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VIDEO STREAMING PERFORMANCE UNDER WELL-KNOWN PACKET SCHEDULING ALGORITHMS

[1]Huda Adibah Mohd Ramli, [2]Kumbesan Sandrasegaran, [3]Riyaj Basukala, [4]Rachod Patachaianand and [5]Toyoba Sohana Afrin

[1,2,3,4,5]Faculty of Engineering and Information Technology, University of Technology, Sydney, Australia

[[1]HudaAdibah.MohdRamli, [2]kumbes, [4]rpatacha]@eng.uts.edu.au

[3]Riyaj.Basukala@uts.edu.au

[5]ToyobaSohana.Afrin@student.uts.edu.au

[1]International Islamic University Malaysia (IIUM), Kuala Lumpur, Malaysia

ABSTRACT

Video streaming is becoming increasingly popular among the wireless users. However, supporting video streaming over the wireless networks is not an easy task due to the dynamic radio propagation environment, limited radio resources as well as Quality of Service (QoS) requirements of the video streaming that need to be satisfied at acceptable levels. Most studies proposed to support video streaming are computationally expensive to be used in Orthogonal Frequency Division Multiple Access (OFDMA) based wireless IP networks. This paper evaluates video streaming performance under three well-known algorithms that are more practical to be used in the OFDMA based wireless IP networks due to their reduced complexity. It is demonstrated via computer simulation that Proportional Fair (PF) algorithm outperforms other well-known algorithms by providing video streaming QoS at acceptable levels whilst maximizing cell throughput.

KEYWORDS

Video streaming, Wireless IP Network, Packet scheduling, OFDMA, LTE

1. INTRODUCTION

Significant progresses in both wireless networking and video coding technologies have enabled video streaming to be supported over the wireless networks. An actual video stream contains data of a large size. The video stream is encoded into a smaller variable bit rate stream based on Moving Picture Experts Group (MPEG) or H.26x standards to reduce the bandwidth requirements [1]. The MPEG and H.26x are the two prominent standards developed by International Organization for Standardization/Information Centre (ISO/IEC) and International Telecommunication Union (ITU) standardization bodies.

There are multiple Groups of Pictures (GoP) within an encoded video stream. A sequence of frames starting with an I frame and all frames prior to the subsequent I frame constitutes a GoP [2]. An I frame represents a fully specified picture. It functions as a reference frame within a GoP. It is encoded without reference to any other frames except to itself [3]. A P frame contains only changes in the information from the previous frame. It is encoded by predicting the changes in the information in the previous I or P frame within its GoP. On the other hand, a B frame is encoded with reference to both previous and sub-sequent I or P frames. It contains the changes in the information from previous and subsequent frames.

Each frame within a GoP has different importance and is highly co-related due to the encoding. Therefore, it can be concluded that the I frame has the highest importance followed by the P and B frames. A lost of a reference frame within a GoP results in the lost of other dependent frames within the same GoP [4].

OFDMA is a multi-carrier access technology that divides the wide available bandwidth into multiple equally spaced and mutually orthogonal sub-carriers [5],[6]. Due to its robustness to inter-symbol interference and immunity to frequency selective fading [7], the OFDMA technology has been considered as a promising candidate for future wireless IP networks. Packet scheduling is one of the key RRM functions within the OFDMA based wireless IP networks for efficient radio resource utilizations and satisfying the QoS requirements of multimedia users. Moreover, the OFDMA technology allows packet scheduling to be performed in both time and frequency domains, thus enabling multi-user diversity to be exploited rather than combated.

Given that the video streaming service is becoming increasingly popular, a good portion of the radio resources needs to be allocated to provide support for the video users. However, majority of the packet scheduling studies that are proposed to support the video streaming are computationally expensive to be used in the OFDMA based wireless IP networks. Therefore, this paper evaluates video streaming performance under three well-known algorithms that are more practical to be used in the OFDMA based wireless IP networks due to their reduced complexity. This paper contributes to the identification of a well-known algorithm that is best suited for video streaming over the OFDMA based wireless IP networks.

The rest of this paper is organized as follows: Section 2 reviews on the related studies followed by detailed descriptions of the system model in Section 3. Discussions on the well-known packet scheduling algorithms and metrics used for performance evaluation are discussed in Section 4 and Section 5 respectively. Section 6 contains environment and results of the performance evaluation whereas conclusions are summarized in Section 7.

