International Journal of Engineering & Technology, 7 (3.34) (2018) 97-102

International Journal of Engineering & Technology

Website: www.sciencepubco.com/index.php/IJET

Research paper

Priority based Proactive Buffer Management for Aggregated Streaming in Heterogeneous Wireless Networks

S. Duraimurugan[1]*, P. Jesu Jayarin[2]

[1]*Research scholar, Department of Computer Science and Engineering, Satyabhama Institute of Science and Technology, Department of Information Technology, St. Joseph's College of Engineering, OMR, Chennai-119, Tamilnadu, India.*
[2]*Associate Professor, Department of Computer Science and Engineering, Jeppiaar Engineering College, OMR, Chennai-119,Tamilnadu, India.*
**Corresponding author E-mail: duraimuruganceg@gmail.com*

Abstract

Now a days, most of the data is transmitted between two devices using Wireless network. Although this has made it easier to transmit data, it also has its disadvantages. One of its disadvantage being the irregular data transfer due to the presence of congestion in the intermediate nodes. This paper proposes an idea to overcome the said disadvantage by developing a priority based proactive buffer management for the wireless network through the use of buffer which will be maintained at the source. The buffer is used to regulate the transmission rate of the data packets in accordance with the congestion level which is present in the transmission path. For faster transmission, the data packets having high priority are transmitted while the data packets having lower priority are dropped thereby reducing the quantity of data packets being transmitted, hence overcoming the problem of congestion.

Keywords: *Buffer management; Congestion Control; Network coverage map server; Priority based buffer; Source buffer management.*

1. Introduction

1.1. WLAN

A wireless local area network (WLAN) is a wireless system which performs data transmission through wireless medium and is based on the infrared or radio frequency functioning [1]. In WLAN, many systems are connected wirelessly. In WLAN, operations are performed conveniently since there is no requirement of linking the users physically through wires. In WLAN, the basis technique used is the spread spectrum modulation, which allows mobile as well as wireless communication [2].

1.2. Video Streaming in WLAN

In WLAN, when video traffic demands more bandwidth than the available bandwidth, then the network gets overloaded. This leads to increased delay, jitter as well as reduced throughput. This degrades the video quality of the user present at the receiving side, thus creating bad user experience [3]. There are some basic problems faced in the WLAN. At a wireless station, the average service rate is based on the congestion level of the channel, which is in turn dependent on the number of active nodes as well as the load at each node. Also, at an active station, the packet inter service time changes irregularly because of the irregular operating characteristics of the CSMA/CA operation, even when if the network load is maintained same. Due to these reasons, the multiplexing and buffer backlog behavior gets influenced, which also influences the selection of the buffer sizes [4].

1.3. Congestion Control

In video related applications, the video quality gets degraded when the transmission of data occurs in the form of bursts and also due to path loss. This occurs mainly due to network congestion. So, in order to reduce packet loss and delay, the TCP congestion control technique has to be used. One of the congestion control technique is the rate control technique. In this technique, the bandwidth needed for video transmission is compared with the currently available bandwidth. For various applications, various kinds of rate adaptive video encoding mechanisms are available. These rate adaptive schemes improve the quality of the video at around specific encoding rate [5].

1.4. Prioritized Data Buffering in WLAN

Buffers perform an important task in wireless networks. Buffers usually store data packets for small time intervals [6]. These buffers are very useful since it controls the data packet movement into the network. The buffers usually let out the data packets into the network in a queue fashion, where the data packets enqueue based on its order of arrival and gets transmitted one by one into the network. But in case when the buffer is also loaded due to high congestion in the network, it makes the buffer to drop out most of the arriving data packets. This can result in high inconvenience since some of the high priority data packets which arrived in later point of time may get dropped out. To overcome this issue, priority based buffer management technique is used. In this technique, when a data packet arrives into the network, its priority is estimated. If the priority of the incoming data packet is higher than the priority of the data packet existing in the buffer, then the arriving higher priority data packet is accepted and lowest priority data

Copyright © 2018 Authors. This is an open access article distributed under the Creative Commons Attribution License, which permits unrestricted use, distribution, and reproduction in any medium, provided the original work is properly cited.

among the existing data is dropped out. If the arriving data packet is of lower priority, then it is dropped out.

