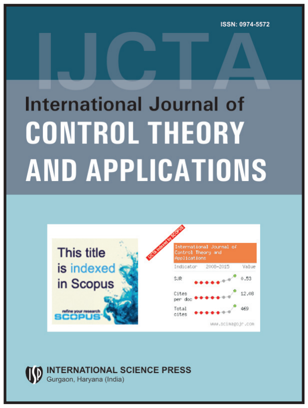

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Improved Quality of Service (QoS) in Video Streaming LTE Networks using PID Controller and Exhaustive Search Mechanism

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Abstract: Quality of Service (QoS) is an important consideration in Video streaming services over LTE networks. The video streaming services suffers from latency and other communication error due to buffering during network streaming. This often occurs due to limited bandwidth and the video quality chosen by the users. This paper aims at reducing latency and improving QoS in inefficient bandwidth allocation in video streaming on wireless networks. The problem is solved with DASH encoder that sends the packets over the streaming networks. The QoS is improved using PID controller that takes into account the past and future bandwidths. The use of such feedbacks provides improved decision feedback during the selection of packets as per the allocated bandwidth to the user. Further, the decision is improved with the use of Exhaustive Search (ES) algorithm that exploits well the multicast communication as well the encoder in a limited space. The simulated results proved that the proposed method effective in reducing the latency with high QoS in delivery huge data in wireless networks.

Keywords: DASH, LTE, Exhaustive Search, PID controller.

1. INTRODUCTION

Since decades, the multimedia services in wireless communication play a vital role in indoors. Eventually, it is deployed and implemented well in home, office and other infrastructures [1]. The influence or the success widened the multimedia services to other networks. The LTE/LTE-A cellular network based services benefits its users through unified coverage and resource allocation through centralized approach. Such networks maintains longer distance transmission, however, stable communication is still a nightmare with huge data services. Demanding of more video services from wireless networks, affects seriously the Quality of Service (QoS) of the users exist in wireless networks. This can be defined as huge data services affect the existing voice service due to limited spectrum [2]. The factors that influences the QoS in LTE networks is shown in Figure 1.

Certain communication requires high reliability and lower jitter [3], especially in case of vehicular networks. In Adhoc wireless networks, larger throughput is required due to higher bandwidth requirement [4]. LTE networks require larger data rate with high quality requirement [5]. So, diverse requirement of video streaming service with various QoS in wireless networks is a major constraint that needs to be addressed. In addition to QoS requirements, handling delay while allocating bandwidth effectively to the users is another major constraint.

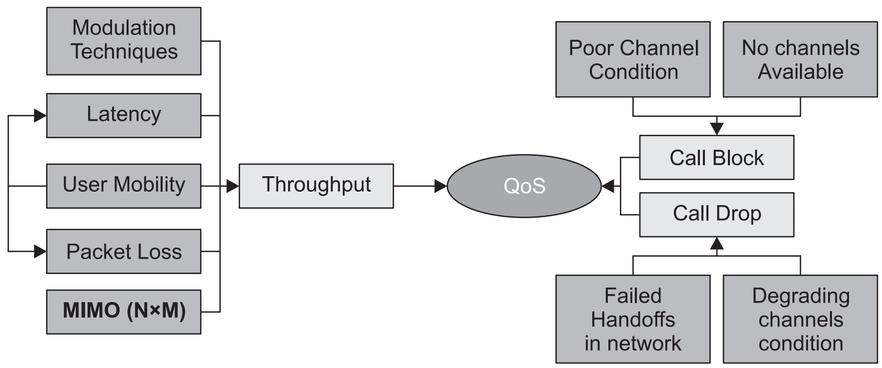

Figure 1: Influential Factors of QoS

The current research focus on reducing the delay in wireless networks during allocation of bandwidth based on the user quality requirements. This research aims at improving the QoS by possibly reducing the delay while allocation of resources and its aggregate achievable throughput. To achieve such constraint, the research proposes a unified approach that uses a proportional–integral–derivative (PID) controller that adjusts the quality of video with exhaustive search algorithm. Further, the suitable bitrate is selected effectively as per the user requirement and this encompasses the entire architecture for improved QoS and reduced latency.