2. RELATED STUDIES

Adapting the wireless networks that have dynamic radio propagation environment and limited radio resources to satisfy the QoS requirements of video users at acceptable levels remains a challenge that needs to be addressed when supporting the video streaming over the OFDMA based wireless IP networks [8-11]. A number of packet scheduling studies have been proposed in the literature to address this challenge. The authors in [12-16] considered Channel Quality Information (CQI), average throughput and packet delay information of each user when scheduling video packets over the OFDMA based wireless networks. It is assumed in these studies that each video frame has equal importance, which is not typically the case in a realistic scenario.

To satisfy the QoS requirements of video users, the studies in [8, 17-19] considered video contents, commonly referred as content-aware, in their proposed algorithms. The proposed algorithms prioritize a video user on the basis of frame importance/deadline and its contribution towards improvement of the perceptual video quality. These studies assumed that a deadline is associated to each frame. If a frame arrives after the deadline, the frame and all other dependent frames are discarded. The overall goals of these studies are to improve perceptual video quality (minimize distortion) while minimizing the number of lost frames.

A more realistic video streaming scenario that utilized a playback buffer at the user is modelled in [20-23]. Under this scenario, the start of video playback is delayed until the total number of frames within the playback buffer exceeds a buffering threshold. This delay is referred as initial buffering delay and it aims to ensure a continuous video playback. The video and the wireless networks characteristics may lead to interruptions during the video playback. The video playback resumes at the same position where the interruption occurs after re-buffering takes place. Re-buffering is a process that re-fills the playback buffer into the specified buffering threshold.

The packet scheduling algorithms proposed to support video streaming under this realistic scenario [20-22] take video contents into consideration, which are similar to the studies in [8, 17-19]. However, on the contrary to the [8, 17-19] that mainly focused on satisfying the QoS requirements of the video users, the studies in [20-22] attempted to find a trade-off between maximizing throughput/capacity and satisfying the QoS requirements of video users.

There is always a trade-off between performance gain and complexity. A content-aware packet scheduling algorithm in the OFDMA based wireless IP networks will be computationally expensive with increasing number of users and the number of available radio resources [20], therefore not preferable from an implementation point of view [24]. In order to reduce complexity, this paper evaluates the video streaming performance under three well-known algorithms (i.e. Maximum-Rate (Max-Rate) [25], Round Robin (RR) [26] and PF [27]) over an OFDMA based wireless IP network. These algorithms are content un-aware packet scheduling algorithms which have been proposed for use in single-carrier wireless networks.

3. SYSTEM MODEL AND DESCRIPTIONS

A downlink Long Term Evolution (LTE) system consisting of a base station and N video streaming users are considered to represent an OFDMA based wireless IP network. It is assumed in this paper that (i) the serving base station buffers video streams that are received from the streaming server through a lossless and a high bandwidth backbone network [28]; (ii) each video frame of a variable size is segmented into groups of packets of a fixed size and (iii) all packets belonging to a frame need to be completely received for successful decoding at a user. The radio resource that is available to be shared by all users in the downlink LTE system is known as a Resource Block (RB). A RB is made up of 12 sub-carriers (180 kHz total bandwidth) of 1 ms duration [29],[30]. Moreover, a transmission time interval (TTI) of 1 ms duration is defined for packet scheduling in the downlink LTE system.

A user that has data to receive estimates its instantaneous Signal-to-Interference-Noise-Ratio (SINR) on each RB, converts it into a CQI value with the target block error rate (BLER) to be less than 10 % [29] and reports each CQI value to the serving base station. It is assumed in this paper that the CQI reporting is performed in each TTI and on each RB. However, these CQI values are available to be used by the packet scheduler after a certain delay.

The packet scheduler selects a user to receive its packets based on a packet scheduling algorithm on each RB and in each TTI. Note that a RB may only be allocated to a single user in a TTI but a user may be allocated variable number of RBs in the TTI. The packets to be transmitted to a user in a TTI are packed into one Transport Block (TB). The size of a TB depends on the supportable Modulation and Coding Schemes (MCSs) on each RB that is allocated to the user. The MCS is determined from the CQI value reported by the user.

