Indonesian Journal of Electrical Engineering and Computer Science
Vol. 15, No. 3, September 2019, pp. 1535~1543
ISSN: 2502-4752, DOI: 10.11591/ijeecs.v15.i3.pp1535-1543

Study on real-time media congestion avoidance technique for video streaming over wireless local area network

Govindarajan J, Mohanapriya C
Department of Computer Science and Engineering, Amrita School of Engineering, Amrita Vishwa Vidyapeetham, India

Article Info

Article history:

Received Jun 18, 2018
Revised Aug 24, 2018
Accepted Apr 11, 2019

Keywords:

Congestion
RMCAT
TFRC
Video streaming
WLAN

ABSTRACT

The video streaming is one of the important application which consumes more bandwidth compared to non-real-time traffic. Most of the existing video transmissions are either using UDP or RTP over UDP. Since these protocols are not designed with congestion control, they affect the performance of peer video transmissions and the non-real-time applications. Like TFRC, Real-Time Media Congestion Avoidance (RMCAT) is one of the recently proposed frameworks to provide congestion control for real-time applications. Since the need for video transmission is increasing over the wireless LAN, in this paper the performance of the protocol was studied over WLAN with different network conditions. From the detailed study, we observed that RMCAT considers the packet losses due to the distance and channel conditions as congestion loss, and hence it reduced the sending rate thereby it affected the video transmission.

Corresponding Author:

Govindarajan J,
Department of Computer Science and Engineering,
Amrita School of Engineering,
Amrita Vishwa Vidyapeetham, Coimbatore, India.
Email: j_govindarajan@cb.amrita.edu

1. INTRODUCTION

TCP (Transmission Control Protocol) is widely used for assuring the reliable data transfer in communication between the endpoints. It helps the sender to deliver a large number of packets by using maximum available bandwidth while controlling the congestion over the network. In contrast to TCP, UDP (User Datagram Protocol) does not provide the reliable delivery of packets to the endpoints. Due to the absence of congestion control, UDP fails to control the packet losses over the highly congested network. When both the TCP and UDP protocols are running over the same network, the uncontrolled traffic from a UDP based real-time application affects the applications that use the TCP and also other UDP based applications.

TFRC (TCP-Friendly Rate Control) [1] protocol is a congestion control mechanism for real-time applications. The protocol was designed to provide the reliability to the real-time applications and also to avoid the problems due to the uncontrolled traffic from UDP based applications. Even though TFRC provides congestion control, it is suitable for streaming media which requires the smooth sending rate. However, in the recent days the applications are running variable encoding along with video streaming and hence they require the variable sending rate. UTFRC (Utility-driven TCP-Friendly Rate Control) [2-3] is an extended version of TFRC which considers the media requirement along with congestion control and hence it is suitable for streaming the video in best-effort networks than TFRC.

Like UDP, Real-time Transfer Protocol (RTP) [4] is not designed with congestion control procedure. In general video streaming requires more bandwidth compared to non-real-time flows. In the absence of higher bandwidth, the buffer at the sender will start to grow and at the same time buffer at receiver will be filled at the low data rate. Also, the packet losses due to congestion inside the network affect the delivery of video frames. To improve the performance of video streaming RMCAT (Real-time Media Congestion Avoidance Technique) [5] has been proposed recently.

In general, the protocol which was designed without congestion control or only with congestion control had failed over the wireless network. For example, TCP which was designed during wired network era experiences the performance degradation due to various characteristics of the wireless network. Also, ad-hoc wireless network introduces additional problems like packet reordering and unfairness [6-7]. Various research works are carried out recently to improve and to analyze the performance of applications over wired and wireless networks [8-10, 14-15]. In addition to these problems, the mobility of the nodes increases the complexity of wireless network design. Hence recently many research works are being carried out to reduce the signaling overhead [11].

The main aim of this paper is to study the performance of RMCAT over wireless LAN. The rest of paper is organized as follows: Section 2 summarizes the operations of RMCAT. Section 3 describes the need for video streaming over WLAN and the recent WLAN standard. Section 4 elaborates our experimental design and performance analysis of RMACT. Finally, Section 5 summaries our study and the future directions.

2. RMCAT

RMCAT [5] has defined a framework to provide the congestion control for the real-time traffic. The framework consists of five functional modules namely Network congestion controller, Transmission queue, Rate controller, Network probe generator and a live video encoder. RMCAT working group has designed a rate based congestion controller called as NADA (Network-Assisted dynamic adaption) [12]. As per the specification of NADA, the sending rate of the real-time flow is controlled using the feedback from the endpoints or using ECN (Explicit congestion notification) from the network routers. Like TFRC, the NADA calculates the sending rate using the feedback from the receiver. In comparison to TFRC, NADA considers queuing delays in addition to the packet loss for the calculation of sending rate.

The receiver continuously estimates the end-to-end delay and packet loss from the received packets, and periodically reports to the sender in terms of aggregated congestion signal. It also measures the receiving rate of the packets from the received packets and communicates to the sender. The receiving rate is calculated by dividing the observation window from the total size of the packets received during this window. The default observation window is 500ms. The packet loss is detected using the sequence number gap in the received packets. The packets which are arriving out-of-order are discarded and its count has been included in losses.

