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Channel Aware/Buffer Aware Adaptive Video Quality Scaling Algorithm for LTE Networks

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Abstract: Video-on-Demand (VoD) and video conferencing over wireless broadband networks require optimal dynamic resource allocation to accommodate the time-varying wireless channel and video content conditions. In order to enhance the system performance with time-varying wireless channel, Long Term Evolution (LTE) adopts Adaptive Modulation and Coding (AMC) schemes which dynamically adjust the Modulation and Coding Schemes (MCS) in accordance with the reception conditions to achieve higher throughput. Also in LTE network, as the number of multimedia users increase, bandwidth becomes scarce. Due to this, number of Packets received per user decreases and delay increases which leads to reduction in Quality of Experience (QoE) for end users. Hence in order to enhance the QoE, this paper presents a novel adaptive video quality scaling algorithms namely Channel Aware Adaptive Video Quality Scaling (CAVQS) algorithm and Buffer Aware Adaptive Video Quality Scaling algorithm (BAVQS). The proposed CAVQS and BAVQS algorithms adapt video packet size based on radio reception condition of the UEs and Packet Delivery Rate (PDR) at the received buffer of the UE respectively. These adaptive algorithms offer tailored solutions to the challenges posed by bandwidth scarcity and dynamic wireless channel conditions in LTE networks.

Keywords: BAVQS, CAVQS, CQI, PDR, QoE, Video Streaming Service (VSS) system.

1. INTRODUCTION

Advanced multimedia applications like video conferencing and video streaming are increasing tremendously over mobile broadband networks. These applications have unique challenges due to higher bandwidth requirements and the delay sensitive nature compared to other data services such as web browsing etc. However in a mobile broadband network, due to limited bandwidth and stochastic nature of wireless channels, it is a challenging task to provide Quality of Experience (QoE) like fixed broadband networks [1]. In order to cope up channel variations, Long Term Evolution (LTE) uses Adaptive Modulation and Coding (AMC) schemes, which adapts different modulation and coding schemes (MCS) such as QPSK, 16QAM and 64QAM etc., in accordance with the channel conditions, which results in varied spectral efficiency. Therefore, the High Definition (HD) video streaming users moving towards bad reception conditions may experience discontinuous video playout as a consequence of reduced spectral efficiencies which would deteriorate the QoE. Further, in LTE network as the number of multimedia user increases, bandwidth may become scarce. Due to this, number of video packets received per user decreases and delay increases leading to degradation of QoE for end users.

Hence in this paper, the adaptive video quality scaling algorithms: Channel Aware Adaptive Video Quality Scaling (CAVQS) and Buffer Aware Adaptive Video Quality Scaling (BAVQS) are proposed for video services which vary the video quality according to reception conditions and received packet delivery rate respectively. The rest of this paper is organized as follows. Section 2 outlines the related work in the literature. The Video Streaming Service (VSS) system in LTE network is discussed briefly in Section 3. Section 4 describes the proposed Channel Aware Adaptive Video Quality Scaling (CAVQS) algorithm for LTE network. In Sections 5, the proposed Buffer Aware Adaptive Video Quality Scaling (BAVQS) algorithm is presented. Section 6 discusses the simulation results followed by conclusion in Section 7.

2. LITERATURE REVIEW

The authors of paper [2] have investigated the benefits of flexible resource allocation when performing Hypertext Transfer Protocol (HTTP)-based Adaptive Streaming (HAS) across cellular systems such as LTE. Sergio Cicalò et.al., have proposed algorithms to enhance fairness between multiple HAS video clients sharing the same wireless channel may experience different video qualities as well as different play-out buffer levels, as a result of different video content complexities and different channel conditions in paper [3]. In paper [4] it is depicted that the HTTP progressive video downloading is becoming increasingly relevant in wireless and mobile networks. Therefore, improving the QoE for this service is important for customer satisfaction. In paper [4], authors have proposed a QoE-aware scheduling algorithm for the HTTP progressive video downloading in Orthogonal Frequency Division Multiple Access (OFDMA) systems. Authors of paper [5] have explored new ways to optimize wireless networks for video services towards delivering enhanced QoE. One of these key video enhancing solutions is HTTP adaptive streaming as discussed in [5]. In Paper [6], cross-layer techniques have been widely adopted in literature for dynamic resource allocation to maximize data rate in OFDMA based systems. In the paper [7], authors present a hybrid edge cloud and client adaptation framework for HTTP adaptive streaming by taking advantage of the new capabilities empowered by recent advances in edge cloud computing. In particular, emerging edge clouds are capable of accessing an application layer and radio access networks information in real time. Paper [8] illustrates, in a HTTP adaptive streaming, video content is temporally divided into multiple segments, each encoded at several quality levels. According to [8], the client can adapt the requested video quality based on tariff for smoother playback. Paper [8] has presented a novel algorithm by reducing the segment distance for live streaming solutions which often suffer from play out freezes and a large end-to-end delay. Authors have studied in paper [9] that the QoE of live mobile TV users in the presence of a play out buffer at the receiver side for LTE network for delivering both unicast and broadcast services over lossless channels where interruptions in mobile TV delivery are due to the absence of resources at the base station. The paper [10] proposes an effective real-time video uplink framework for mobile wireless camera networks (WCN) over OFDMA based infrastructure. Within this dynamic environment, the Dynamic Adaptive Streaming over HTTP protocol, utilizing the Transmission Control Protocol (TCP), has emerged as the primary method for delivering video streaming services [11, 12, and 13]. Paper [14] illustrates mobile DASH algorithm where a client controls the streaming rate and the base station in the mobile network decides the resource allocation. Authors of paper [15] have proposed mechanisms for delivering video over mobile wireless networks and illustrated several unique challenges faced like limited bandwidth and packet delay variation which can cause artifacts such as picture freezes or blockages that leads to unacceptable QoE for a mobile client. In the paper [15], authors have addressed techniques to use feedback from the wireless network like congestion, channel conditions, content, user profiles etc. In paper [16] a Channel Aware Optimized Proportional Fair (CAOPF) scheduling policy is proposed to optimize the channel behaviour that shares traffic considering the measured CQI values. A novel resource scheduling algorithm is proposed in [17] that offer improvements in QoS for video traffic under heavy load conditions. The proposed algorithm in [17] optimizes the EXPRule scheduler more efficiently by utilizing information on the channel that is queued without previous information on arrival or channel statistics for traffic. Authors in [18] proposed an efficient video streaming strategy that incorporates GRU-based bandwidth estimation and an adaptive bit rate selection algorithm to enhance the QoE of adaptive bit rate video streaming mobile systems.

3. VIDEO STREAMING SERVICE (VSS) SYSTEM FOR LTE NETWORK

The block diagram of typical end-to-end 3GPP Video Streaming Service (VSS) system in LTE network is shown in Figure 1.