Data transmissions over the wireless networks are prone to error. Therefore, the LTE system uses a stop-and-wait Incremental Redundancy Hybrid Automatic Repeat Request (IR HARQ) technique to recover wireless transmission errors. This is done by multiple retransmissions and soft combining of a corrupted TB. This leads to an improvement in the decoding efficiency at a user's soft buffer. If a TB of a user is corrupted during transmission, the user sends a Negative Acknowledgement (NACK) to the packet scheduler indicating retransmission of the TB is required. The packet scheduler placed all packets belonging to the corrupted TB at the front of the user's buffer. This indicates that the packets of a user that require retransmission are given a higher priority compared to its packets that are waiting for the first transmission. The packet scheduler drops all packets that have exceeded maximum number of retransmission, contrary to [21] that retransmit these packets until they are correctly received.

All packets that arrive at a user are stored into the playback buffer, which resides at the user's application layer, and re-constructed for video playback. These packets are packed into their associated frame. Some packets belonging to a frame may be lost. Therefore, all packets belonging to the frame are dropped and the frame is considered as a lost frame. Each video frame is decoded before they can be played back. Some frames cannot be decoded if a reference frame is lost. These frames are dropped at the user. The video frames are played back in sequence and at a fixed rate. If a frame within a sequence is lost, a simple error concealment method is used to estimate the missing information associated to the lost frame. This is done by duplicating the information of the latest successfully decoded frame into the position where the lost frame occurs [28]. The video frames are being played back in sequence and at a fixed rate.

4. WELL-KNOWN PACKET SCHEDULING ALGORITHMS

The Max-Rate, RR and PF are the well-known packet scheduling algorithms that are more likely to be used in the OFDMA based wireless IP networks due to their reduced complexity. These algorithms were proposed to support best effort services in the single-carrier wireless networks and aimed to maximize throughput and satisfy fairness of these services.

The Max-Rate is an algorithm that always selects a user having a good channel quality. On the other hand, a user having a poor channel quality is less likely to be given any opportunity to receive its packets unless its channel quality improves. The Max-Rate algorithm is a good candidate for throughput maximization but is not capable of satisfying fairness between the users. The mathematical expression for the Max-Rate algorithm is given as follows:

$$k = \arg\max\ r_i(t) \quad i \in N \tag{1}$$

where $r_i(t)$ is the total number of supportable bits (based on MCS) of user i at time interval t and N is the total number of users.

Given that fairness has been an issue in the Max-Rate algorithm, the RR algorithm is proposed to address this situation. The RR algorithm gives equal opportunity to each user to receive its packets in a cyclic fashion. Though the RR algorithm is able to considerably improve fairness performance between the users, the throughput performance is significantly degraded.

The PF algorithm provides a better trade-off between throughput maximization and fairness satisfaction. This algorithm is originally proposed to support non-real time services in Code Division Multiple Access (CDMA) system. The PF algorithm selects user k to receive its packets at time interval t based on the following equations:

$$k = \arg\max \frac{r_i(t)}{R_i(t)} \quad i \in N \tag{2}$$

$$R_i(t) = \left(1 - \frac{1}{t_c}\right) R_i(t-1) + \frac{1}{t_c} * r_i(t-1) \quad i \in N \tag{3}$$

where $r_i(t)$ and $R_i(t)$ are the total number of supportable bits (based on MCS) and the average throughput of user i at time interval t, t_c is a time constant and N is the total number of users.

The t_c value determines between maximizing throughput and satisfying fairness in the PF algorithm. The PF algorithm behaves similar to the Max-Rate algorithm if a higher t_c value is used while similar to the RR algorithm for a lower value of t_c. A t_c value of 1000 ms as proposed in [27] that can provide a trade-off between throughput and fairness is used in this paper.