2. Related Work

Mohamed M. Abo Ghazala et al [7] have presented a Performance Evaluation of Multimedia Streams over Wireless Computer Networks (WLANs). The performance of the WLAN is examined by using methodical steps in streaming video data packets between WLAN systems when the number of devices in the overall network is larger. Few instances are given. Several network features are considered like end to end delay, network traffic, number of packets dropped, delay, load, throughput, etc. This paper mainly focuses on the problems related to scalability and efficiency, in cases when the number of devices are higher.

Yi-hua Zhu et al [8] have proposed Access Point Buffer Management for Power Saving in IEEE 802.11 WLANs. An AP-priority TPM technique is presented and experimented w.r.t a realistic model which consists of two queues; one containing the incoming frame and the other handling the outgoing frames. The proposed PBMS is capable of managing several ARs, with higher power usage but with lower probability in packet drop rate, which is attained by regularly varying the value of the involved timers.

Gabriel Scalosub et al [9] have proposed the Buffer Management for Aggregated Streaming Data with Packet Dependencies. In this paper, the buffer overflow issue is discussed when the network includes several streams which are dependent on one another. Algorithms to overcome this issue is described and the performance of one algorithm is demonstrated w.r.t both worst condition competitive scenario as well as based on simulation observations. This paper gives proofs that this approach has bounded competitive ratio and hence ensures successful performance at any traffic situations. It is also shown that for any algorithm, the competitive ratio will reduce linearly w.r.t the number of streams functioning in the network

Varun Singh et al [10] have proposed a Predictive Buffering for Streaming Video in 3G Networks. In this paper, the client is made capable of determining the limitations of the network. The client achieves it by taking help from the network coverage map service. The network coverage map service collects information related to coverage as well as capacity. The information present at the network coverage map service are given to the client and also to the server, thus allowing the client to pre-buffer or change the streaming rate. The received information from the NCMS can be used for predictive rate switching as well as pre buffering. It also aids in overcoming complete packet loss.

Anand Balan et al [11] have proposed an Integrated Buffer Management and Congestion Control for Video Streaming. In this paper, the congestion issue is overcome by using an integrated buffer which manages the dropping of packets at the source based on the congestion level in the network. This is the basis for the proposed buffer management technique. To control the encoder which operates on the basis of the motion compensated wavelet decomposition, a packet filter is introduced. Based on certain analysis, it is shown that packet loss issue and FEC need can be decreased by following the dispersive packetization technique and error concealment technique. By using the integrated buffer along with the robust technique introduced, the network performance can be enhanced.

3. Priority based Proactive Buffer Management for Aggregated Streaming

In our previous paper [12], a proposal to optimize the streaming of multimedia data and to control the congestion was made and demonstrated successfully for heterogeneous wireless network. The current work's framework is built on two-tier integrated heterogeneous wireless networks. Fuzzy Logic congestion (FLC) controller is used to determine the data streaming rate. The congestion level is deduced using the congestion level determination unit and the rate at which the congestion level varies with respect to the gaps present in between each packet (Jitter rate) is provided as input to fuzzy logic congestion controller. The new data streaming rate is obtained as output from fuzzy logic congestion controller. The buffer levels are maintained appropriately with the support from the application that adapts to new streaming rate also the greedy streaming to threshold and stop (GSTS) algorithm is applied to optimize the threshold level for buffers, resulting in reduced streaming monetary cost. In wireless networks, sudden degradation of bandwidth in locations during subsequent movements of users cause overflow. In [1], a buffer management scheme for aggregated multiple streams with inter-packet dependencies is proposed. But it fails to detect the overflow or congestion status of the traffic for which the buffer management should be applied. As an extension to this work, we propose to design a priority based proactive buffer management for aggregated streaming. In this proposed scheme, an adaptive buffer at the sender is used and a video sequence is coded using a robust coding scheme that can adapt to the change in network rate [3]. The client estimates the congestion level as per our previous paper and then predicts coverage holes [2] (where the bandwidth suddenly drops below a threshold value). After detecting the coverage holes, it pre buffers the video from the source or else adaptively estimates the new streaming rate based on the current available rate at the coverage holes using fuzzy logic. The node speed, available throughput and time available to buffer are the inputs for the Fuzzy logic. If it detects congestion, the priority based buffer management scheme [1] is applied in which the packets with lower priorities are dropped. The Fig.1 and algorithm 1 both are explaining the overview of priority based proactive buffer management for aggregated streaming.