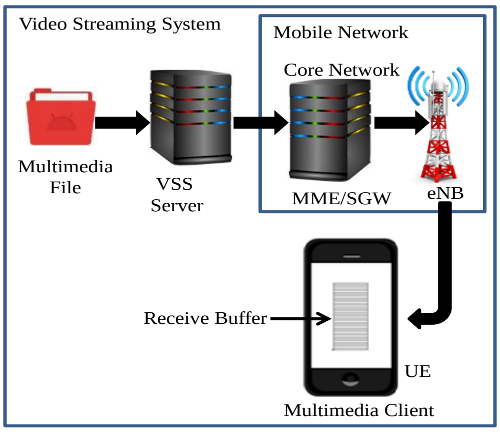

Figure 1. Block diagram of typical End-to-End 3GPP VSS system in LTE network

Media from a VSS server is delivered to the streaming client located in the UE through LTE network that includes core network and Radio Access Network (RAN). The QoE experienced by users in VSS system depends on traffic characteristics and LTE network constraint such as PDCP, RLC, MAC selection etc. Media transmission from VSS server can occur either by Constant Bit Rate Packet (CBRP) transmission or by Variable Bit Rate Packet (VBRP) transmission.

CBRP adjusts the delay between consecutive packets to constant rate and VBRP uses a variable rate for packet transmission. Also in VBRP, transmission time of packet estimated solely on the timestamp of the frame from which the packet is derived. Hence in conversational multimedia services such as video calling and video conferencing typically employs CBR control. In CBR control, the next frame is not obtained from the source until all bits of the current frame have been encoded and transmitted at a constant channel rate. Whereas Variable Bit Rate (VBR) video rate control strategies are employed for applications that can tolerate variations in delay and don't have stringent constant transmission rate requirements. This flexibility enables more efficient utilization of bandwidth and can lead to improved video quality, especially in scenes with varying levels of detail or motion.

3.1 Packet Size for Video Transmission

Streaming server media transmission strategies such as VBRP and CBRP can use smaller and larger video packet sizes. There are no theoretical limitations for the usage of small packet sizes, however using too small video packets (16, 32 or 64 bytes) would increase the bandwidth requirement due to increased header size of packet for a given media bit rate. On the other hand, using large packet sizes causes the higher end-to-end latency for the reception of the packets at the VSS client. Also, retransmission of packets in LTE network may incur additional delay and jitter for larger packet sizes for a given packet loss rate. Also, fragmentation of larger packets into smaller segments may result in increased bandwidth requirement due to additional header overhead. In order to tackle these issues, VSS servers can adapt packet sizes to varying network conditions. VSS can adapt video packet sizes with simple transmission of a single pre-encoded bit stream without relying on the feedback from the streaming client or adaptive transmission of pre-encoded bit streams based on the feedback from the streaming client [19].

Hence in this paper, two novel adaptive transmission algorithms have been proposed for adapting the packet size according to radio reception and packet delivery rate feedback from media client with CBRP rate control.

4. PROPOSED CHANNEL AWARE ADAPTIVE VIDEO QUALITY SCALING ALGORITHM

Mobile TV and Video on Demand (VoD) streaming are popular services over LTE networks. Delivering these video applications over LTE networks while maintaining QoE is a challenging task due to stochastically varying channel quality and higher bandwidth requirements. In LTE networks, the reception conditions are fed back to eNB using 4-bit Channel Quality Indicator (CQI) from each UE as a part of Channel Status Information (CSI) periodically for every 10ms [20]. Based on reported CQI value, eNB adapts suitable Modulation and Coding Schemes (MCS) for downlink connections of UEs to reduce Bit Error Rates (BER) [20] as listed in table 1. In general, the UE near the eNB may be in the good reception condition with higher CQI value in the range 15-10 and UE near cell edge may be in the bad reception condition with lower CQI value of 6-0 [20].

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Table.1. Relationship between CQI and Video Quality Adopted [20].

SINR (dB)	CQI	Modulation Scheme	Coding rate	Spectral Efficiency (bps/HZ)	CRI	Video packet size
-	0	-	-	-	0	128
-7.27	1	QPSK	0.076	0.1523	0	128
-4.76	2	QPSK	0.12	0.2344	1	256
-2.06	3	QPSK	0.19	0.377	1	256
0.61	4	QPSK	0.3	0.6016	2	384
2.81	5	QPSK	0.44	0.877	2	384
4.69	6	QPSK	0.59	1.1758	3	512
6.29	7	16QAM	0.37	1.4766	3	512
8.69	8	16QAM	0.48	1.9141	4	640
11.37	9	16QAM	0.6	2.4063	4	640
13.11	10	64QAM	0.45	2.7305	5	768
16.44	11	64QAM	0.55	3.3223	5	768
19.62	12	64QAM	0.65	3.9023	6	896
23.01	13	64QAM	0.75	4.5234	6	896
26.19	14	64QAM	0.85	5.1152	7	1024
28.66	15	64QAM	0.93	5.5547	7	1024

From Table 1, it is evident that the eNB adapts 64QAM modulation scheme with higher coding rates to downlink connections of UEs with the higher CQI values. This achieves higher spectral efficiency which in turn suffices HD quality video streaming services with allocated Resource Blocks (RBs). Further, for UEs with the CQI values of 9-7, the eNB adapts 16QAM modulation scheme to achieve moderate spectral efficiency. However for UEs with the lower CQI values (6-0), the eNB accord QPSK modulation scheme with lower coding rates which leads to lower spectral efficiency. Due to the lower spectral efficiencies, the requirement of HD quality video streaming services may not be sufficed with the allocated RBs leading to service interruptions, buffer underflow and video glittering problems, causing reduction in QoE for end users (figure 2).

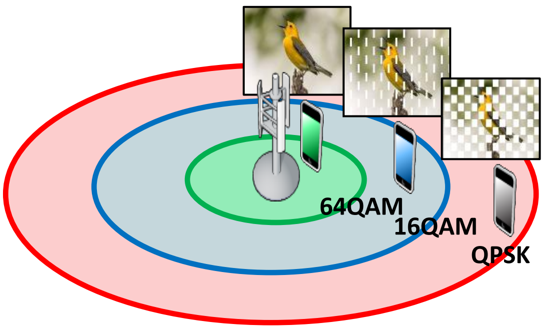


Figure 2. Illustration of video glittering problems caused when lower number of RBs is allocated to UE to suffice the HD quality video streaming services

In pursuance of enhancing the QoE of the end user in the bad reception conditions, a novel CAVQS algorithm has been proposed to scale the video quality (video packet size) according to the reported CQI value. In this proposed work, the cell is divided into eight regions based on CQI values. Each of the regions is represented by a Cell Region Indicator (CRI), calculated by the eNB using the equation (1). The video packet size is assigned to UEs based on the CRI values (equation (2)).

$$\text{CRI} = \text{Integer value of} \left(\frac{\mathit{The\ number\ of\ regions}}{\mathit{Maximum\ CQI\ Value}+1} \times \text{retrieved CQI value} \right) \quad (1)$$

$$\text{Video Packet Size} = (CRI + 1) \times Video\ Resolution \quad (2)$$

Where the video resolution=128 bytes

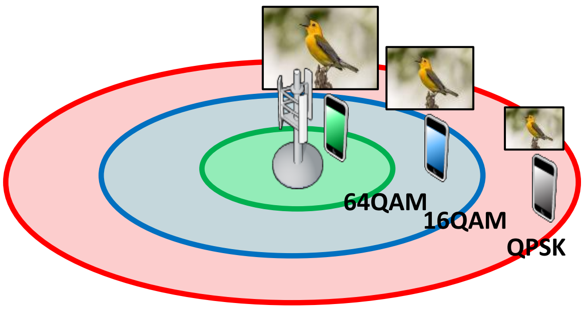


Figure 3. Illustration of graceful degradation of Video packet size to enhance QoE for users with HD quality video streaming services

CRI values and corresponding video packet size are calculated for various proposed regions listed in table 1. The flowchart and illustrations of effect of proposed Channel Aware Adaptive Video Quality Scaling (CAVQS) algorithm on video streaming are shown in figure 3 and figure 4 respectively.