The sender side operation is to calculate the reference rate from the received feedback reports. Then, using the calculated reference rate it adjusts the target rate of live video streaming encoder and sending rate of packets over the network. Instead of transferring the encoded data to the network for transmission directly, it will be stored them temporarily in rate shaping buffer before the transmission. To avoid the long waiting time inside the buffer due to a mismatch between the target encoding rate and sending rate, the sending rate will be increased and at the same time, target encoding rate will be reduced. i.e., whenever the encoding rate is higher than the sending rate, buffers start to grow. Now, the increased sending rate and decreased encoding rate will increase the depletion rate of the buffer. To respond to the congestion and delay inside the network, each feedback message from the receiver supplies three variables namely receiving rate, aggregated congestion signal, recommended rate adaptation mode to NADA congestion control at the sender. The total size of these three variables in a feedback message is 48bits. The recommended interval for feedback message is 100ms.

3. VIDEO STREAMING OVER WLAN

Wireless LAN (WLAN) was designed to provide the wireless connectivity to the nodes which are in the transmission range. The basic WLAN standards provided the maximum rate of 10Mbps (IEEE 802.11b) and hence WLAN was used only to provide basic services like web browsing, file transfer. IEEE 802.11g was designed to provide the maximum data rate of 54Mpbs at 2.4GHz. Later, the design of WLAN is extended to provide the data rate of 100Mbps (IEEE 802.11n). However, after the introduction of gigabit WLAN (IEEE 802.11ac), the basic service is extended to transmission of multimedia content over the WLAN. There are many use cases for the multimedia content over WLAN. One of the known use cases is streaming HD videos to serval receivers inside the home network. Video conferencing, delivery of online video lectures and in-room gaming are the few other use cases.

In the IEEE 802.11 family, IEEE 802.11ac is called as Gigabit Wi-Fi. The design of 802.11ac provides the streaming of HD videos for multiple users simultaneously and also reduces the time to back up the large files. In comparison to the earlier standards for WLAN, it provides the 80MHz channel bandwidth which can be extended to 160MHz. It uses 256-QAM for encoding compared to 64-QAM in 802.11n. The maximum number of spatial streams is eight in comparison to four spatial streams in 802.11n. Along with multi-user MIMO, it supports beamforming. 802.11ac defines MCS (Modulation and Coding Scheme) from 0 to 9. In 802.11ac the channel width varies from 20 MHz to 160MHz (20, 40, 80, 160 MHz) and modulation varies from low order encoding of BPSK to higher order encoding of 256QAM. The 802.11ac uses long guard interval of 800nsec and a short guard interval of 400nsec.

Even though the 802.11ac theoretically achieves 1000Mbps and higher data rate using higher channel bandwidth, higher order encoding and increased number of spatial streams, due to varying network and channel conditions, the actual data rate achieved will be lesser than the theoretical values [13]. For example, increasing the distance between the nodes, the interference will affect the performance of 802.11ac.

4. EXPERIMENTAL DESIGN AND PERFORMANCE ANALYSIS

To study the performance of RMCAT over the 802.11ac the following scenarios are considered. The simulations were carried out in ns3 simulator. YansErrorRate Model is used in all simulations to simulate the bit error inside the network. We have considered a use case of streaming of online video lecture to all participants. To map this scenario to the simulation, a topology consists of one Access Point (AP) with 5 station nodes is simulated. AP transmits a video to all stations. Two scenarios are considered to reflect the floor plans with different station positions.

The distance of 5m between the nodes is considered to simulate the best-case scenario of near-zero packet loss. In contrast distance of 30m is considered to simulate the worst case of the scenario of more number of packet losses. Both fixed resolution and variable resolutions are considered. In the fixed resolution, all frames are transmitted with a fixed resolution. In contrast, in variable resolution, the resolution of each frame varies according to target encoding rate and bits per pixel (bpp). The Table 1 summarizes the various parameters and the values considered in our simulations.

Table 1. Configuration Parameters

Parameter	Values
MCS	4
Channel Width	20MHz, 80MHz
Guard Interval	Short guard interval
Error Model	YansErrorRatemodel
Number of Nodes	5
The distance between the Nodes	5m, 30m
Packet size	500bytes, 1000bytes
Maximum rate of video transmission	1500Kbps, 4000Kbps
Resolution of video	Fixed (360p and 720p) and Variable Resolutions

For both scenarios, an extensive set of experiments are conducted. For each experiment the distance between the nodes, resolution (fixed/variable), packet size and, RMCAT min and maximum transmission range are varied. To study the impact of various network and channel conditions over the performance of video transmission, sending rate and receiving rate calculated by the sender are considered as the metrics for the analysis. In addition, the resolution of the video is also considered as one of the metrics. For every case, the calculated sending rate and receiving rate are plotted with a bitrate of the video. For experiments with variable encoding, the resolutions of the frames are plotted.