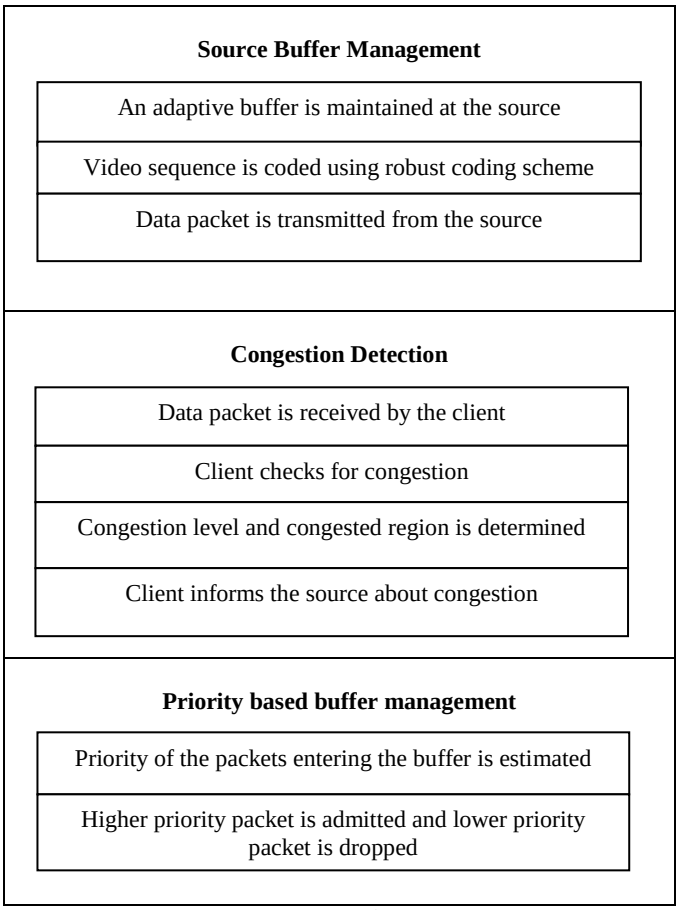


Fig.1: Overview of PPBMAS

Algorithm 1: Overview PPBMAS (PPBMAS ())
SBM_I () // Initial transmission of packets from the Buffer
If CN is high **then**
Loop Threshold bandwidth is not available **then**
Congestion region ()
SBM_C ()
End Loop
SBM_I ();
End

3.1. Source Buffer Management

In order for a data to be transferred from a device to the other through the use of wireless network, the data packets that are being transferred need to be transmitted through the intermediate nodes that will be present when the data gets transmitted from the source to the destination. The data packets will be forwarded from the source which is then received by the intermediate nodes and forwarded again continuously until the data packets reach its destination. Once the data packets that are being transferred are very large in number it leads to the source being overwhelmed by the data packets that it results in the congestion at the source. A good reference for this would be trying to basket more than one ball into a basket. This is due to the fact that, the number of data packets that are being transmitted is very large, when compared to the bandwidth of the channels that are available at the source. This results in the data packets getting accumulated at the source which in turn gradually leads to congestion resulting in the data packets being dropped at the source, meaning the data packets are not transmitted or forwarded to the destination channel. In order to overcome this issue of congestion at the source channel, a source buffer management scheme is proposed. According to which, the outgoing data packets that are ready to be transmitted or forwarded is stored in a buffer that is being maintained at the source. This in turn prevents the situation in which large number of data packets are being transferred simultaneously, which results in the prevention of congestion. In better terms this proposal avoids a situation in which the congestion occurs by transmitting the data packets in an ordered manner. A robust coding technique is being applied to the video packets to ensure that the video adapts to the changing conditions that is present in the network due to the ups and down that are present in the network speed or rate. The Algorithm 2 is used to make this possible in given below.