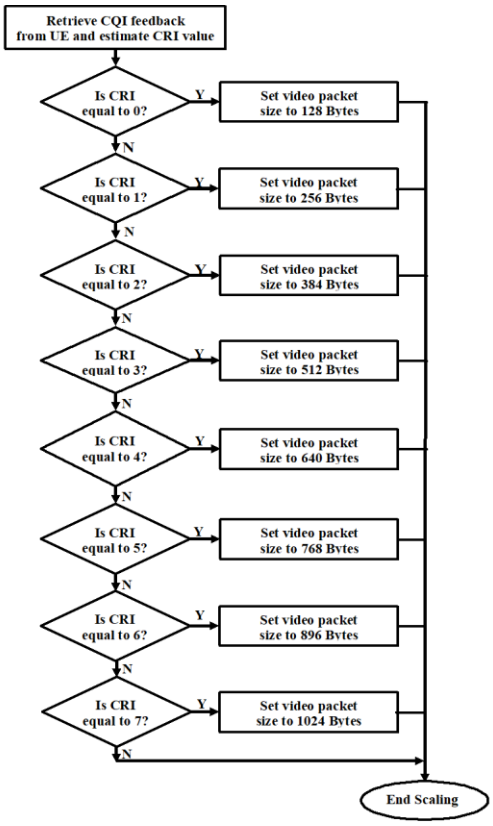


Figure 4. Flow chart of proposed Channel Aware Adaptive Video Quality Scaling algorithm

5. PROPOSED BUFFER AWARE ADAPTIVE VIDEO QUALITY SCALING ALGORITHM

In recent days, exponential growth of multimedia services such as HD video streaming, video on demand, video conferencing by mobile users has paved a way for the network venders and operators to adapt LTE network technology which promises higher data rates at reduced latencies than the 3G systems. In order to realize data rates required for sufficing multimedia streaming services over stochastically varying wireless channels, LTE has adopted advanced Radio Resource Management (RRM) mechanisms such as AMC, Call Admission Control (CAC) and scheduling. AMC schemes in LTE achieves higher data rates (Spectral efficiencies) by adapting 64QAM modulation schemes with higher coding rates for users with good radio reception conditions and with lower Bit Error Rates (BER). Further for users with bad reception conditions, in order to reduce BER, LTE network has adapted 16QAM and QPSK modulation schemes with lower coding rates which reduces the spectral efficiency achieved than with 64QAM modulation scheme. Since spectral efficiency is the number of successfully transmitted bits per second per Hertz of bandwidth, number of users which can be served by LTE network depends on adapted modulation and coding scheme and on the bandwidth with which the network is deployed. On the other hand, in a deployed network, the bandwidth is limited. Due to this limited bandwidth, the number of users provisioned with services would also be limited for an adopted modulation and coding scheme within a cell. Hence even when the number of UEs assigned with 64QAM modulation and higher coding rates may increase beyond the capacity of the cell, which is the product of maximum achievable spectral efficiency per user and number of users in the cell for a given outage probability (the probability that the system cannot successfully decode the transmitted symbols) [21], the network congestion occurs.

Table.2. Relationship between PDR, PDR Index and assigned Video Packet Size

Packet Delivery Rate (%)	PDR index	Video Packet Size (Bytes)
0-20	0	128
21-30	1	256
31-40	2	384
41-50	3	512
51-60	4	640
61-70	5	768
71-80	6	896
Above 80	7	1024

Due to network congestion, the LTE network may not able to provide data rates required for the individual UEs to suffice HD video streaming service which leads to discontinuous video play-out, buffer underflow and video glittering problems causing reduction in QoE for the end users. These problems also occur for the UEs assigned with other AMC schemes such as 16QAM and QPSK which have lower spectral efficiencies than 64QAM. In order to reduce discontinuous video play-out, buffer underflow, and video glittering problems and to improve QoE, a Buffer Aware Adaptive Video Quality Scaling (BAVQS) algorithm is proposed. In this algorithm, Packet Delivery Rate (PDR) is calculated using equation (3) at the UE and a PDR index corresponding to the calculated PDR is assigned to UE as per the Table.2. The UE feedbacks the PDR index to the eNB as a part of CSI information which is fed periodically for every 10ms. In accordance with the reported PDR index, the eNB calculates video size using equation (4) and allocates to the respective UEs.

$$\text{Packet Delivery Rate} = \frac{\text{Number of Packets Received per Sec}}{\text{Number of Packets Transmitted per sec}} \times 100\% \qquad (3)$$

$$\text{Video Packet Size} = (\text{Reported PDR index} + 1) \times 28 \text{ bytes} \qquad (4)$$

The flowcharts for BAVQS algorithm in eNB and UE are shown in figure 5(a) and figure 5(b) respectively.

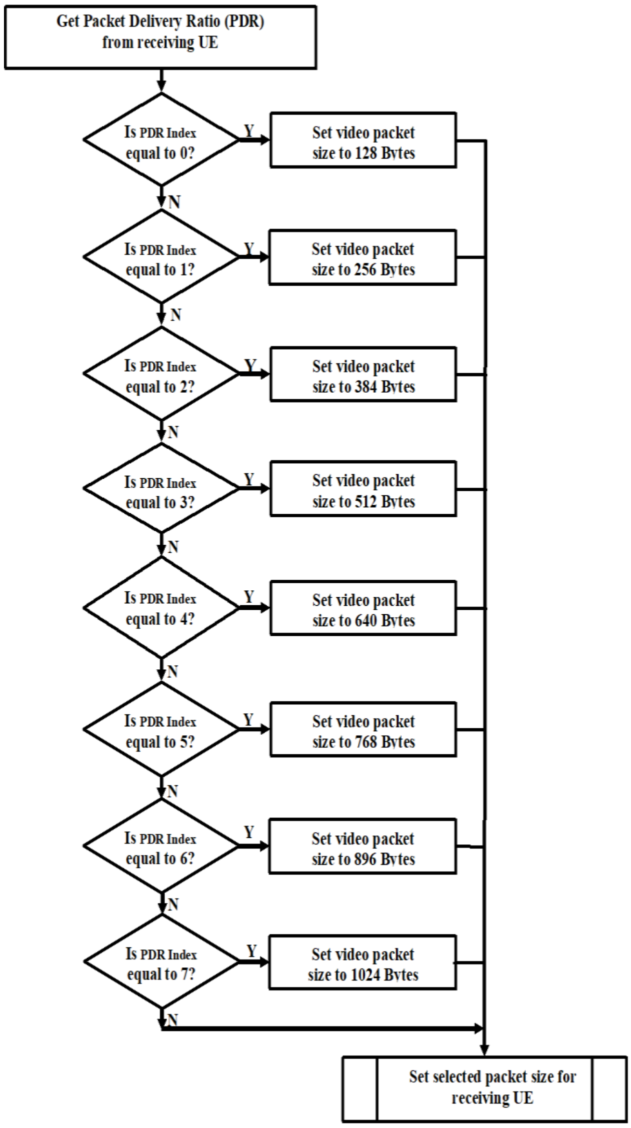


Figure 5(a). Flow chart of proposed Buffer Aware Adaptive Video Quality Scaling algorithms at eNB.