4.1. Performance Analysis Over WLAN with a Floor Plan with Nodes are Positioned in a Grid Topology

Figure 1 presents the floor plan with nodes are positioned at the distance of 5m in a grid topology. Similarly, Figure 2 presents the floor plan nodes are position at the distance of 30m in a grid topology. The x and y-coordinates of the grid points are shown in Figures 1 and 2.

Initially, the experiment is conducted with a distance of 5m in Figure 1 and fixed encoding of 360p. The packet size and maximum rate are configured as 1500bytes and 4000kbps respectively. The results of the experiments are presented in Figure 3. Due to near-zero packet losses, the calculated sending rate at the

RMCAT sender has reached the maximum configured rate of 1500Kbps and the receiving rate of the video is same as the bit rate of the video. i.e., all five stations have received the video at the same rate.

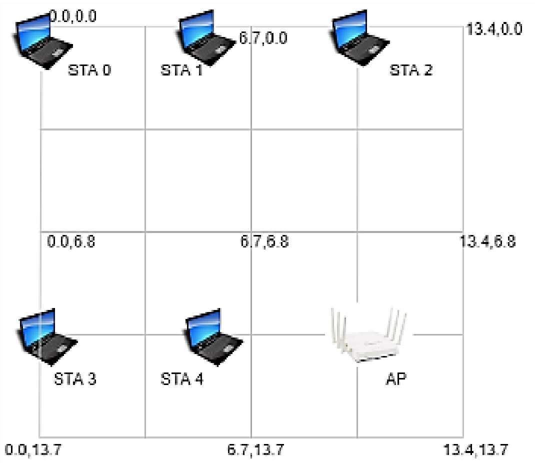


Figure 1. Distance between the nodes is 5m

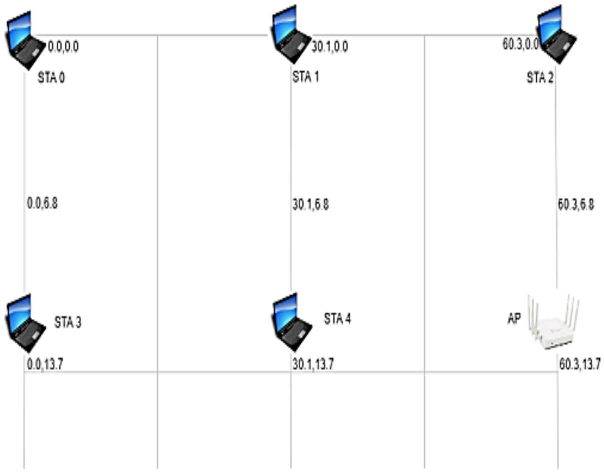
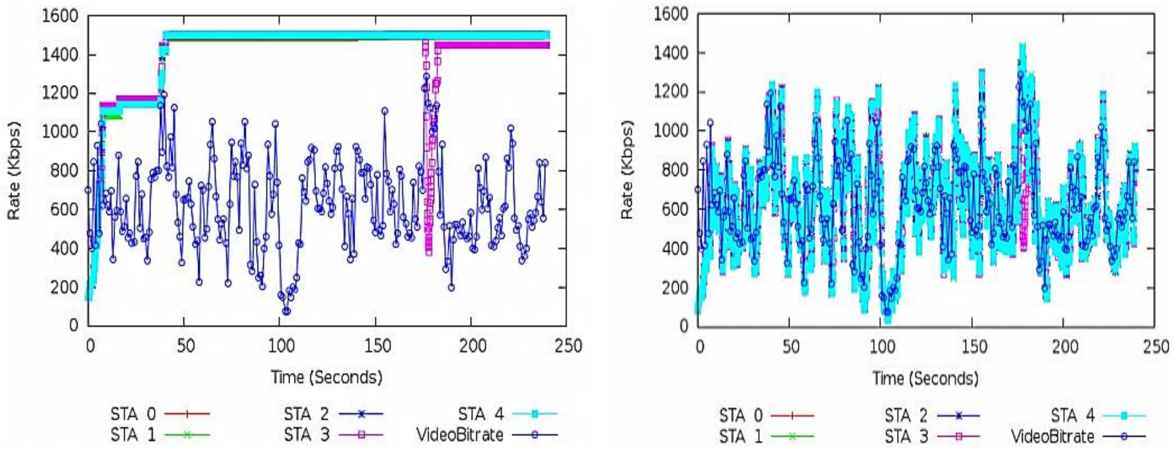