Packet scheduling in the single-carrier wireless networks allocates all the available radio resources to a single user in each scheduling interval. Therefore, some modifications are made to enable these well-known algorithms to support packet scheduling in the OFDMA based wireless IP networks (specifically focusing on the downlink LTE system). Since there are multiple RBs available to be shared by the users in each TTI, these algorithms select user k to receive its packet on RB j based on equation (4) for Max-Rate and equations (5-7) for PF. Similarly the RR algorithm is modified such that a user that is to receive its packet on RB j is selected in a cyclic fashion.

$$k = \arg\max\ r_{i,j}(t) \quad i \in N \tag{4}$$

$$k = \arg\max\ \frac{r_{i,j}(t)}{R_i(t)} \quad i \in N \tag{5}$$

$$R_i(t) = \left(1 - \frac{1}{t_c}\right) R_i(t-1) + \frac{1}{t_c} * rtot_i(t-1) \quad i \in N \tag{6}$$

$$rtot_i(t-1) = \sum_{j=1}^{\max RB} r_{i,j}(t-1) * \Delta_{i,j}(t-1) \quad i \in N \tag{7}$$

where $r_{i,j}(t)$ is the total number of supportable bits (based on MCS) on RB j of user i at time interval t, $R_i(t)$ is the average throughput of user i at time interval t, $rtot_i(t)$ is the total number of supportable bits on the RBs allocated to user i at time interval t-1, t_c is a time constant, N is the total number of users and *max RB* is the total number of RBs. The variable $\Delta_{i,j}(t\text{-}1)$ contains a value between 0 and 1 in which $\Delta_{i,j}(t\text{-}1)$=1 if RB j is selected for user i at time interval t-1 and 0 otherwise.

5. PERFORMANCE METRICS

The video streaming performance under the three well-known algorithms is evaluated on the basis of minimum Mean Opinion Score (MOS), Freezing Delay Ratio (FDR) and cell throughput. The MOS and FDR are the metrics that are related to the QoS requirements of video streaming whereas the cell throughput is a network related performance metric. Detailed descriptions of each metric are provided as follows:

The minimum MOS represents both the perceptual video quality (measured in terms of Peak-Signal-to-Noise Ratio, PSNR [31]) as well as the fairness experienced among all video users. The PSNR gives the difference in terms of peak signal between the source and the re-constructed video frame (i.e. the source video frame is the encoded frame located at the serving base station before it is streamed to the user). It has the following mathematical expression [32]:

$$PSNR_{i,m} = 20\log 10 \frac{255}{D_{i,m}} \quad (8)$$

$$D_{i,m} = \sqrt{\frac{1}{XY}\sum_{x=0}^{X-1}\sum_{y=0}^{Y-1}\left[F_{i,m}(x,y) - f_{i,m}(x,y)\right]^2} \quad (9)$$

where $PSNR_{i,m}$ is the PSNR value at the mth frame of user i, X and Y are resolution in pixels, $F_{i,m}(x,y)$ and $f_{i,m}(x,y)$ are the source and reconstructed pixel's values at the mth video frame at position (x, y) of user i.

It is assumed that each user requests a total of F frames throughout a video session. A video session starts at the beginning of the simulation ($t=1$ *ms*) and continues until the total simulation time ($t=T$ *ms*). The total number of frames that are able to be played back throughout the video session varies among the users. The mean PSNR of each user (averaged over F frames) is determined as follows:

$$mean\ PSNR_i = \frac{1}{F_i}\sum_{m=1}^{pf_i} PSNR_{i,m} \quad (10)$$

where $PSNR_{i,m}$ is the PSNR value at the mth frame of user i, $mean\ PSNR_i$ and F_i are the mean PSNR and the total number of frames requested by user i and pf_i is the frame number of user i that is currently being played back when the session ends.

The minimum MOS, as given in Table 1, represents a user having the lowest mean PSNR among other users. It ensures that equal perceptual video quality is experienced among the video users. A low minimum MOS indicates that at least one user is not satisfied whereas maximization of the minimum MOS ensures fairness and high perceptual video quality are experienced among the video users.

Table 1. Mean PSNR to MOS Mapping [33]

PSNR (dB)	***MOS Value***	***Quality***
< 20	1	Bad
20 – 25	2	Poor
25 – 31	3	Fair
31 – 37	4	Good
> 37	5	Excellent

A user will not be satisfied if it has to wait longer to start or resume its video playback. Therefore, it is necessary to minimize the FDR throughout the video session. The freezing delay, as discussed in [23], is the total delay in the video playback due to (i) scheduling and propagation delay (ε); (ii) initial buffering delay (α) and (iii) re-buffering delay (β).