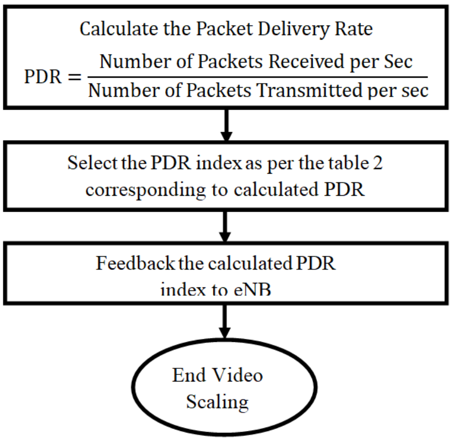


Figure 5(b). Flow chart of proposed Buffer Aware Adaptive Video Quality Scaling algorithm at UE.

6. SIMULATION AND RESULTS

The performance of proposed Channel Aware Adaptive Video Quality Scaling (CAVQS) algorithm and Buffer Aware Adaptive Video Quality Scaling (BAVQS) algorithm are evaluated for CBR traffic with single cell environment using QualNet 7.1 network simulator. Two-ray path loss model with constant shadowing is considered for the simulation studies. The remaining parameters considered for simulation studies are listed in table 3.

Table.3. Simulation Parameters

6.1 Scenario 1

	Property	Value
	Simulation-Time	30 seconds
	Simulation-Area	5Km X 5Km
	Downlink-Channel-Frequency	2.4GHz
	uplink-Channel-Frequency	2.5GHz
	Propagation-Model	Statistical
	Shadowing mean	4dB
	Channel-Bandwidth	10MHz
	Antenna-Model	Omnidirectional
eNB	Scheduling Algorithm	Round Robin
eNB	PHY- Tx-Power	23dBm
eNB	PHY-Num-Tx-Antennas	1
eNB	PHY-Num-Rx-Antennas	2
eNB	Antenna-Height	12m
UE	MAC-Scheduler-Type	Simple-Scheduler
UE	PHY- Tx-Power	20dBm
UE	PHY- Tx-Antennas	1
UE	PHY- Rx-Antennas	1
UE	Antenna-Height	1.5m

Figure 6 shows the snapshot of scenario 1 designed for performance evaluation of proposed CAQAS, BAQAS algorithms and SPS-VSS for 100 nodes placed randomly inside the cell.

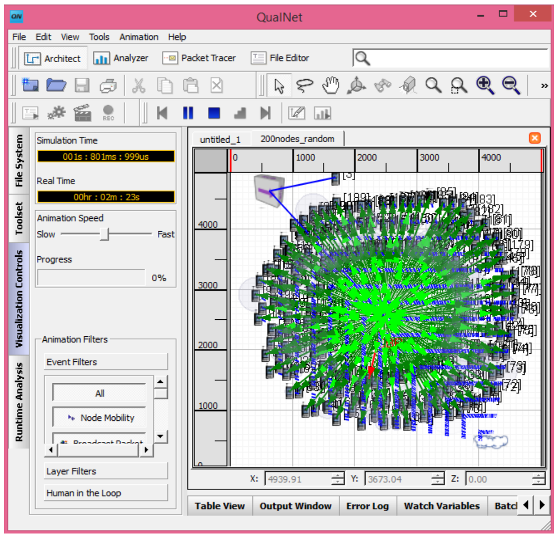
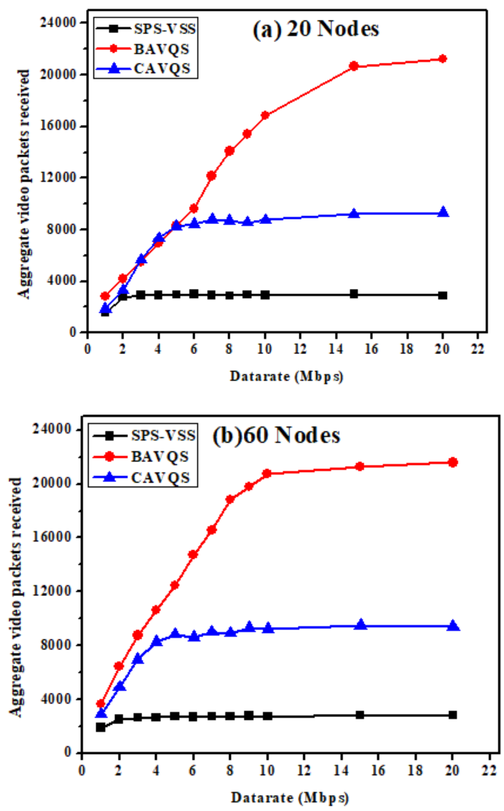


Figure 6. Snapshot of the Scenario1

In this scenario, the simulation studies are carried out by varying the data rates for different node densities in Rayleigh fading environment. Initially the simulation is carried out for CAVQS algorithm by placing 10 UEs randomly throughout the cell with a downlink CBR connection of 1Mbps established between eNB and each UE. The performance metrics such as aggregate video packets received, average delay, average jitter and number of packets discarded due to RLC transmission buffer overflow are recorded. Simulation studies are repeated by varying the data rates from 2Mbps to 10Mbps in steps of 1Mbps and from 10Mbps to 20Mbps in steps of 5Mbps.Further the simulation studies are repeated by increasing node densities from 20 to 100 in steps of 10 nodes. Also, the similar simulation studies are repeated for BAVQS algorithm and Static Packet Size Video Streaming Service (SPS-VSS) system.



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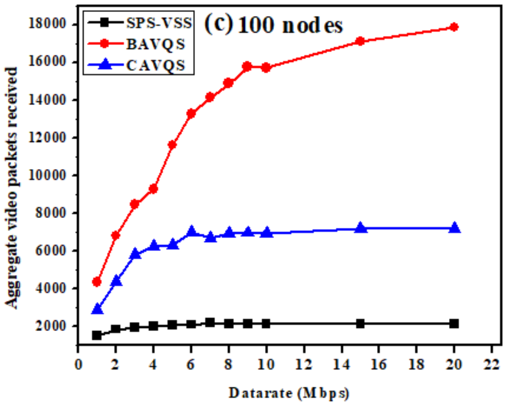


Figure 7 (a-c) shows the aggregate video packets received performance of SPS-VSS, BAVQS and CAVQS algorithms with varying data rates for 20, 60, and 100 node densities respectively.