Figure 2. Distance between the nodes is 30m



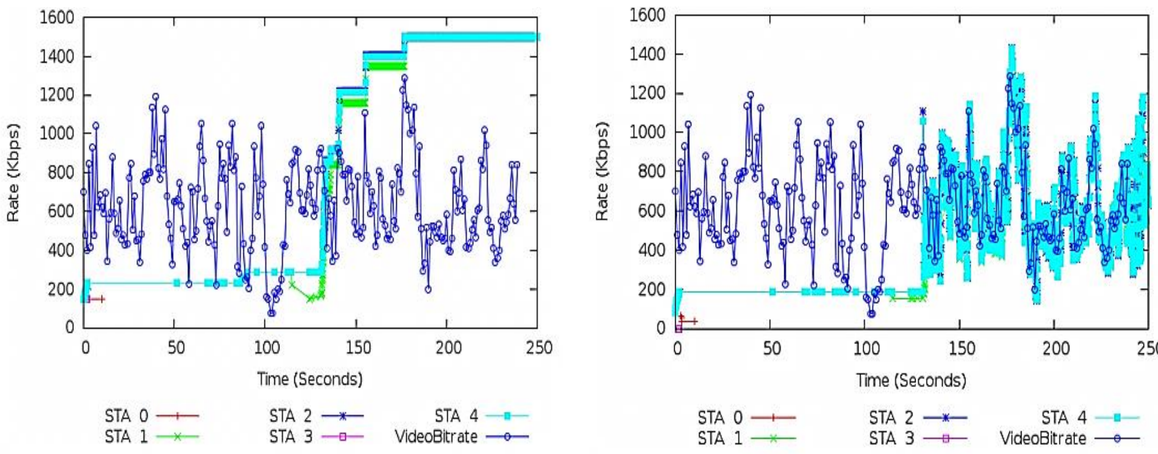
(a) Sending Rate

(b) Receiving rate

Figure 3. Sending Rate and Receiving Rate of the video when the nodes are at the distance of 5m, Packet size of 1000 bytes, and maximum rate of 1500Kbps with the fixed encoding of 360p

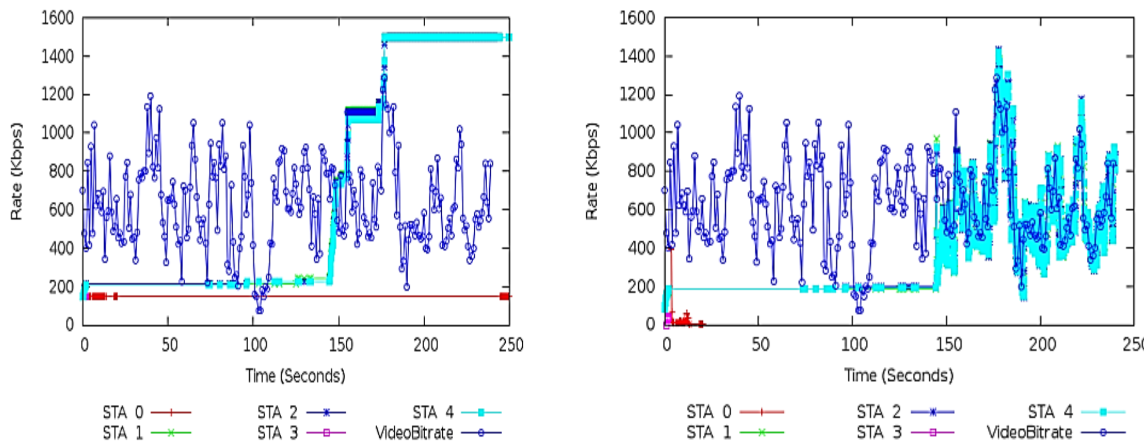
Then, the experiment is conducted again by configuring distance between the nodes as 30m in Figure 2. The results of the experiment are presented in Figure 4. Out of five stations, STA 1, 2 and 4 are within range of 30m. However, STA 0 and STA 3 are within the range of 60m. Hence, the STA 0 and 3 had experienced higher attenuation losses. Due to the higher number of packet losses, both STA 0 and 3 have the received video frames for first 10sec and then all other frames are dropped. In comparison to the results of the experiments with 5m, in this case, stations 1, 2 and 4 had reached the maximum rate of 1500Kbps only after 150 seconds. i.e., during the first 150 seconds, the RMCAT had transmitted and delivered the video frames are at a low data rate and then video frames are delivered at the high data rate. In summary, out of five stations, two stations have no video delivery after first 10 seconds and the remaining three stations had the maximum rate only after 150 seconds.

To study the impact of packet size on the performance of RMCAT, the experiment with 30m distance is re-conducted with a packet size of 500 bytes. The results in Figure 5 show that the stations (STA 0 and 3) which are far away from AP were transmitted the video frames at a low data rate of 200Kbps for the entire duration of the video and those frames are received at the rate of 10Kbps. Transmission of packets with smaller packet size had reduced the packet losses.



(a) Sending Rate (b) Receiving rate

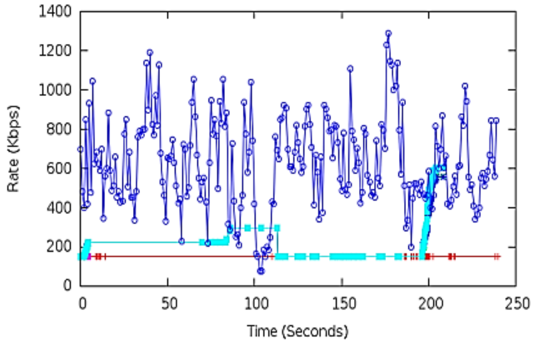
Figure 4. Sending and Receiving rates of the video when the nodes are at the distance of 30m, Packet size of 1000 bytes, and maximum rate of 1500Kbps with the fixed encoding of 360p