The rest of this paper is organized as follows: Section 2 discusses the existing technique available in wireless networks for improving various constraints. Section 3 discusses the problem formulation and architecture of proposed method. Section 4 evaluates the claims proposed in current research over LTE networks. Section 5 concludes the paper and points the future work.

2. RELATED WORKS

This section discusses the available past researches on overcoming the mitigation of video streaming services in wireless networks. Various techniques relating to solving the constraints is discussed here.

The radio resource management scheme for improved watching time ratio for improving QoS of video streaming in wireless networks is proposed by E. Bakin et. al., (2014) [2]. This takes into account the re-buffer and buffering time spent by the user as a parameter to calculate the Quality of Experience (QoE). A joint optimization Resource Block solution is proposed by Q. Pi et. al., (2016) [9], S. Colonnese et. al., (2016) [10] proposed dynamic and cooperative approach, W. Pan, (2016) [12] proposed machine learning-bitrate estimation approach, Y. Wei et. al., (2016) [13] proposed effective cache management, starvation probability framework by Y. Xu et. al., (2016) [14], sub-optimal solutions for multi-user video streaming using matching based optimal algorithms [15] improves the user QoE.

To mitigate the interference in wireless video streaming networks, W.J. Kim et. al., (2014) [5] proposed cache induced CoMP joint processing to improve the QoS by reducing the performance degradation in networks through reduced power consumption. Similar QoS constraint saves the voice or background user from bandwidth limitation [1]. H. He et. al., (2016) [6] proposed semi-Markov decision process. This technique achieves smooth playback of video and also retains the background users. From above review, it is found that the concentration of research over QoS is very much limited in wireless networks. Hence, the present research takes a lead to improve the overall QoS in wireless environment.

D. Kim et. al., (2016) [7] proposed network aware adaptive streaming protocol to improve the video quality. This method effectively reduces the delay and transmits the video based on quality and bandwidth constraints. Such formulation is taken into account for the present research. An end to end distortion model is proposed by H. Fu et. al., (2016) [8] in LTE networks by considering the encoding rate and video quality. The formulation from [7] is applied over LTE by considering encoding rate and video quality is applied in the present research. S. Gao and M. Tao, (2016) [11] proposed DASH based video streaming over LTE. Z. Liu and Y Wei, (2016) [16] proved the inefficiency of DASH system by comparing it with content centric network. However, the use of priority assignment strategy over DASH could improve the video playback quality and reduces the average delay.

3. PROBLEM FORMULATION

Consider a LTE downlink system with n videos in a single base station and the users' shares N bandwidth (BW) or resource. Each BW consists of 12 subcarriers placed adjacent to each other. The subcarriers are denoted as $m(n)$ as m^{th} subcarrier in n BW. The achieved data rate for l user on m^{th} subcarrier in n BW is denoted as:

$$R(l, m(n)) = \mathrm{B}\log_2(1 + p\,(l, m(n))\,h\,(l, m(n)))$$

where, $p(l, m(n))$ represent the allocated power to l user on $m(n)$ and $h(l, m(n))$ is the condition of channel with instantaneous channel gain i, cell interference I and noise power spectrum density, N_0 i.e. $h(l, m(n)) = i/(\mathrm{N}_0\mathrm{B} + \mathrm{I})$. Hence, the data rate of each l user is given as:

$$\mathrm{R}(k) = \Sigma nc(k, n) \cdot \Sigma m\mathrm{R}(k, m(n)) = \Sigma nc(k, n) \cdot \Sigma m\ \mathrm{B}\log_2(1 + p\,(k, m(n))\,\mathrm{H}(k, m(n)))$$