An example of freezing delay experiences by a user throughout a video session is illustrated in Figure 1. It is shown in the figure that, after a request for a video stream is made by the user, it takes ε ms for the first bit of the video stream to arrive and being stored into the user's playback buffer. The video playback starts after frames of α ms duration are available within the playback buffer (i.e. total duration of frames equals or greater than the buffering threshold). The total duration of frames within the playback buffer (T_{pb}) can be determined as follows:

$$T_{pb} = \frac{F_{pb}}{\Omega} \tag{11}$$

where Ω is the frame rate (in fps) and F_{pb} is the total number of frames within the playback buffer.

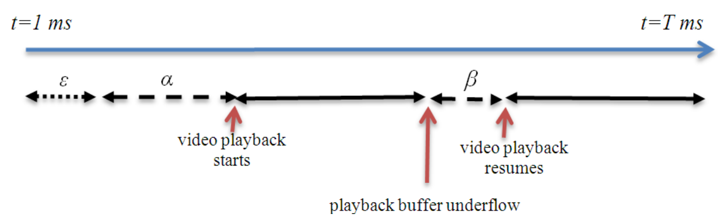

Figure 1. Example of freezing delay during a video session.

Video playback is interrupted and re-buffering takes place if the total duration of frames within the playback buffer falls below $\alpha - \beta$ ms. The FDR is defined in this paper as the mean of the ratio of total freezing delay of each user to the total simulation time as given as folows:

$$FDR = \frac{1}{N}\sum_{i=1}^{N}\frac{Df_i}{T} \tag{12}$$

where Df_i is the total freezing delay of user i, T is the total simulation time and N is the total number of users.

Finally, the cell throughput is defined as the total number of bits correctly received by all users per second. The cell throughput is measured at the base station. It is mathematically expressed as:

$$cell\ throughput = \frac{1}{T}\sum_{i=1}^{N}\sum_{t=1}^{T} tput_i(t) \tag{13}$$

where $tput_i(t)$ is the total size of correctly received packets (in bits) of user i at time interval t, T is the total simulation time and N is the total number of users.

6. Simulation Environments and Results

A computer simulation using C++ platform is used to evaluate the minimum MOS, FDR and cell throughput performances of the three well-known algorithms for varying number of users and buffering thresholds. The environments and results of the computer simulation are discussed in the following sub-sections.

6.1. Simulation Environment

A single hexagonal cell scenario of 5 MHz bandwidth with 25 RBs and 2 GHz carrier frequency is modelled. The base station has a fixed location at the centre cell and it is assumed that equal transmit power (43.01 dBm total base station transmit power) is used on each RB. The cell contains 5 – 20 video streaming users that are moving within the cell radius of 200 m at a constant speed of 10 km/h in random directions. These users are uniformly located within the cell. A user is wrapped-around whenever it reaches the cell edge [34]. The Cost-231 HATA model for an urban environment [35], a Gaussian log-normal distribution with 0 mean and 8 dB standard deviation [36] and a frequency-flat Rayleigh fading model [37] are used to model the radio propagation channel. A total of 16 CQI levels as defined in [29] are used. The CQI and HARQ reporting are modeled error free with 3 ms CQI delay and 4 ms HARQ delay. The maximum number of retransmissions is limited to 3 and the threshold for re-sequencing timer is set to 500 ms.

Three video streams in a 352x288 resolution which are downloaded from a publicly available video traces [2] are used. Video trace is a simple file that contains number, type/importance, decoding/video playback deadline, size and quality of each video frame [38, 39]. The video streams are encoded with a frame rate of 30 fps (frames per second). One GoP contains 16 frames with the IBBBPBBBPBBBPBBB sequence. A user may only request one random video stream throughout its session and the request arrives at the beginning of the simulation. The requested video stream may start at any frame number as long as it is an I frame. Only 300 frames out of the total frames are simulated for each user. The maximum size of the playback buffer is infinite. The video playback is interrupted if there are less than 5 frames remaining within the playback buffer. The buffering thresholds vary between 266.667 ms – 800 ms duration [18].