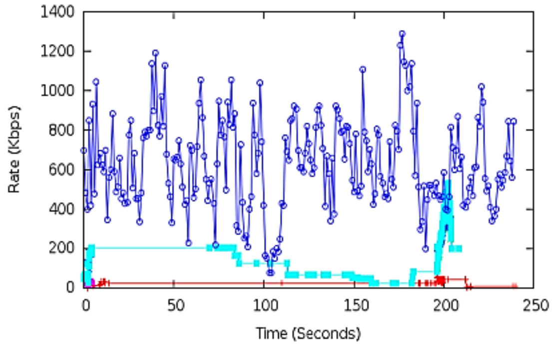
(a) Sending rate (b) Receiving rate

Figure 5. Sending and Receiving rates of the video when the nodes are at the distance of 30m, Packet size of 500 bytes, and maximum rate of 1500Kbps with the fixed encoding of 360p

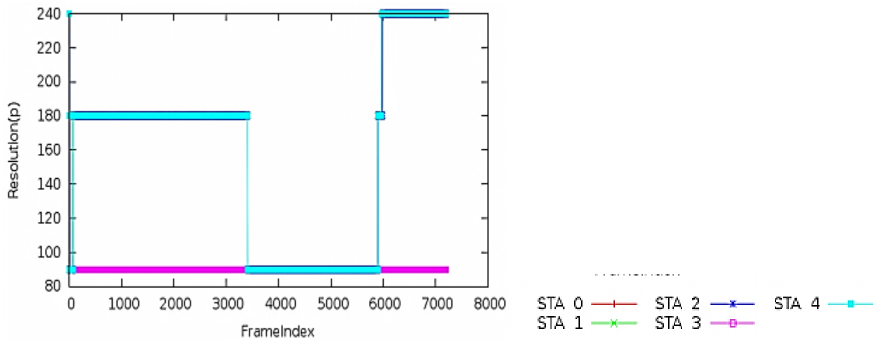
To study the impact of variable encoding on the performance of RMCAT, the experiment with 30m distance is re-conducted with variable encoding and packet size of 1000 bytes. The results in Figure 6) show that the video frames are transmitted to STA 0 and 3, at a low data rate of 200Kbps and received at a maximum rate of 40Kbps. However, in this case, sending rate and receiving rate other stations are significantly dropped. The RMCAT senders had chosen the low resolution (90p) for the video frames. Out of five stations, STA 4 has the chosen the resolution of 180p for first 3000 frames, followed by 90p for the next 3000 frames and the resolution of 240p for the remaining frames. In summary, due to the selection of resolution adaptively by the live encoder, the entire video is delivered to the receivers with losses. To study the impact of smaller packet size and variable encoding, the experiment is re-conducted with 500bytes with variable encoding. The results in Figure 7 show that all stations are received the video frames at low data rate with low resolutions. The stations 1, 2 and 4 were able to transmit the last 2000 frames of the duration of 70seconds at a higher resolution of 240p.



(a) Sending Rate

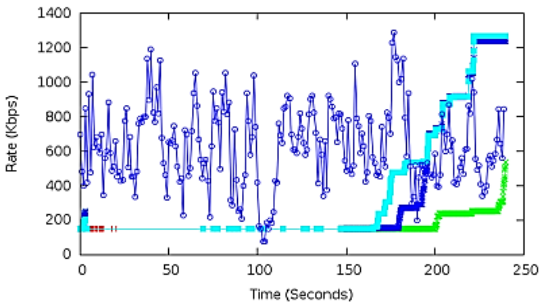


(b) Receive rate

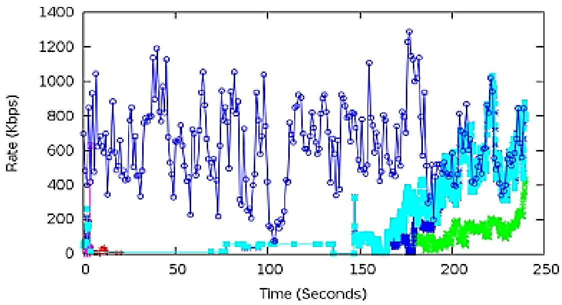


(c) Frame Index vs Resolution

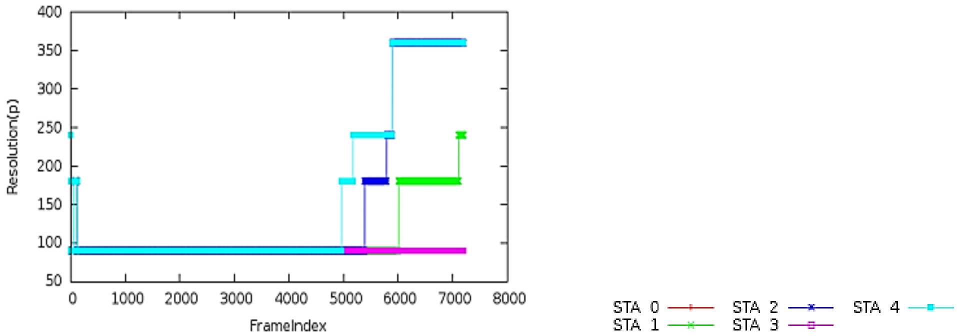
Figure 6. Sending and Receiving rates of the video when the nodes are at the distance of 30m, Packet size of 1000 bytes, and maximum rate of 1500Kbps with variable encoding