where, c denotes the allocation of n^{th} BW to l^{th} user. Consider $c = 1$, if allocation is done else $c = 0$. However, each user occupies a 12 subcarriers, represented as $c(1, k) \leq 1$. Finally, the estimation of QoS evaluation for video streaming is achieved as

$$\mathrm{Q} = x \cdot \ln \mathrm{R}(k) + y$$

This represents the maximum QoS level of each l user with data rate $\mathrm{R}(k)$ into consideration. Additionally, a service class based scheduling is included in the LTE networks that specifies that all the users attains at least a nominal fraction of the entire network streaming performance. This is formulated as a multi-objective function that is expressed as:

$$\max \sum_{a=1}^{n} \sum\nolimits_{r \in \mathrm{RB}_a} r_{a,r}$$

$$\forall a \in \mathrm{N}: \lim_{t \to \infty} \inf\ r_a(t) \geq \gamma_a \bar{r}$$

and

$$\forall a \in \mathrm{N} : \min(l_a) \text{ and } \min(d_a)$$

subjected to

$$\forall a, b \in \mathrm{N}, a \neq b : \mathrm{RB}_a \cap \mathrm{RB}_b = \theta$$

$$\forall a \in \mathrm{N}, k \in \mathrm{K} : d_{a_k} < \mathrm{D}_k$$

$$\forall a \in \mathrm{N}, k \in \mathrm{K} : l_{a_k} < \mathrm{L}_k$$

where, a denotes the user assigned with classes, C_1, C_2, C_3 and C_4 and $a \in \mathrm{N}$. N refers to the total users such that $\mathrm{N} = 1, 2, ..., n$. The k refers to the classes (1 to 4), $\mathrm{k} = 1, 2, 3$ and 4 for transmission of data over the resource block RB_a. RB_a refers to the resource block set dedicated to users, a. $RB_a \in RB = \{1, 2, ..., r\}$. $r_{a,r}$ denotes the achievable throughput by i users over r blocks. γ_a is the minimum fraction of the achievable total throughput (average rate) for a users ≥ 0. L_k is the maximum packet loss target rate for its respective class, k and D_k is the maximum packet target delay for its respective class, k. The d_{ak} is the measured packet delay for users a, I_{ak} is the measured packet loss factor for a users and θ is the fairness throughput target. These objectives provides maximum throughput with minimized packet loss and delay ratio. The resource blocks set assigned/ dedicated for users.

4. PROPOSED FRAMEWORK

The proposed method uses PID controller to smoothen the selection of packets as per the requirement of users'. The main aim of the research is to effectively select the packet as per the limited bandwidth in the network infrastructure. To achieve this, an effective combination of Exhaustive Search (ES) mechanism with PID controller is used.

A. Quality Adaptation for QoS

Most techniques available perceives the network infrastructure, however, information related to bandwidth during network handover is very much limited. Otherwise, the research over this constraint is limited due to lack in knowledge of improving the constraint. Having information about network infrastructure and bandwidth information, quality level has to be adjusted in terms of available bandwidth and buffer status of client.

The proposed method works on selection of quality levels based on the available lesser information and reduces the latency of the current network. To achieve this, initially, the information on available bandwidth and network delay has to be known. Depending upon the bandwidth, the network delay is reduced with selection of past and future bandwidth information using PID feedback mechanism. The network congestion rate is check periodically by the ES mechanism and once if the network congestion is found to be nil or lesser than the threshold, no further control of PID is required during network handover in LTE networks.

The PID controller has two parameters represented as *setpoint* (S) and *process variable* (P). The P represents the current bandwidth information and S represents the target BW allocation to achieve improved QoS. This is achieved using minimization of error between P and S and that could improve well the objective of the current research (i.e. reducing latency). The output of the controller is represented as:

$$u(t) = k_P \cdot e(t) + k_I \cdot \int_0^t e(\tau)\, d\tau + k_d \cdot de(t)/dt$$

where, $e(t)$ is the error variable between P and S at time t. kp, ki and kd (tuning parameters) are the proportional, integral and derivative gain. The formula is used to dynamically adjust P towards S.