6.2 Results –Vary Number of Users

This sub-section evaluates the impacts of varying number of users towards minimum MOS, FDR and cell throughput performances under the Max-Rate, RR and PF algorithms. The buffering threshold of each user is fixed at 800 ms. Figure 2 shows the minimum MOS with increasing number of users. As expected the minimum MOS degrades with increasing number of users. The figure shows that the PF algorithm is able to maintain a good perceptual video quality and satisfy fairness for a higher number of users compared to the Max-Rate and RR algorithms. When the number of users equals to 10, the PF algorithm is able to maintain a good perceptual video quality (minimum MOS=4) for all 10 users. However, at this number of users, at least one user is experiencing a fair (minimum MOS=3) and a poor (minimum MOS=2) perceptual video quality under the RR and Max-Rate algorithms respectively.

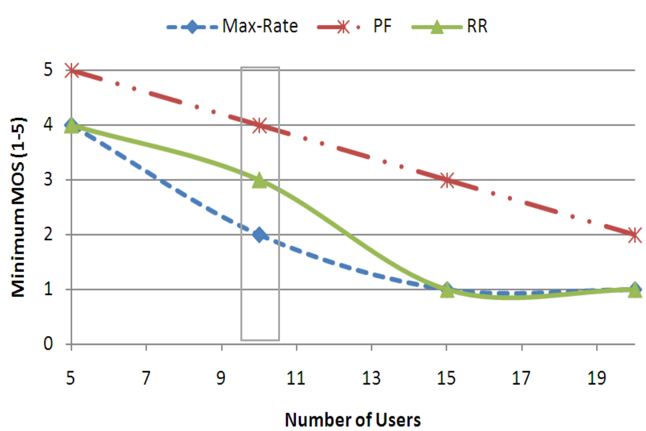

Figure 2. Minimum MOS vs. number of users

The FDR performance with increasing number of users is shown in Figure 3. The FDR degrades with increasing number of users and it is demonstrated within the figure that the PF is having the best FDR performance followed by the RR and Max-Rate algorithms. The worst FDR performance under the Max-Rate algorithm is mostly due to the users having poor channel qualities that have to wait longer before they can start or resume their video playbacks.

Moreover, over-allocation of the available RBs to the users with good channel qualities does not benefit the Max-Rate algorithm as the video is played back at a fixed rate.

The RR algorithm that allocates equal share of transmission time to all users achieves a better FDR performance compared to the Max-Rate algorithm. However, it is not possible for this algorithm to minimize the freezing delay among the users as the channel quality of each user is different. The PF algorithm achieves the best FDR performance for selecting users having relatively good channel qualities whilst ensuring that all users receive equal amount of packets based on their average throughput update (denominator in (5)). This minimizes the delays taken by the users to start or resume their video playbacks.

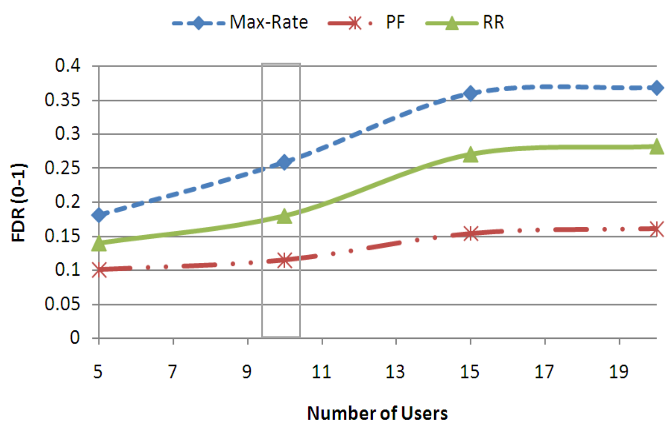

Figure 3. FDR vs. number of users

The cell throughput performance under the three well-known algorithms with increasing number of users is shown in Figure 4. With increasing number of users, the cell throughput achieves under the PF outperforms the Max-Rate and RR algorithms. The PF algorithm always selects the users having relatively good channel qualities for transmissions. By delaying transmissions of some users, there is a possibility that the channel qualities of these users improve over time. This allows the users to receive their packets on the RBs with better channel qualities, thus improving the cell throughput under the PF algorithm.