Notations:
1. Q_{size} :Maximum Queue size
2. $Buffer_{threshold}$:Maximum threshold value of the buffer
3. Q_{size} :Queue size
4. FR :Frame Rate

Algorithm 2: Source Buffer Management (SBM_I (): Initial State)
Q_{max_size} = Constant
$Buffer_{threshold}$=Constant
$Q_{size} = 0$
Loop packet are available **then**
Increment the Q_{size}
If $Q_{size} > Q_{max_size}$ then
Drop the packet
Else if $Q_{size} > Buffer_{threshold}$ **then**
P: Probability of (Drop the packet) **If** P is less **then**
Drop the packet
Else
Enqueue the packets
End Else
Enqueue () // $Q_{size} < Q_{max_size}$ and Qsize < $Buffer_{threshold}$
End Loop
int n=1;
n layer ← FR
Loop buffer is not null **then**
n+=1
FR=FR / 2
n layer ← FR
End Loop
End

A buffer is maintained at the source in order to avoid congestion by using the buffer to maintain a queue. This buffer stores the data packets that need to be transmitted or forwarded in the queue which follow the first in first out format using the time of arrival as the base. The maximum size of the queue (Q_{max_size}) and the threshold of the buffer ($Buffer_{threshold}$) are predefined at the source according to the size of the channel that is present at the source. The Buffer threshold needs to be less than the size of the queue since the buffer threshold acts as the warning line for the size of the data packets, while the size of the queue determines the full capacity of the queue. As a result of which, Once the size of the queue exceeds the buffer threshold the probability of the data packets being transmitted or forwarded to the client falls exponential. When a data packet that needs to be transmitted or forwarded arrives at the source, it is inserted into the queue by the buffer that is maintained at the source. Once the data packets that have arrived are added into the queue the size of the queue (Q_{size}) is determined by the buffer. The size of the queue is determined by using the number of data packets that is inserted into the queue as the reference. If the data packets that are being inserted exceed the maximum size of the queue then the data packets would be dropped. Mathematically the data packets would be dropped when Q_{size} is greater than Q_{max_size}. While when the number of data packets that are inserted into the queue by the buffer exceeds the buffer threshold, then there is 'p' probability that the data packets would be dropped. As a result of which the data packets are only inserted into the queue successfully if the current size of the queue is less than the buffer threshold. Meaning the arriving data packets has the fulfill the condition of having its size being less than the maximum size of the queue in order for the data packet to be inserted into the queue successfully by the buffer maintained at the source. Only the data packets that are present in the queue maintained by the buffer will be transmitted or forwarded by the source to the client side. Once the data packet is successfully inserted into the queue by the buffer that is maintained at the source, the full frame rate of the packet is considered as the first layer. The data packet is send to the client from the server in the form of layers, each layer having the varying number of data packets that is determined by the frame rate of the data packets. Once the first layer is sent to the client, the second layer is formed by decomposing the neighboring frames using the filter bank to produce a temporal frequency band and temporal high frequency band. Next we need to send the next set of data packets. For this the high temporal frames that are present in the second layer is sub band coded and transmitted using the motion vectors at the source or server side. Now the low temporal band that is present in the second layer is half of the full frame rate. This half is further decomposed by half to get one-fourth of the full frame rate which is then transmitted or forwarded by the source in the form of layers, where each layer is packetized independently to prevent conflicts that may occur during the transmission process. A good example of this would be streaming that is used in most of the video hosting websites that is available in the internet where the video that needs to be played is broken into bits of data which is packet to form data packet and transmitted or forwarded to the client where it get unpacked to get the original video before the breaking up of the video into data packets. In each one of the layers that is being transmitted or forwarded the data is coded and packetized in a dispersive format to allow the sub band data from similar band appear in various packets. Meaning, a small of the data is stored in neighboring packets so that any packets that is dropped can be identified at the client side, for which we need the data to be in dispersive format meaning in a form in which it is in small bits which are put together to

form data packet. This enables us to reconstruct the higher layers using the lower layers as the base. This also helps us in the detection of dropped packets using the neighboring data as the reference, So that if any packets are dropped, the client side can interact with the source to request the source to resend the required data packets. This way we would be able to recover any data packets that have been dropped due to the occurrence of congestion in the transmission or forwarding path between the client and the source server. Using the below mentioned Algorithm we are able to avoid congestions and situations in which congestion occurs during the transmission or forwarding of data packets using the buffer management technique. The Algorithm also aids in the detection of the data packets dropped.