The P variable is considered as interval ratio (ir) that possesses push and play time. Push time denotes sending data packets to video stream and play time denotes the time for the data packets to be played. This is checked periodically with two intervals actual, A(ir) and check intervals, C(ir). Consider an ideal situation, where A(ir) = C(ir) i.e. data pushed from server is play over time = C(ir). If A(ir) is less than C(ir), the data is pushed less to client due to limited BW and vice versa. Thus, ir = A(ir)/C(ir) = 1, which represents the mismatch between BW and current R(k) being pushed. This parameter helps suitably in maintaining the balance to achieve smooth playback of the video.

The PID is used to calculate the output variable using tuning parameters of the error between P and S. The P considered here is ir and the S is set as 1 and the parameters of PID are changed accordingly.

$$u_t = k_P \cdot e_P(t) + k_I \cdot e_I(t) + k_D \cdot e_D(t)$$

$$e_P(t) = A(ir, t)/C(ir, t)$$

$$e_I(t) = \text{long\_A}(ir, t)/\text{long\_C}(ir, t)$$

$$e_D(t) = e_P(t)/e_P(t-1)$$

$$\text{long A}(ir, t) = \Sigma\, A(ir, \tau)$$

$$\text{long C}(ir, t) = \Sigma\, C(ir, \tau)$$

where τ varies from T_l to t and $e_P(t)$, $e_I(t)$ and $e_D(t)$ represent the observed values for each tuning parameters, P, I, D, respectively. T_l gives the rate of previous quality changes. The $R(k)$ is used to find the data rate available for transmission and further determination of quality level suitable for transmission.

The integral parameter is recorded as the initial packet is sent and the BW changes significantly. Hence, the integral part is recorded for longer time and fails in reflecting the network congestions. So, problem occurs in selecting better quality, since the network slows down. This problem is rectified using longer interval recording or the time interval between T_l and the interval calculation. Finally, selection of quality rate and the periodical checking of network congestion are further defined in the algorithm.

Algorithm 1: Exhaustive Search Algorithm for QoS
Previous packet is sent;
$C(ir) = t_c - t_{lc}$
$A(ir) = t_{cm} - t_{lcm}$
long $C(ir)$ += $C(ir)$
long $A(ir)$ += $A(ir)$
Calculate $e_P(t)$, $e_I(t)$ and $e_D(t)$
Output = $k_P \cdot e_P(t) + k_I \cdot e_I(t) + k_D \cdot e_D(t)$
nD = Output × cD
$nl = g\ (n\text{D}, l\text{D})$
if Ql is modified **then**
long $C(ir) = 0$
long $A(ir) = 0$
$c\text{D} = l\text{D}(n\text{l})$
end if
Check nc during handover, $r(k)$ and BW limit
S_c = Choose best subcarrier(min_power, data_rate)
O_u = Select the optimal user (quality_level) and normalize it
Send packets (nl, S_c, O_u) // packets will be sent based on the value of nl

where,

t_c – current time

t_{lc} – last check time to calculate the check interval

t_{cm} – current media time

t_{lcm} – last check media time to calculate the actual interval

long C(*ir*) – long check interval

long A(*ir*) – long actual interval

nl – new level

g – get level

*n*D – new data rate

*l*D – data rate level

Q*l* – quality level

Output – output of PID controller

*c*D – current data rate

NC – network congestion

The algorithm is designed to vary the quality rate as per the bandwidth available and network congestion rate. The algorithm predicts the BW rate to be high or low for subsequent change of quality level in future. This is done to avoid unnecessary spikes in quality levels. The algorithm is designed to operate on *n* quality levels in a video streams for *l* users'. This helps in enabling the faster quality drop that suitably avoids latent playback/ pause during sudden declination/inclination in BW.