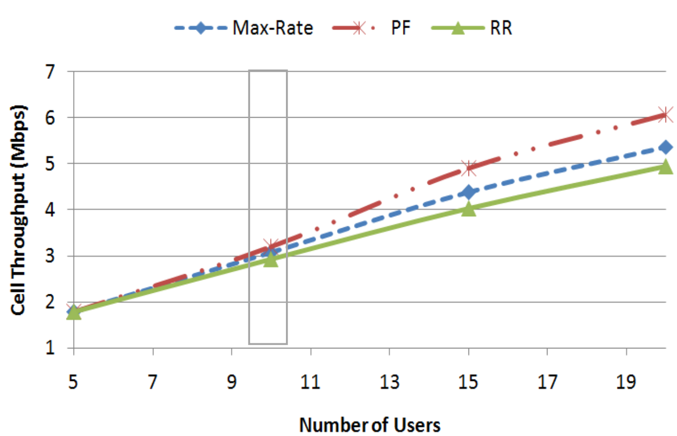

Figure 4. Cell throughput vs. number of users

6.3 Results –Vary Buffering Thresholds

The impacts of varying buffering thresholds while setting the number of users fixed at 10 users are evaluated in this sub-section. The minimum MOS under the three well-known algorithms is shown in Figure 5. The figure demonstrates that the minimum MOS under the RR and PF algorithms remain the same within the buffering thresholds ranging from 266.667 – 800 ms. However, the minimum MOS under the Max-Rate algorithm degrades when the buffering threshold is set to 800 ms. The Max-Rate algorithm deprives the users having poor channel qualities from receiving their packets. With increasing buffering thresholds, only small number of the requested frames may be able to be played back since the users may have to wait longer to start or resume their video playbacks. This leads to the degradation of the minimum MOS at a higher buffering threshold.

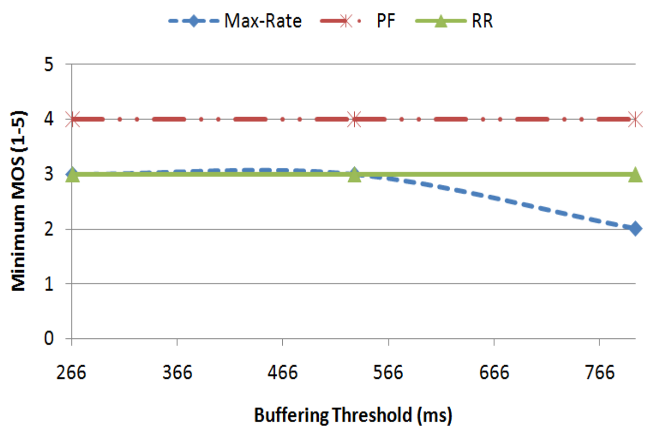

Figure 5. Minimum MOS vs buffering threshold

Figure 6 shows the FDR performance under the three well-known algorithms. With increasing buffering thresholds, the PF algorithm is having the best FDR performance followed by the RR and Max-Rate algorithms. Moreover, Table 2 shows that, the FDR performances under all algorithms slightly degrade with increasing buffering thresholds.

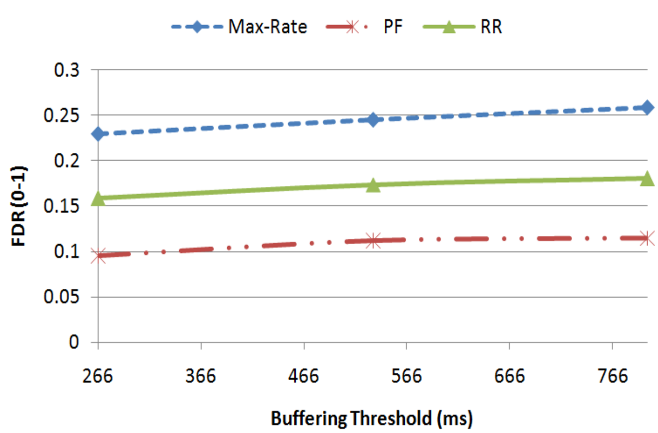

Figure 6. FDR vs buffering threshold

Table 2. Percentage of FDR degradation

	% of FDR Degradation (266.667 – 533.33)	*% of FDR Degradation (533.33 – 800)*	*% of FDR Degradation (Mean)*
Max-Rate	6.910835	5.4323	6.171568
PF	16.57972	2.622393	9.601055
RR	9.757021	3.657064	6.707043

The buffering of video frames takes place at a user's application layer. Therefore, varying the buffering thresholds does not have any impact towards the cell throughput performance (as shown in Figure 7) which is a metric being determined at the physical layer of the base station.