3.2. Determination of the Congested Regions

Even though the data packets are transmitted or forwarded from the source channel via the use of buffer management system, there exists a risk of congestion occurring while the data packets are travelling through the intermediate nodes. For reference, imagine a scenario where pipes are interconnected. When the small stones are inserted into the pipes to reach the destination side, there may be a situation in which some stones come together to block the pipe thereby preventing the other stones from reaching the destination, resulting in congestion. Since it is not possible to apply buffer management system in each and every one of the intermediate nodes, we need to predict the congestion which will be present in each node and estimate the coverage hole. In order to predict the congestion occurrence in any of the intermediate nodes, a new streaming rate is computed using fuzzy logic. The Algorithm 3 that is required to solve the congestion occurrence in the intermediate nodes is given below. Once the data packets that were transmitted or forwarded from the source is received by the client, it will be made possible for the client to estimate the congestion level that is present on the transmission or forwarding path used by the data packets as described in the previous paper. The equation to denote the congestion in the transmission and forwarding path is given by

$$C_N = \frac{B_S - B_C}{B_S} = 1 - \frac{K^S_{aj}}{K^C_{aj}} \quad (1)$$

Once the value of congestion level exceeds the value of threshold for congestion level it denotes the presence of congestion in the transmission or forwarding path. While if the value of congestion is less than the threshold hold value of congestion then it denotes the absence of congestion in the transmission or forwarding path. After the congestion is determined to be present in the transmission or forwarding path, the client sends a 'Location Request' to the source which includes the information of the congestion location, transmission or forwarding path and thee average transmission rate of data packets to the Network coverage map server in order to determine the region where the congestion is present in the transmission or forwarding path that the data packets travel. On receiving the request from the client, the Network coverage map server samples the neighboring locations with respect to the transmission rate. Thus the Network coverage map maintains a throughput vector for the location that is near that area. Then the NCMS compares bandwidth region of the current region with respect to the threshold of the bandwidth, this comparison is repeated in order to check the regions near the transmission or forwarding path for the presence of congestion. As soon as the NCMS detects a region to have bandwidth of the said region less than the threshold of the bandwidth, it determines that particular region to be congested that is the congestion hole is determined. After the coverage holes that are present in the transmission or forwarding path and in its nearby region have been determined, it is possible for the data packets to be transferred in alternate path or else the streaming rate of the data packets can be adjusted that the congestion is absent. A good reference for this would be reducing the amount of stones that are sent through a pipe to reduce a situation in which the stones gather together in a particular path which results in the presence of congestion. The new streaming rate of the data packets that are transmitted or forwarded is determined using fuzzy logic as determined in the previous paper. In order for the streaming rate to be estimated using fuzzy logic the node speed, available throughput and time available to the buffer in provided as the input for the said logic. From the Algorithm 3, we can determine the congested regions and the congestion level in each region. The problem that we are currently facing due to the adverse effect of congestion on data transmission and forwarding can be overcome through the use of new streaming rate which is estimated through the use of fuzzy logic. The streaming rates for various scenarios that may occur resulting in a particular value for node speed , Available throughput, time available for buffering are shown in a detailed manner in the below table 1

Table.1: Fuzzy output for various fuzzy input conditions

Node Speed	Available Throughput	Time available for buffering	New Streaming Rate
VH	VH	VH	VH
H	H	H	H
M	H	M	H
M	M	M	M
L	H	H	M
L	M	M	M
M	H	M	M
L	L	L	L
M	L	M	L
H	L	L	L
VL	VL	VL	VL
H	L	M	VL
H	L	L	VL
M	M	L	VL

Notation	Explanation
H	High
L	Low
M	Medium
VL	Very Low
VH	Very High

Notations:

1. C_N: Congestion Level
2. B_C :receiving rate of the data packet by the client
3. B_S :transmission rate of the data packet into the network from source
4. K^S_{aj} :frequency weighted average transfer time for the source
5. K^C_{aj} :frequency weighted average transfer time for the client
6. $C_{Threshold}$: Threshold value for congestion level
7. *NCMS:* Network Coverage Map Server
8. $BS_{regionb}$: Bandwidth of the current region
9. $BE_{Threshold}$: Threshold value for bandwidth