5. THROUGHPUT CONSIDERATION

The computation of QoS parameter additionally leads to the computation of the throughput using the ranks of parameters like past throughput, loss, delay, queue depth and priority. These QoS metrics are considered as a normalized ranking function from [17] and the combination of these five parameters leads to the formulation of the throughput calculation. The ranking function of each parameter (*p*) is gives a weighted rank value bounded within its limit [0, P_a].

The proposed framework with QoS calculation with PID controller is used for the calculation of the throughput value. This is shown in Figure 2. Here the entire scheduling process for the throughput calculation is carried out with Exhaustive search mechanism. The calcualtion of the normalized value for each five parameters is calculated from:

$$r.f(x_a, p_a) = p_a.\tanh(x_a)$$

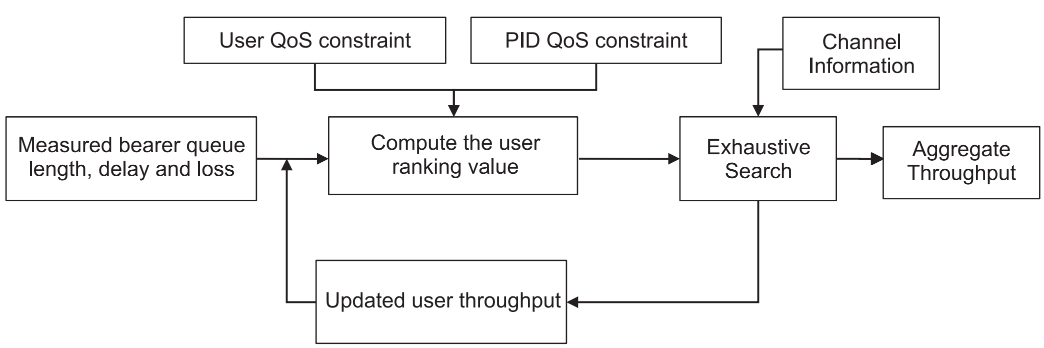

Figure 2: The Throughput Calculation

where,

p_a = Adjustable weight of each QoS metric of *a* users

x_a = Normalized weighted value of each QoS metric *a* users

x_a = (value measured from *a* user) / (QoS constraint of *a* user)

6. RESULTS

The proposed method seems to be an effective mechanism to calculate the quality levels. The ratio based PID controller is very simple and effective means to choose P and S. The tuning parameters are normalized around unity and R(t) is seen as the weighted average of all the three tuning parameters. The controller output is always nearer to one, which reflects the effective use of tuning parameters over achievable data rates. The proposed method is compared with Time Ratio method [1], Cache-induced hybrid CoMP method [2] and Spectrum Sensing and Access method [12], to prove its effectiveness. The network topology used for testing is shown in Figure 3.