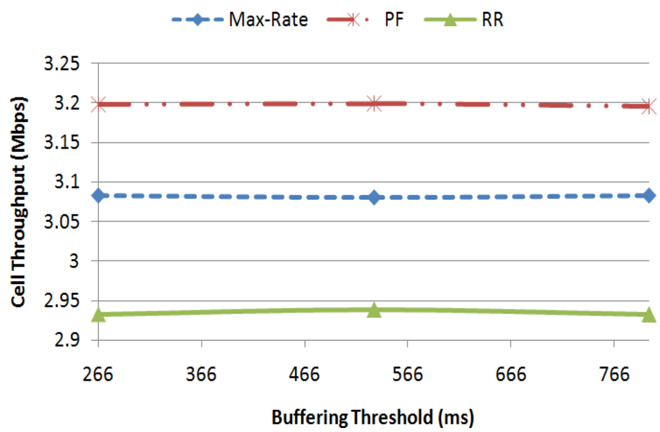

Figure 7. Cell throughput vs buffering threshold

7. CONCLUSIONS

This paper evaluates video streaming performance on the basis of minimum MOS, FDR and cell throughput under Max-Rate, PF and RR algorithms over an OFDMA based wireless IP network. These well-known algorithms have a reduced complexity compared to the content-aware packet scheduling algorithms. A more realistic video streaming scenario and performance metrics are used in the performance evaluations. It is demonstrated in the simulation results that the PF algorithm is able to provide the QoS requirements at acceptable levels for a higher number of video users compared to the Max-Rate and RR algorithms. Moreover, it is also demonstrated in the simulation results that the buffering thresholds in the range of 266.667 ms to 800 ms do not significantly impact the FDR and cell throughput performances under the three well-known algorithms when the number of users is small.

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Authors

Huda Adibah Mohd Ramli is currently a PhD candidate in the Faculty of Engineering and Information Technology, University of Technology, Sydney (UTS), Australia. She received an M.Sc. in Software Engineering from University of Technology Malaysia (2006) and B.Eng. in Electrical and Computer Engineering from International Islamic University Malaysia (2003). Her current research interests focus on radio resource management for the future wireless IP networks.

Kumbesan Sandrasegaran (Sandy) holds a PhD in Electrical Engineering from McGill University (Canada) (1994), a Masters of Science Degree in Telecommunication Engineering from Essex University (UK) (1988) and a Bachelor of Science (Honours) Degree in Electrical Engineering (First Class) (UZ) (1985). He was a recipient of the Canadian Commonwealth Fellowship (1990-1994) and British Council Scholarship (1987-1988). He is a Professional Engineer (Pr.Eng) and has more than 20 years experience working either as a practitioner, researcher, consultant and educator in telecommunication networks.

During this time, he has focused on the planning, modeling, simulation, optimisation, security, and management of telecommunication networks.

Riyaj Basukala received his Masters of Engineering (Telecommunication Engineering) Degree from UTS in 2009 and Bachelors of Engineering in Electronics and Communication from Tribhuwan University, Nepal in 2005. Currently he is involved in working as a Research Assistant in Centre for Real Time Information Networks (CRIN) in UTS focussing mainly on developing simulation tools for simulating radio resource management in various radio interface technologies and also few other communication-based research projects.

Rachod Patachaianand is currently a doctoral candidate in the Faculty of Engineering and Information Technology, University of Technology Sydney (UTS), Australia. He received an M.Eng. in Telecommunications Engineering from UTS in 2007 and B.Eng. in Electrical Engineering from Chulalongkorn University, Thailand, in 2004. His current research has focussed on radio resource management of future mobile communication systems specifically focusing on link adaptation techniques with limited feedback.

Toyoba Sohana Afrin is currently a PhD candidate in the Faculty of Engineering and Information Technology, University of Technology, Sydney, Australia. She received a B.Sc Engineering in Electrical and Electronic Engineering from Bangladesh University of Engineering and Technology (2004). She gains 6 years working experiences in the field of Radio Access Network with telecom industry (2004-2010). Her current research explores in Radio Resource Management for future mobile communication systems.