Algorithm 3: Determination of Congestion region (Congestion region ())

BS ←Number of packets transmitted from source
BC ←Number of packets received by client
CN = (BS – BC) / BS: CN ←Congestion Level
$C_{Threshold}$ = Constant
If CN < $C_{Threshold}$ **then**
Return // No Congestion
End if
// CN >= $C_{Threshold}$
Bandwidth $_{threshold}$ = Constant

NCMS ←Location Request (**Transmission Path, Location, Average_Transmission_rate**)
Loop Location is not Source **then**
// NCMS Compare Current Location
If Bandwidth $_{region}$ < Bandwidth $_{threshold}$ **then**
Coverage hole is detected
Record Coverage hole
End if
Location ←Move towards source
End Loop
Return Coverage hole
Estimate new streaming rate with fuzzy logic
End

3.3. Priority Based Buffer Management Scheme

After the client receives the data from the source, the client need to determine the congestion level that is present in the transmission of forwarding path. The previous Algorithm is used in order to determine the congestion level that is present in the transmission or forwarding path. This data needs to be given to the source in order for the source to apply the priority based management scheme. Once the source receives the data on congestion level from the client in the form of acknowledgement message, the source implements the priority based buffer management scheme. The acknowledgement message denotes the presence of congestion in the path between the source and the destination. The Algorithm that is required to implement the priority based buffer management scheme is given below.
If f_i that is present inside of A_t (f_i) has all the packets that need to be transmitted or forwarded without have any one of the packets that were present from being dropped during transmission or forwarding and if R_t (f_i) has a lower priority in comparison with A_t (f_i) then D_t (f_i) that is present in the buffer that is maintained at the source is pre-empted from it while A_t (f_i) is admitted into the buffer. At the same time if f_i that is present inside of A_t (f_i) has some of its packets dropped from the buffer and if R_t (f_i) has a lower priority in comparison to A_t (f_i) present in the buffer then A_t (f_i) that is present in the buffer or inserted into the queue present at the source is dropped along with $f_{i\ which}$ has some of its packets dropped from the buffer will have all of its data packets dropped by the buffer. By using the below mentioned Algorithm we are able to handle the congestion by reducing the load at the buffer. As a result during a congestion situation only the higher priority frames are transmitted while the lower priority frames are dropped. Using this method the congestion in the network is controlled by the use of priority based buffer management technique.

Notations:
1. f_i : Frame containing a set of packets
2. $A_t(f_i)$: set of packets of frame, f_i arriving into buffer at time t
3. $R_t(f_i)$: set of packets of frame, f_i residing in buffer when $A_t(f_i)$ arrives
4. $L_t(f_i)$: set of packets with priority lower than f_i residing in the buffer when considered with $A_t(f_i)$
5. $D_t(f_i)$:belongs to $L_t(f_i)$ and is the minimal size prefix of $L_t(f_i)$ which yield to $A_t(f_i)$

Algorithm 4: Source Buffer Management - After Coverage hole (SBM_C ())
Calculated new Streaming rate based on **Fuzzy Logic**
If A_t (f_i) ←none of the packets dropped and R_T (f_i) is low priority **then**
Dequeue (R_t (f_i)**)**
Enqueue (A_t (f_i)**)**
Enqueue (R_t (f_i)**)**
Else if at (f_i) ←some of the packets dropped **then**
Drop (A_t (f_i)**)**
Else
Enqueue (A_t (f_i)**)**

4. Simulation Results

4.1. Simulation Parameters

We use NS2 [12] to simulate our proposed Priority based Proactive Buffer Management for Aggregated Streaming (PPBMAS) protocol. We use the IEEE 802.11 for wireless LAN 3-G networks as the MAC layer protocol. It has the functionality to notify the network layer about link breakage. In our simulation, the packet sending rate is varied as 150,200,250,300 and 350Kb Similarly the Cache Size is varied as 300,350,400 and 450bytes. The area size is 600 meter x 600 meter square region for 50 seconds simulation time. The simulated traffic is Constant Bit Rate (CBR). Our simulation settings and parameters are summarized in table 2

Table 2: Simulation parameters

No. of Mobile Nodes	10
Area	600 X 600
MAC	802.11
Simulation Time	50 sec
Traffic Source	CBR
Rate	150,200,250,300 and 350Kb
Propagation	TwoRayGround
Antenna	OmniAntenna
Cache Size	300,350,400 and 450 bytes

4.2 Performance Metrics

We evaluate performance of the new protocol mainly according to the following parameters. We compare the BMAS [9] protocol with our proposed PPBMAS protocol. The parameters are delivery ratio, bandwidth, delay and packet drop

4.3 Results & Analysis

The simulation results are presented in the next section.