(a) Sending Rate



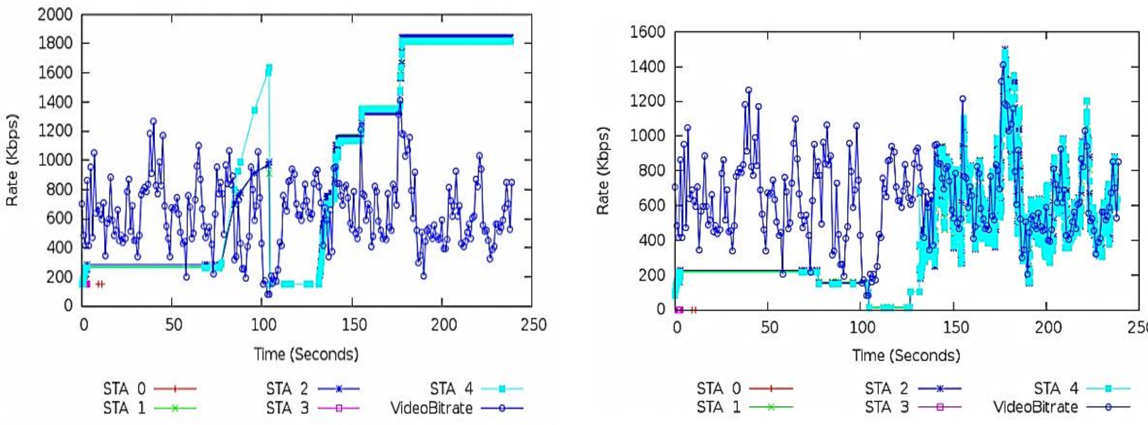
(b) Receive rate



(c) Frame Index vs Resolution

Figure 7. Sending Rate and Receive Rate of the video when the nodes are at the distance of 30m, Packet size of 500 bytes, and maximum rate of 1500Kbps with variable encoding

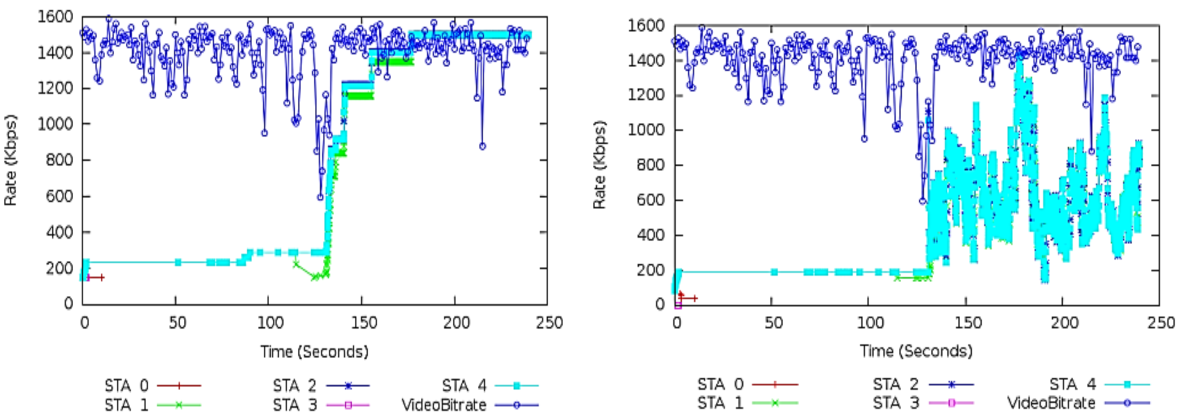
To study the impact of the configured maximum rate of RMCAT sender, the rate is reconfigured as 4000Kbps. The packet size is configured as 1000 bytes with fixed encoding. In this case, the sending and receiving rates are compared with a bit rate of the video encoded with 2000Kbps. The results in Figure 8 show the increased maximum rate has not significantly contributed to the performance of the stations far away from AP (STA 0 and 3), and even to other stations (STA 1, 2 and 4). The stations had reached the maximum sending rate of 1800 Kbps.



(a) Sending Rate (b) Receive rate

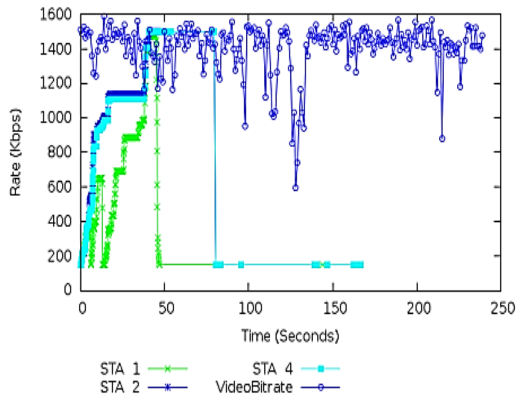
Figure 8. Sending Rate and Receive Rate of the video when the nodes are at the distance of 30m, Packet size of 500 bytes, and maximum rate of 4000Kbps with the fixed encoding of 360