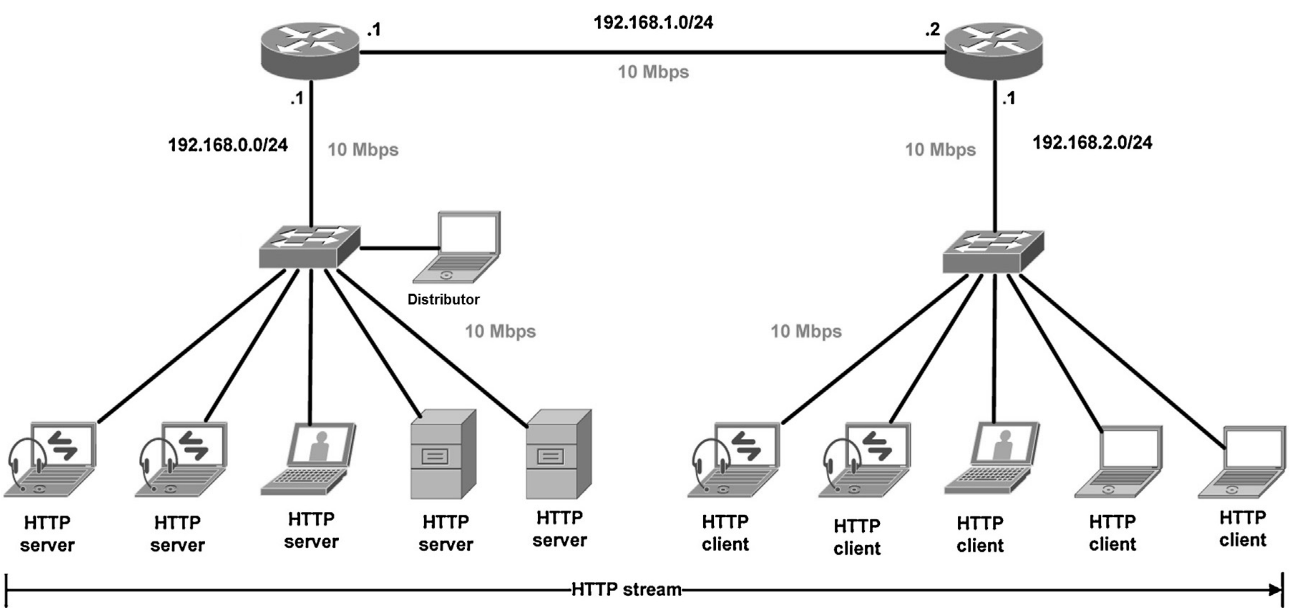

Figure 3: Topology for QoS Testing and throughput performance in multimedia networks

The aggregate throughput parameter is calculated as a factor that contains the ranks of existing throughput, loss, delay, queue depth and priority.

$$r.\ \Sigma f(x_a, p_a) = \forall \text{a } \{\text{past\_throughput, loss, delay, queue depth and priority}\}$$

The achievable throughput is estimated through the parameters like past throughput, loss, delay, queue depth and priority. The past throughput is achieved using the existing history of each user.

From Figure 4, it is found that the proposed PID controller outperforms well when compared with the other systems. The proposed method is compared with other conventional methods in terms of total number of users in LTE networks and aggregate throughput. Thus, the throughput of the proposed method is higher than other methods. The proposed system is evaluated with ES algorithm with and without PID controller. It is found that the method with both PID and ES algorithm proved to have higher throughput than other methods. The resouce block allocated to each user by the proposed method is much higher than the conventional techniques.