A. Based on Rate

In our first experiment we are varying the rate as 150,200,250,300 and 350Kb for CBR traffic.

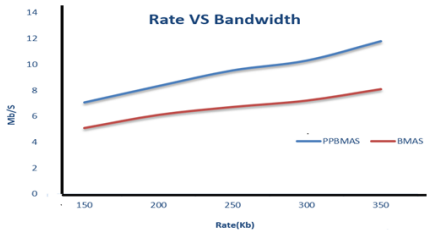


Fig 2: Rate Vs Bandwidth

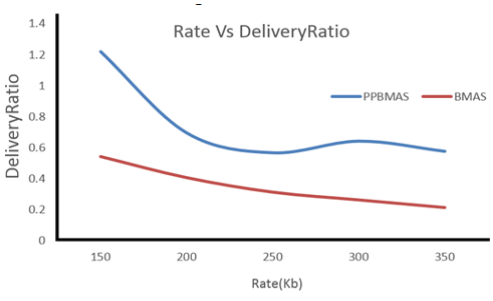


Fig 3: Rate Vs Delivery Ratio

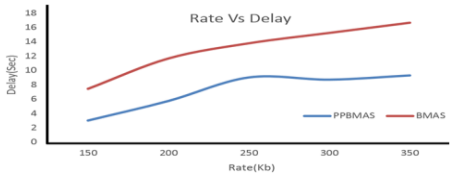


Fig 4: Rate Vs Delay

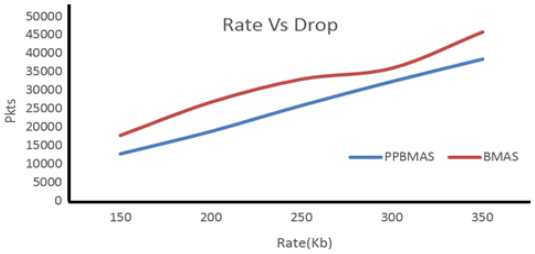


Fig 5: Rate Vs Drop

B. Based on Cache Size

In our second expirement, we vary the cache size as 300,350. 400 and 450 bytes**.**

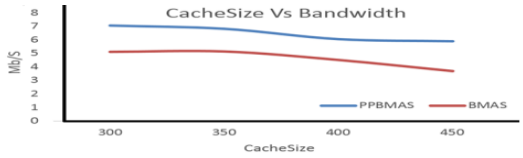


Fig 6: Cache size Vs Bamdwidth

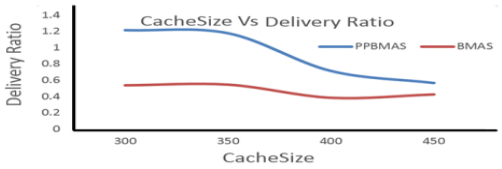


Fig 7: Cache size Vs Delevery ratio

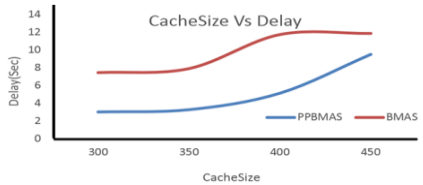


Fig 8: Cache size Vs Delay

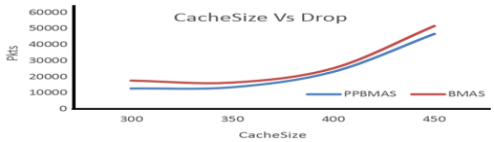


Fig 9: Cache size Vs Drop

Figures 6 to 9 show the results of received bandwidth, delivery ratio, delay and drop by varying the Cache Size from 300 to 450 for the CBR traffic in PPBMAS and BMAS protocols. When comparing the performance of the two protocols, we infer that PPBMAS outperforms BMAS by 29% in terms of received bandwidth, 45% in terms of delivery ratio, 48% in terms of delay and 16% in terms of drop.