The experiment with 30m distance with a packet size of 1000 bytes is conducted again to study the performance of RMCAT for higher resolution of 720p. In higher resolution, receiving rate is lower than the bitrate of the video. Even though the sending rate has reached the maximum rate of 1500Kbps after 150 seconds, the receiving rate is lower than the bit rate of the video in Figure 9. Finally, the experiment with higher resolution is reconducted with a higher channel bandwidth of 80MHz and the results are presented in Figure 10. As per the specification of IEEE 802.11ac, the aggregated throughput at channel bandwidth of 80MHz should be 3 to 4 times the throughput at channel bandwidth of 20MHz. In contrast, to experiment with 20MHz, the maximum rate is achieved in short time of 40 seconds. However, the sending rate and receiving rate had declined significantly and only the part of the video is delivered to the receivers. This is due to the higher aggregated congestion signal observed by the RMCAT sender. Also, the far away nodes (STA 0 and 3) have not received even a single frame.

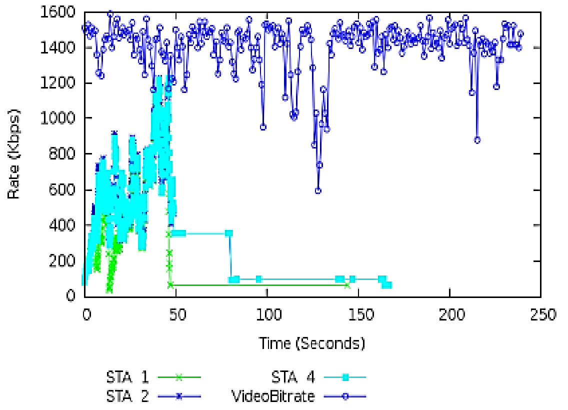


(a) Sending Rate (b) Receive rate

Figure 9. Sending Rate and Receive rates when the node are at the distance of 30m, Packet size of 1000 bytes, and maximum rate of 1500Kbps with the fixed encoding of 720p



(a) Sending Rate

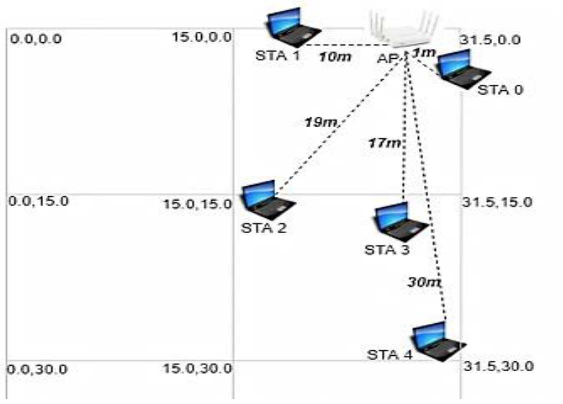


(b) Receive rate

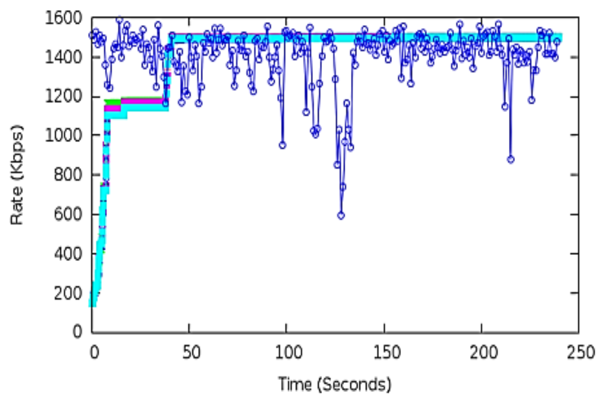
Figure 10. Sending and Receive rates of the video when the Packet size of 1000 bytes, and maximum rate of 1500Kbps with the fixed encoding of 720p and channel bandwidth of 80MHz

4.2. Performance Analysis Over WLAN with a Random Floor Plan

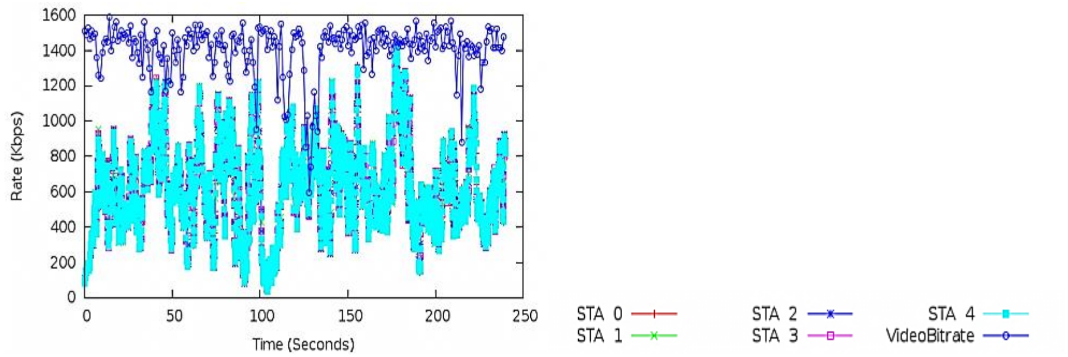
The scenario with a random floor plan is presented in Figure 11(a). In this scenario, Stations 0 to 3 are within the range of 20m and station 4 is at the distance of 30m from the AP. The experiment is conducted with higher resolution of 720p and a higher channel bandwidth of 80MHz. The results in Figures 11(b) and (c) show that the all stations had achieved the same sending rate and received rate. However, the receiving rate is not same as the bit rate of the video like the results of the experiment presented in section 4.1 with the resolution of 720p.