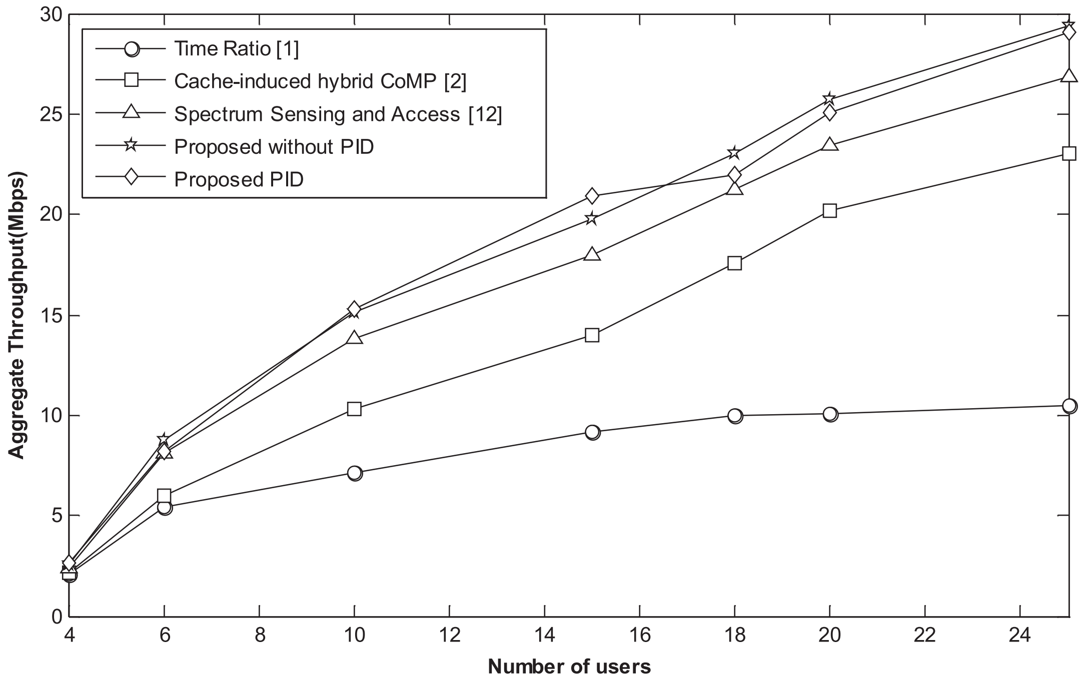

Figure 4: Comparison of Aggregate throughput for 25 Users

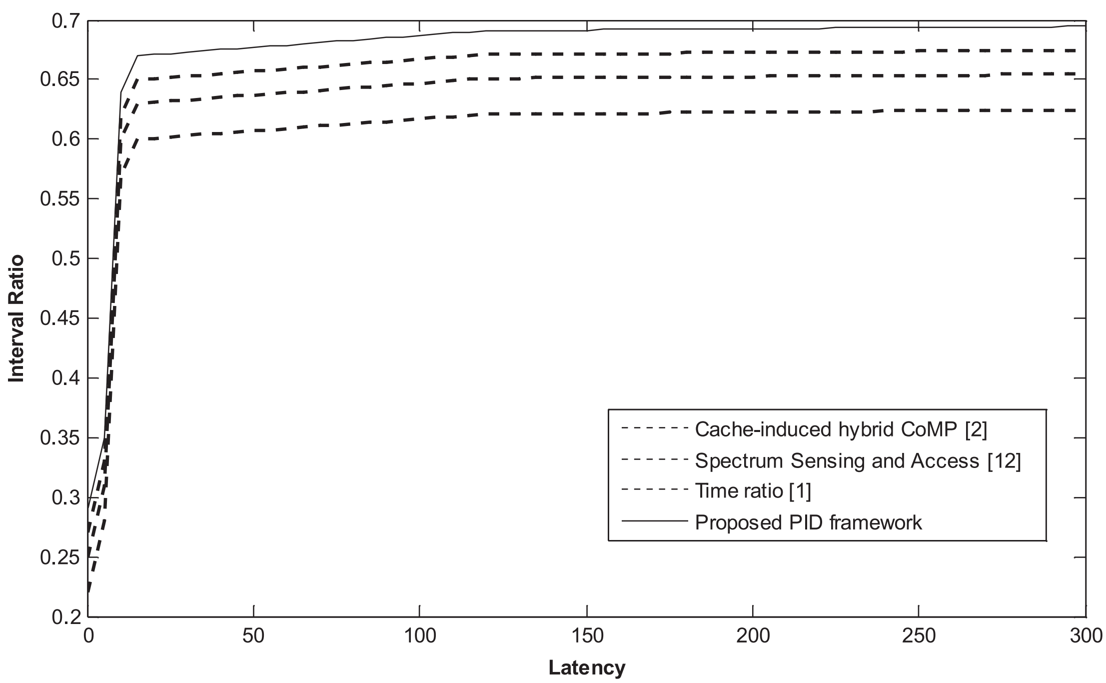

Figure 5: Interval Ratio vs. Latency without Exhaustive Search Algorithm

From Figure 5, it is found that the proposed method posses higher interval ratio as the latency (latency = 1/ delay) reduces the interval ratio increases. The proposed method is compared with other conventional technqiues without ES algorithm.