5. Conclusion

A Priority based Proactive Buffer Management for Aggregated Streaming in heterogeneous wireless Networks is proposed and discussed in this paper. Initially, a source buffer management scheme is proposed to avoid congestion at the source. This buffer maintains a queue of data packets to be transmitted, and transmits them one by one towards the client. On receiving the data packet, the client estimates the congestion level to check if congestion was observed at any region during data packet transmission. If the congestion was observed, then it determines the congested region and also the new streaming rate accordingly. Then the source is informed about the observed congestion. Hence the source performs priority based buffer management, which estimates the priority of each arriving packet, and only the higher priority packets are admitted whereas the lower priority packets are dropped. In this way, the network is kept safe and data transmission is carried out in an uncongested way.

References

[1] Rafidah Abdul Hamid, “Wireless LAN: Security Issues and solutions”, © SANS Institute 2003.

[2] Jashanpreet Kaur and Amoldeep kaur, “Evaluating Data drop in Wireless LAN using TTL and Fragmentation”, International Journal of Engineering Research and Applications (IJERA) ISSN: 2248-9622 www.ijera.com Vol. 2, Issue 5, September- October 2012, pp.585-588

[3] Ramesh Goel and Mahasweta Sarkar, “Enhancing QoS for Streaming Video over WLANs”, Proceedings of the World Congress on Engineering and Computer Science 2008, WCECS 2008, October 22 - 24, 2008, San Francisco, USA

[4] [4] Tianji Li, and Douglas J. Leith, “Buffer Sizing for TCP Flows in 802.11e WLANs”, this work is supported by Science Foundation Ireland Grant 03/IN3/I396.

[5] [5] Aqsa Malik, Junaid Qadir, Basharat Ahmad, Kok-Lim Alvin Yau, and Ubaid Ullah, “QoS in IEEE 802.11-based Wireless Networks: A Contemporary Survey”, arXiv: 1411.2852v1 [cs.NI] 11 Nov 2014.

[6] [6] Depavath Harinath, “Buffers in 802.11-Based Networks”, International Journal of Advanced Research in Computer Science and Software Engineering, Volume 2, Issue 12, December 2012, ISSN: 2277 128X

[7] [7] Mohamed M. Abo Ghazala, Mohamed F. Zaghloul, and Mohammed Zahra, “Performance Evaluation of Multimedia Streams Over Wireless Computer Networks (WLANs)”, International Journal of Advanced Science and Technology Volume 13, December, 2009

[8] [8] Yi-hua Zhu, Han-cheng Lu, and Victor C. M. Leung, “Access Point Buffer Management for Power Saving in IEEE 802.11 WLANs”, IEEE Transactions On Network And Service Management, Vol. 9, No. 4, December 2012

[9] Gabriel Scalosub, Peter Marbach, and Jorg Liebeherr, “Buffer Management for Aggregated Streaming Data with Packet Dependencies”, IEEE Transactions on Parallel and Distributed Systems, Vol. 24, No. 3, March 2013

[10] Varun Singh, Jorg Ott and Igor D. D. Curcio, “Predictive Buffering for Streaming Video in 3G Networks”, 978-1-4673-1239-4/12$31.00 c_2012 IEEE

[11] Anand Balan, Omesh Tickoo, Ivan Bajic, Shivkumar Kalyanaraman, John Woods, “Integrated Buffer Management and Congestion Control for Video Streaming”, This work was supported in part by the ARO grant number 19-00-1-0559.

[12] S.Duraimurugan and P.Jesu Jayarin,” Optimized Multimedia Streaming and Congestion Control for WLAN-3G Networks”, Research Journal of Applied Sciences, Engineering and Technology, 2015.

[13] K. Vijayakumar and C. Arun, “Continuous Security Assessment of Applications in Cloud Environment”, International Journal of Control Theory and Applications, ISSN: 0974-5645 volume No. 9(36), Sep 2016, Page No. 533-541.

[14] G. Indrajith and K.Vijayakumar, “Automatic Mathematical and Chronological Prediction in Smartphone Keyboard” International Journal of Engineering and Computer Science ISSN: 2319-7242Volume 5 Issue 5 May 2016, Page No. 16714-16718.