(a) A Random Floor Plan



(b) Sending Rate



(c) Receive rate

Figure 11. (a) Random Plan (b) Sending rate (c) Receive rate when Packet size of 1000 bytes and maximum rate of 1500Kbps with fixed encoding of 720p and channel bandwidth of 80MHz

In summary, the distance of the station from the AP, the error, and higher channel bandwidth had affected the performance of RMCAT. Sending smaller size packet and variable encoding had increased the performance of RMCAT. The congestion control of RMCAT had considered loss and delay to calculate the sending rate. The protocol assumes that higher loss and higher queueing delay are the indications of congestion and hence the sender reduces the sending rate. In 802.11ac the level of congestion is less at the same time attenuation and interference are higher. Also at higher channel bandwidth, a significant decline in performance was observed. Since the channel conditions of WLAN are not considered in the calculation of sending and encoding rates at RMCAT sender, the video transmission is affected. Even when the sender reached the higher sending rate, the receiving rate of the video frames at the receiver is lesser than the required bit rate of the video.

5. CONCLUSION

In this paper, we have studied the performance of RMCAT over WLAN with various network and channel conditions. The results have shown that the performance of video transmission is affected by the distance and channel conditions. Congestion control of RMCAT considers the packet losses due to channel conditions as congestion and hence the performance is declined. To improve the performance of RMCAT over WLAN, the congestion control procedure should be redesigned by considering various network and channel parameters.

REFERENCES

[1] S. Floyd, et al., "TCP friendly rate control (TFRC): protocol specification," *in IETF RFC 5348*, Sep 2008.

[2] A. Sterca, "*UTFRC-Utility-driven TCP-friendly rate control for multimedia streams*," in Proc. 17th Euromicro International Conference, pp. 167-172, Feb 2009.

[3] A. Sterca, et al., "A.: Media-friendly and TCP-friendly rate control protocols for multimedia streaming," *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 26, pp. 1516-1531, 2016.

[4] H. Schulzrinne, et al., "RTP: A Transport Protocol for Real-Time Applications," *RFC 3550*, Jul 2003.

[5] X. Zhu and Z. Sarker, "Framework for Real-time Media Congestion Avoidance Techniques," Internet-Draft, Jul 2016.

[6] Govindarajan J., et al., "Enhanced TCP NCE: A Modified Non-Congestion Events Detection, Differentiation and Reaction to Improve the End-to-End Performance Over MANET," *Progress in Intelligent Computing Techniques: Theory, Practice, and Applications*. Springer, Singapore, pp. 443-454, 2018.

[7] Govindarajan J., et al., "Analysis of TCP- unfairness from MAC layer perspective in wireless Ad-hoc networks," *Indian Journal of Science and Technology*, vol. 8, 2015.

[8] G. M. Asha, "An Adaptive Cross-Layer Architecture to Optimize QoS Provisioning in MANET," *Indonesian Journal of Electrical Engineering and Computer Science (IJEECS)*, vol. 6, Apr 2017.

[9] U. Ahmad, et al., "Fairness Evaluation and Comparison of Current Congestion Control Techniques," *Indonesian Journal of Electrical Engineering and Computer Science (IJEECS)*, vol. 1, Jan 2016.

[10] N. R. Reddy, et al., "Performance Enhancement of TCP Friendly Rate Control Protocol over Wired Networks," *International Journal of Electrical and Computer Engineering (IJECE)*, vol. 6, Dec 2016.

[11] K. Anshu and S. Mathi, "*Active Home Agent Load Balancing for Next Generation IP Mobility based Distributed Networks*," International Conference on Ubiquitous Communications and Network Computing, Springer, 2017.

[12] X. Zhu, et al., "NADA: A Unified Congestion Control Scheme for Real-Time Media," Internet- Draft, 2017.

[13] M. D. Dianu, et al., "*Measurement-based study of the performance of IEEE 802.11 ac in an indoor environment*," Communications (ICC), 2014 IEEE International Conference on. IEEE, 2014.

[14] Leung, K. C., Lai, C., & Ding, H. "TCP-NCL: A serialized-timer approach for enhancing TCP over heterogeneous wired/wireless networks", *Computer Communications*, vol.*144*, pp.198-211, 2019.

[15] Daigavhane, Monika U., and Manish D. Chawhan. "*Congestion control algorithm for TCP in wireless network.*" 2018 4th International Conference on Recent Advances in Information Technology (RAIT). IEEE, 2018.