It is depicted form figure 7(a-c) that Aggregate video packets received performance of SPS-VSS, BAVQS and CAVQS algorithms increases for increase in data rates initially and saturates for further increase in data rates. Since bandwidth is fixed in any deployed network, system capacity in terms of number of bytes delivered over network is limited, leading to saturation of aggregate video packets received performance for higher data rate. It is also observed from figure 7(a-c) that the aggregate video packets received performance for proposed CAVQS and BAVQS algorithm is better compared to performance of SPS-VSS. SPS-VSS allocates static packet sizes preferably packet sizes greater than 750 bytes for HD video streaming services [19]. However large packet sizes may lead to increased video packet delivery time due to segmentation in LTE network, which yields reduction in number of video packets delivered to UE. In addition, segmentation increases the bandwidth requirement due to increased headers overhead, number of retransmissions and increases the possibility of further packet loss [19]. Due to this, aggregate video packet received performance of SPS-VSS is lower than proposed CAVQS and BAVQS algorithm [19]. In contrast to SPS-VSS, the proposed CAVQS and BAVQS algorithms use reduced packet size for HD video streaming services by UEs which experience lower CQI values (13-0) and lower PDR (less than 80%) respectively. The reduced packet size in turn reduces the number of bytes in a packet thereby reducing the possibility of requirement for fragmentation of the packets, video packet transmission latency and retransmission time if packet loss. Due to this, the proposed CAVQS and BAVQS algorithms perform better than SPS-VSS. Further, as the number of users increase or the data rates increase, the number of video packets generated from media server also increases. This causes decreased packet delivery rate (PDR) due to limited system capacity of a deployed network. Consequently BAVQS algorithm achieves higher aggregate packets received performance as PDR value decreases, since BAVQS algorithm assigns reduced packet size and CAVQS algorithm assigns same packet size to UEs with same CQI value.

Figure 8 (a-d) illustrates the aggregate video packets received performance of the SPS-VSS, BAVQS and CAVQS algorithms for varying node densities with 2, 6, 10, and 20Mbps data rates respectively. It is evident from figure 8 (a-d) that the aggregate video packets received performance for BAVQS is better with increasing node density than CAVQS and SPS-VSS algorithms.

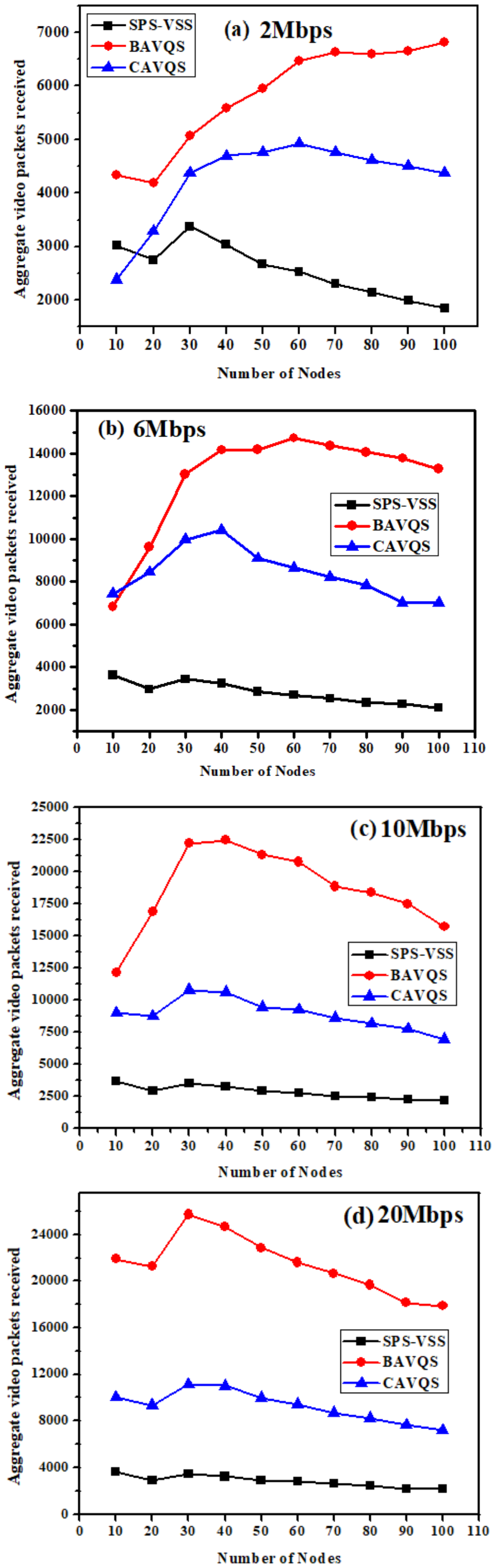


Figure 8. Aggregate video packets received for various node densities (a) 2 Mbps, (b) 6 Mbps, (c) 10 Mbps and (d) 20 Mbps data rates

As the number of users increase in a deployed network with fixed bandwidth, PDR for individual user decreases due to reduced radio resources allocated to individual user which leads

BAVQS algorithm to assign reduced packet size than CAVQS and SPS-VSS algorithms.

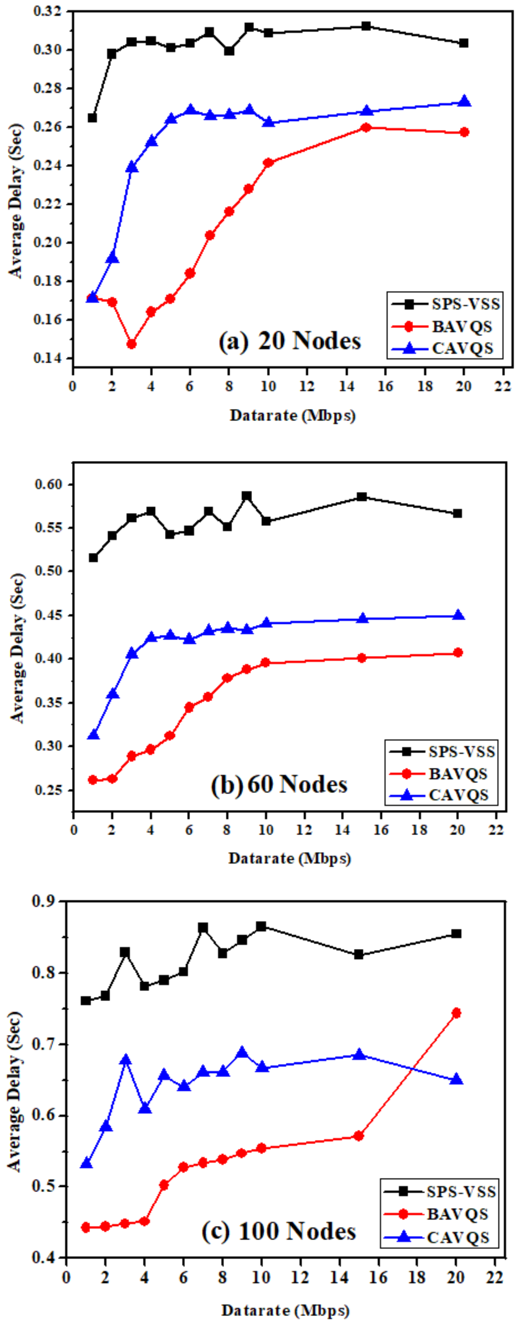


Figure 9. Average delay for various data rate for (a) 20, (b) 60, (c) 100 node densities.