From Figure 6, it is clear that the proposed method with ES algorithm is higher than the interval ratio in Figure 5. Also, ES algorithm improved the dynamicity of the interval ratio, which improves well the level of *ir* to 0.8. This value is much higher than the other methods and thus proves the effectivness of the proposed ES algorithm with PID controller over DASH system in LTE networks.

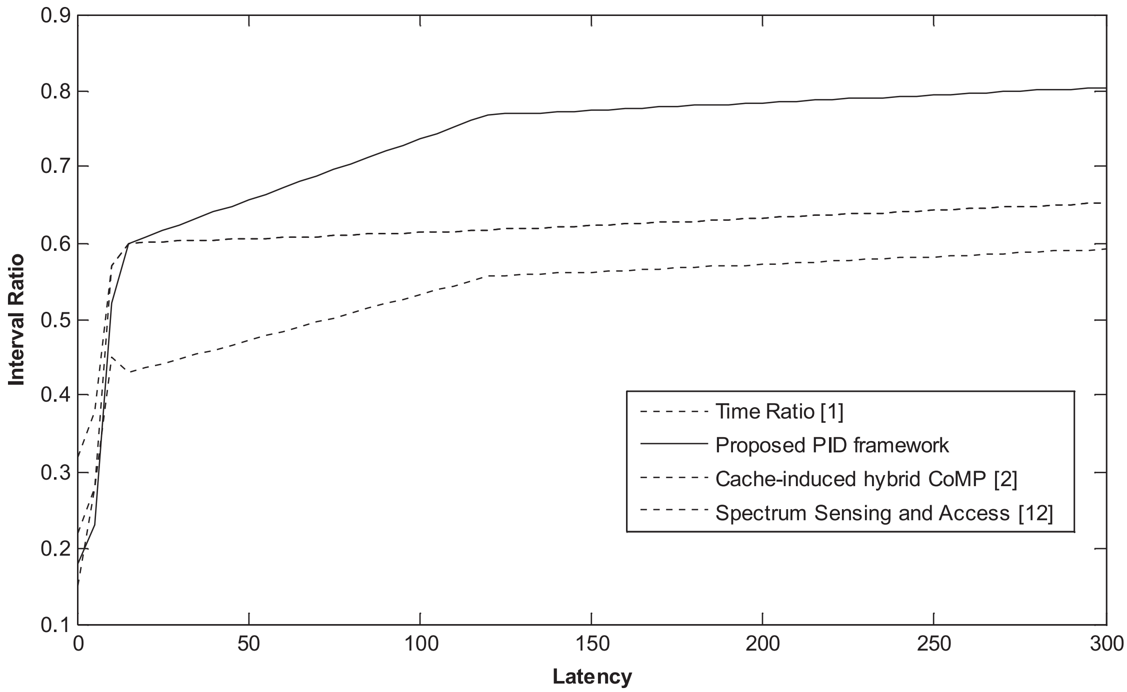

Figure 6: Interval Ratio vs. Latency with Exhaustive Search Algorithm

This reduced latency with improved interval ratio improved the seamless watching experience of the users in LTE networks. The users could experience watching the video without buffering through higher interval ratio.

7. CONCLUSIONS

The present research finally fulfils the seamless watching experience of the video to the user with reduced latency. The user experiences high video quality without waiting or stopping. The use of PID controller and ES algorithm helped in achieving the objective of the proposed method. Here, the DASH system is set as an application, where the proposed method is implemented over it to achieve lesser latency. The selection of data rate as per the evaluated quality levels improves the playback of video at user end with reduced delay. Thus, the system is proved to be effective over other QoS mechanism in streaming servers. Further, the system could be implemented over other CDMA, OFDM networks apart from video streaming services. Also, the system can be used in other scalable video applications to improve the quality rate of videos being streamed.

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