Park, S. Kang and H. P. In, "An adaptive cross-layer FEC mechanism for video transmission over 802.11 WLANs", *Proceedings of the First International Conference on Internet*, pp. 209-215, 2009.

[23] Cheng-Han Lin, Ce-Kuen Shieh and Wen-Shyang Hwang, "An Access point-based FEC mechanism for video transmission over wireless LANs", *IEEE Transactions on Multimedia*, Vol. 15 No. 1, pp. 195-206, 2013.

[24] Siu-Ping Chan, Chi-Wah Kok and Albert K. Wong, "Multimedia Streaming Gateway With Jitter Detection", *IEEE Transactions on Multimedia*, Vol. 7, No. 3, pp. 585-592, 2005.

[25] Cheng-Han Lin, Chih-Heng Ke, Ce-Kuen Shieh and Naveen K Chilamkurti, "The Packet Loss Effect on MPEG Video Transmission in Wireless Networks", *Proceedings of the 20th International Conference on Advanced Information Networking and Applications*, Vol. 1, pp. 565-572, 2006.

[26] Jirka Klaue, "EvalVid - A Video Quality Evaluation Tool-set", Available at: www.tkn.tu-berlin.de/research/evalvid

[27] "NS-2 Multimedia Communication tool", Available at: http://140.116.72.80/~smallko/ns2/MultimediaComm_en.html.

[28] Joan L. Mitchell, W. Pennebaker, Chad E. Fogg and Didier J. Legall, "*MPEG Video: Compression Standard*", Chapman and Hall, 1996.

[29] C. H. Lin, C. H. Ke and C. K. Shieh, "A study of MPEG Video Transmission in Lossy Wireless Network", *Workshop on Wireless, Ad Hoc and Sensor Networks*, pp. 253-258, 2005.

[30] X. Jin and G. Min, "Modelling and analysis of priority queueing systems with multi-class self-similar network traffic: a novel and efficient queue decomposition approach", *IEEE Transactions on Communi*cation, Vol. 57, No. 5, pp. 1444–1452, 2009.

[31] M. Usha and R. S. D. Wahida Banu "Towards a simple service class for distributed Multimedia Applications in next generation IP networks", *Proceedings of International Conference on Parallel and Distributed Computing and Systems*, pp. 161-166, 2005.

[32] M. Usha, "Enhanced Quality of Service through Jitter Control for Next Generation Service Computing", *Proceedings of National Conference on Advanced Computing*, 2006.

[33] Sanjeev Mehrotra, Jin Li and Ying-Zong Huang, "Optimizing FEC Transmission Strategy for Minimizing Delay in Lossless Sequential Streaming", *IEEE Transactions on Multimedia*, Vol. 13, No. 5, pp. 1-23, 2011.

[34] ZHAO Min-cheng, GONG Xiang-yang, QUE Xi-rong, WANG Wen-dong and CHENG Shi-duan, "Context-aware adaptive active queue management mechanism for improving video transmission over IEEE 802.11E WLAN", *The Journal of China Universities of Posts and Telecommunications*, Vol. 19, No. 2, pp. 65-72, 2012.

[35] M. Usha and K. Madhana, "Differentiated QoS Framework for Video Transmission over Wireless LANs Using Push-out Policy", *IEEE International Advance Computing Conference*, pp. 323-328, 2014.