Figure 9 (a-c) and figure 10 (a-c) illustrates the average delay and average jitter performance of the SPS-VSS, BAVQS and CAVQS algorithms for varying data rates for different node densities. It is evident from figure 9 (a-c) and 10 (a-c) that the average delay and jitter for proposed CAVQS and BAVQS algorithms is better than SPS-VSS, since CAVQS and BAVQS algorithms use reduced packet sizes. With reduced packet sizes, number of bytes per packet reduces in turn reducing packet transmission time. Also, SPS-VSS has larger end-to-end delay due to segmentation and reassembly of the larger video packets in LTE network [19]. In continuation, BAVQS algorithm has reduced delay because of further reduction in Video packet sizes as the data rate increases. Also, it is apparent from figure 9 (a-c) that for SPS-VSS algorithm, loss of individual fragment of packet may raise a need for retransmission of all the fragments of a video packet yielding an increased jitter.

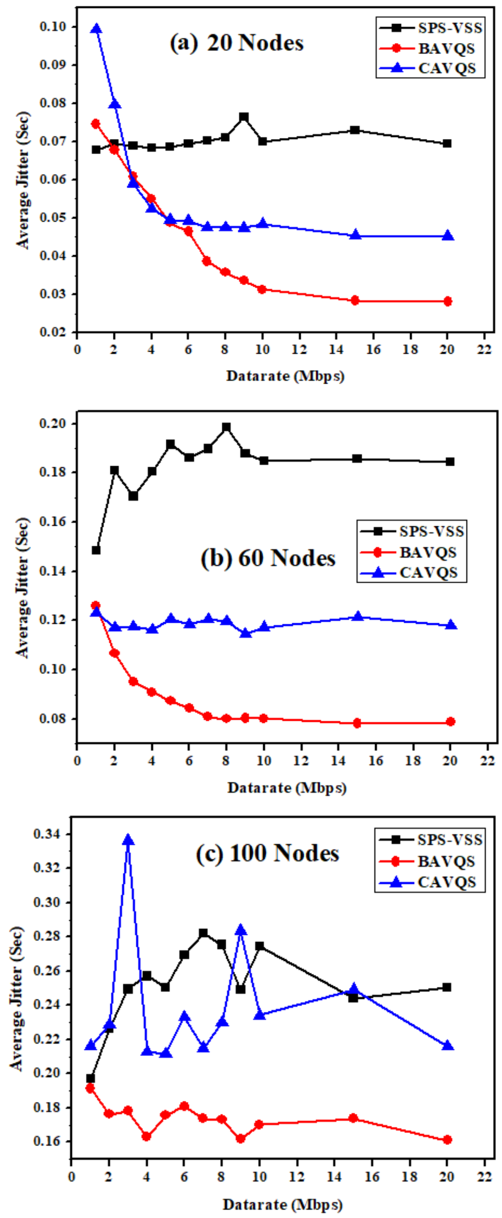
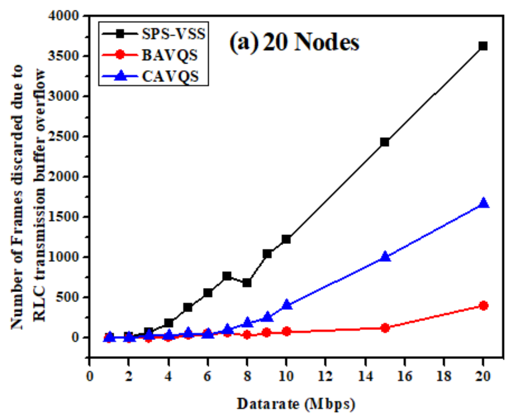


Figure 10. Average Jitter for various data rate for (a) 20, (b) 60, (c) 100 node densities.



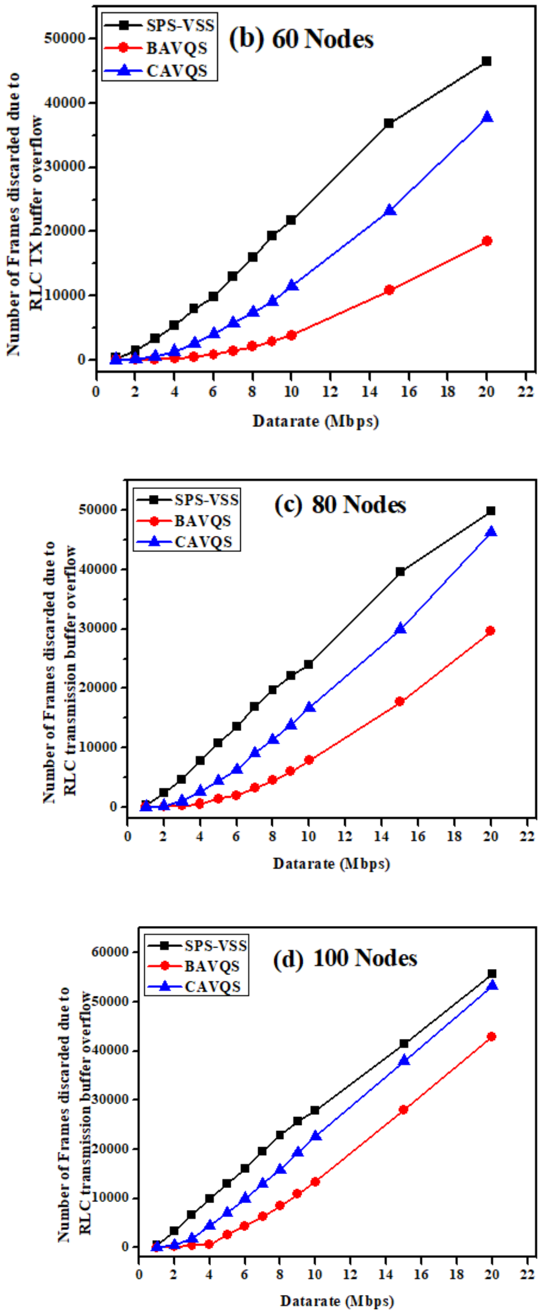


Figure 11. Number of Packets discarded due to RLC TX buffer overflow for (a) 20, (b) 60, (c) 80, (d) 100 node densities

Figure 11(a-d) shows the number of video packets discarded due to RLC Transmission buffer overflow performance of SPS-VSS, BAVQS and CAVQS algorithm. It is evident from figure 10 (a-c) that the number of video packets discarded for BAVQS, SPS-VSS and CAVQS algorithms increases with increasing data rates. As the data rates increases, number of packets generated at media server increases. Since the bandwidth is fixed for a deployed network, this increasing number of packets in RLC buffer increases packet drops. Also, it is evident from figure 11 (a-d) that the number of video packets discarded performance of BAVQS is better compared to CAVQS and SPS-VSS. Since SPS-VSS algorithm uses large video packet sizes, waiting time of packets in RLC transmission buffer increases leading to video packets drop. In addition to this, as BAVQS uses reduced packet sizes than CAVQS, BAVQS achieves superior packet drop performance.

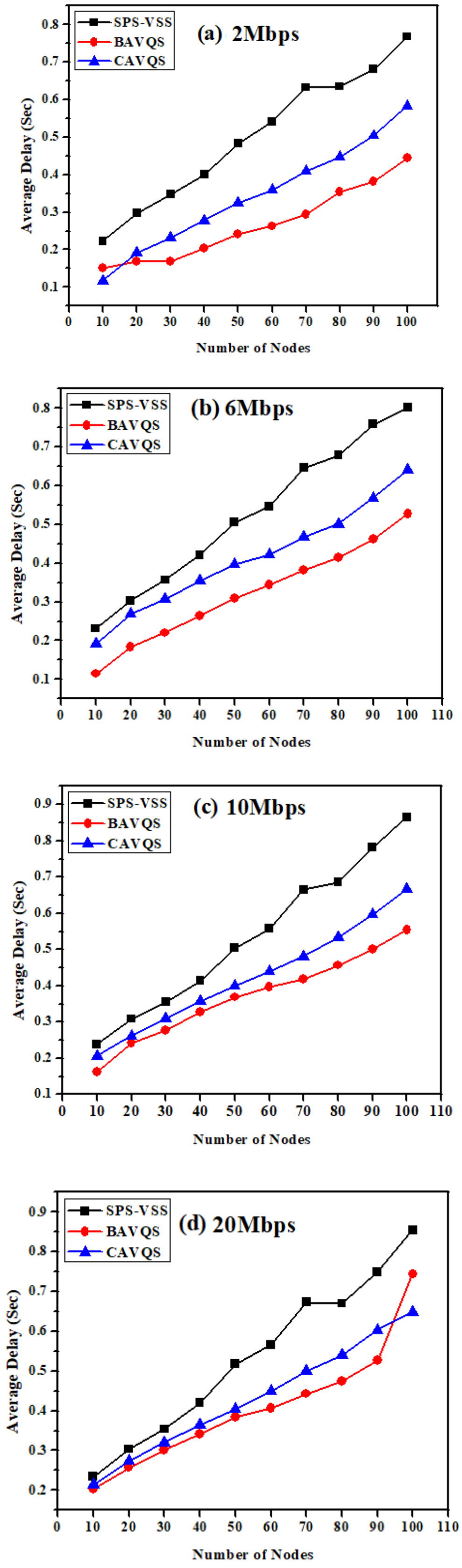


Figure 12. Average delay for (a) 2 Mbps, (b) 6 Mbps, (c) 10 Mbps and (d) 20 Mbps data rates

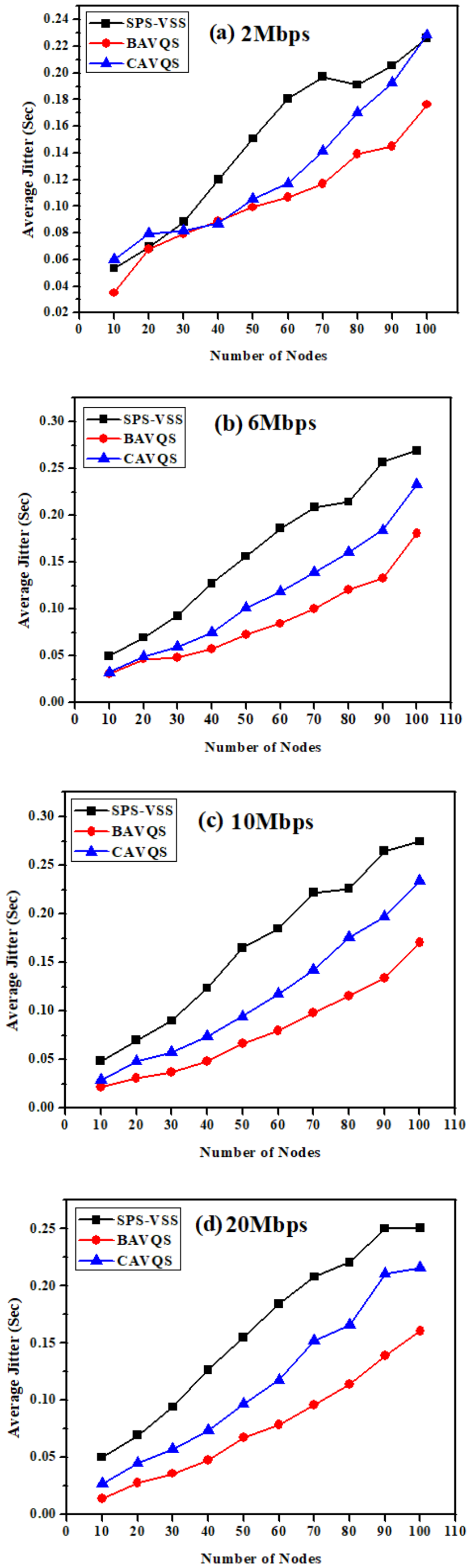


Figure 13. Average jitters for (a) 2 Mbps, (b) 6 Mbps, (c) 10 Mbps and (d) 20 Mbps data rates

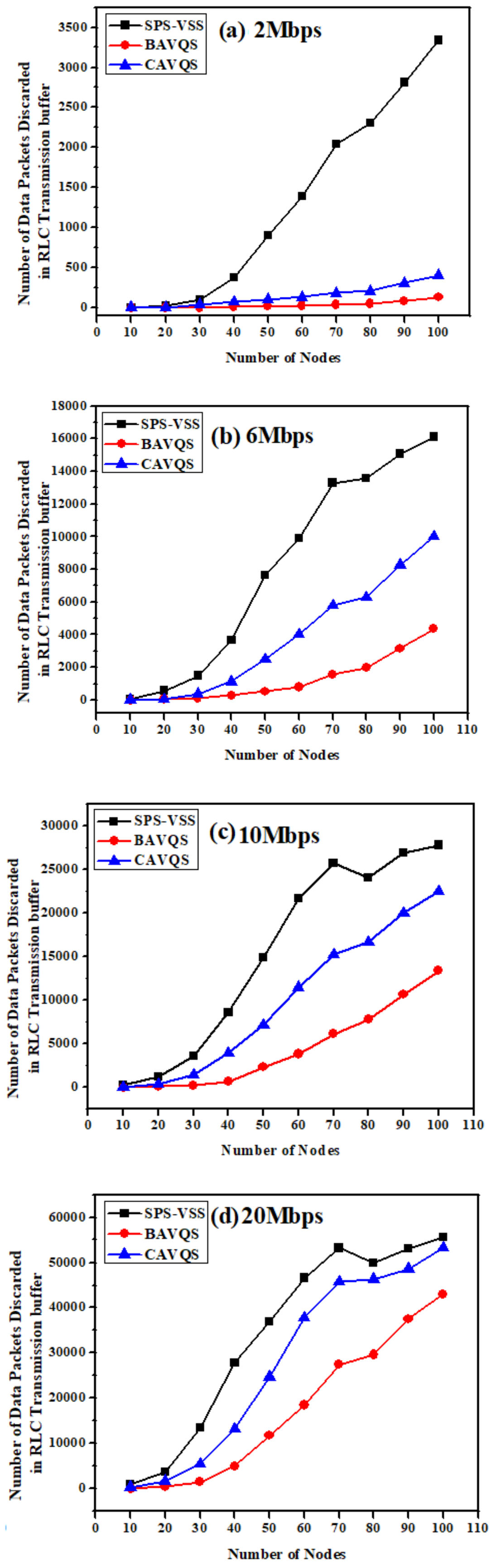


Figure 14. Number of data packets discarded due to RLC tx buffer overflow for (a) 2 Mbps, (b) 6 Mbps, (c) 10 Mbps and (d) 20 Mbps data rates

Figure 12(a-d) and Figure 13(a-d) show the average delay and average jitter performance of the SPS-VSS, BAVQS and CAVQS algorithms for varying node densities with different data rates. It is observed from the figures 12(a-d) and 13(a-d) that the average delay and jitter performances for SPS-VSS, BAVQS and CAVQS algorithm increases with increase in node density. This is because, as the node density increases, the number of radio resources allocated to a user decreases leading to increased delay and jitter. Also as the number of UEs increases in a deployed network with limited bandwidth, waiting time for number of packets in buffers of LTE network increases. This increases average delay for SPS-VSS, BAVQS and CAVQS algorithms.

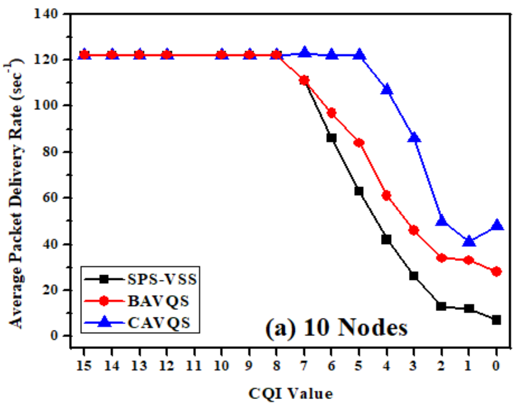
Figure 14 (a-d) shows the number of video packets discarded due to RLC transmission buffer overflow performance of SPS-VSS, BAVQS and CAVQS algorithm. It is evident from Figure 14 (a-d) that the number of video packets discarded due to RLC transmission buffer overflow performance of BAVQS is better compared SPS-VSS and CAVQS algorithm. Since BAVQS uses lower bytes per packet for video transmission, number of bytes loaded in RLC buffer is less. As a consequence, the video packets dropped at RLC transmission buffer reduce.

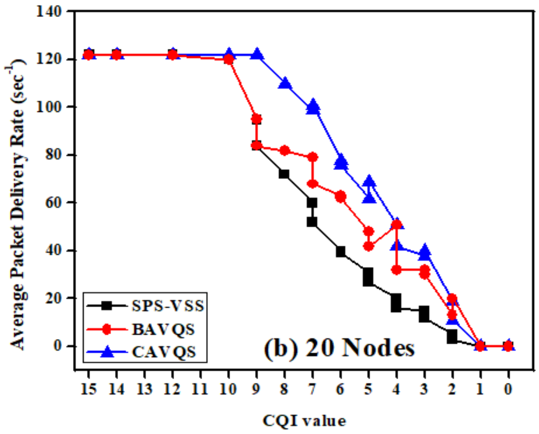
6.2 Scenario 2

In scenario 2, the simulation parameters are retained as in scenario 1 except the fading model. The simulation studies are carried out in no-fading environment with 1Mbps CBR connections established between eNB and each UE. In the considered scenario, 10 UEs are placed at the cell center where the UEs experience good radio reception conditions corresponding to CQI value 15. Further, UEs are made to move in a pre-determined path with a vehicular mobility of 104mps towards the cell edge experiencing radio reception conditions corresponding to all possible CQI values from 15 to 0. During the simulation, the simulation time and CQI value at each video packet reception are logged for each UE. After the simulations, total number of video packets received by each UE corresponding to a CQI value and duration for which that UE experiences a particular CQI value are counted. Further video packets received for a particular CQI value per second is calculated using equation (5).

$$\text{Video packets recieved per a CQI Value} = \frac{\text{Number of Video packets Received per a CQI}}{\text{Duration for which UE experiance a CQI}} \quad (5)$$

The simulation is repeated for 20 and 30 node densities. Further, similar simulation studies are repeated for BAVQS and CAVQS algorithms.





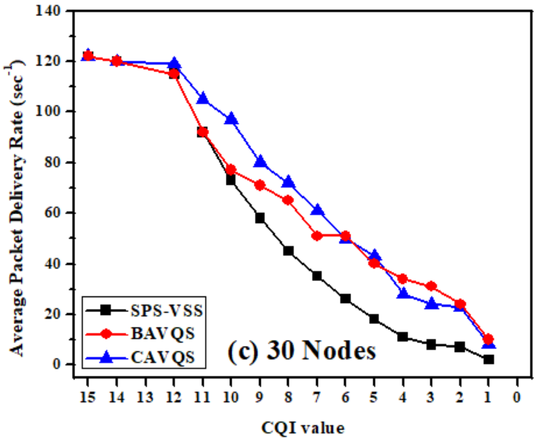


Figure 15. Number of video packets received per second for (a) 10, (b) 20 and (c) 30 node densities.

Figure 15(a-c) shows the average Packet Delivery Rate (PDR) for SPS-VSS, CAVQS and BAVQS algorithms for various CQI values. It is observed from figure 15 that PDR is constant for higher CQI values due to higher spectral efficiency with the adaptation of higher order modulation and coding schemes. Also it is depicted from figure 15 that PDR for SPS-VSS, CAVQS and BAVQS algorithms decreases for lower CQI values because of reduced spectral efficiency with the support of lower order modulation and coding schemes. Further CAVQS algorithm performs better than BAVQS and SPS-VSS even for lower CQI values due to reduction of packet sizes based on CQI values. Whereas BAVQS algorithm scales down the packet size as the PDR values drops below 80% (Table 2) leading to better than SPS-VSS.

7. CONCLUSION

Efficient video streaming over LTE networks requires adaptive video quality scaling that considers both channel conditions and buffer levels to optimize bandwidth utilization and enhance user experience. This paper introduces two novel CBRP-based transmission strategies: Channel Aware Adaptive Video Quality Scaling (CAVQS) and Buffer Aware Adaptive Video Quality Scaling (BAVQS). The performance of these algorithms was evaluated and compared against the existing SPS-VSS using the Qualnet 7.1 network simulator. Simulation results demonstrate that the proposed algorithms significantly improve throughput, reduce packet discards at the RLC transmission buffer, and enhance average delay and jitter performance. These improvements collectively contribute to a superior Quality of Experience (QoE) for multimedia users, making the proposed approach a promising solution for adaptive video streaming in LTE networks.

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