Content-Aware Radio Resource Management for IMT-Advanced

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***Abstract*—Radio Resource Management (RRM) is crucial to properly handle the delivery of quality-of-service (QoS) in IMT-Advanced systems. Normally, cross-layer optimization (CLO) involving the PHY and MAC layers, is used to provide proper resource scheduling to the overall system. Significant researches on CLO techniques incorporating the APP layer are also performed, however, the studies on the performance parameters such as system throughput, packet loss ratio and delay for a certain time are lacking. Furthermore, compatibility with the legacy systems and standards was not considered as one of the major criteria for design. Consequently, a content-aware radio resource management (RRM) model employing cross-layer optimizer focusing on video conferencing/streaming application for single cell long-term evolution (LTE) system is proposed. Based on a developed look-up table, the cross-layer optimizer can dynamically adjust the transmitted data rate depending on the user-equipment (UE) or eNodeB SINR performance. It is shown that for video packet delivery in both uplink and downlink transmissions, the content-aware RRM model vastly outperforms the legacy LTE baseline model in terms of packet loss ratio for the same amount of throughput.**

***Index Terms*—Content Aware Radio Resource Management; Cross Layer Optimisation; IMT-Advanced; Radio Resource Management.**

I. INTRODUCTION

Nowadays, more people are more likely and gradually becoming familiar with using wireless network medium to transfer various forms of data such as e-mails, pictures and videos, all which have benefitted from the fast growing wireless communication technologies. As more and more users get access to the wireless broadband system especially in LTE system, the network traffic is becoming more and more congested. This situation is made even worse when users are using multiple or heterogeneous services concurrently, especially broadband video applications and dynamically moving from one cell to another cell at the same time. This is why Radio Resource Management (RRM) is crucial to properly handle the delivery of quality-of-service (QoS) in LTE systems. One of the techniques used for RRM in IMT-Advanced is cross layer optimization (CLO) which normally involves the interaction between the PHY and MAC layers before proper resource scheduling can be decided [1]. As the IMT-Advanced standard [2] only defines the PHY and MAC layers [3,4], the effect of CLO is limited as nature of the transmitted information is not taken into account.

Many researchers have developed new RRM techniques to improve the performance of the IMT-Advanced system and to have some degree of fairness among the users. Kumwilaisak et al [5] and Zhang [6] have introduced a generic end-to-end cross-layer QoS mapping architecture for video delivery over wireless environment. However, the framework does not consider the mobility of the either end-users and furthermore, it was not intended to be specifically implemented in IMT-2000 or IMT-Advanced (e.g. WiMAX or LTE-Advanced) systems.

Another critical issue in video applications is in healthcare and more importantly, in mobile healthcare. Markarian et al [7] have introduced a novel segmented distribution framework to support object-based MPEG-4 video streams over WiMAX network. By using coded representation of media objects, each individual segmented video streams (called Elementary Stream) was treated as a part of complex audiovisual scene and could be perceived and processed separately. A cross-layer mapping table was also introduced to set up matching rules between the individual segment video stream and the assigned QoS class from APP layer down to MAC layer for delivering packets through the protocol suite. However, the system only considers uplink communication and the cross layer mapping table does not take into account PHY layer information. Apart from that, most video distribution techniques aim at delivering MPEG streams with a defined recommendation for protocol stack exploited within the communication procedures which means a WiMAX base station (BS) would misjudge the bandwidth requirement and could possibly allocate an excess bandwidth to the mobile terminal for the uplink delivery.

Previously, Mohd Sultan et al [8] have proposed a cross-layer scheduling for WiMAX disaster management situation. In normal operation, realtime applications are tied up to UGS, rtPS and ertPS QoS classes whilst non-realtime applications are hooked up to nrtPS and BE. By using cross-layer approach, we have realigned or rescheduled the non-realtime applications to rtPS QoS and also the realtime applications to BE QoS with the purpose to investigate the possibilities of BE service class producing better performance than the rtPS class. However, only certain combinations of users and QoS, BE QoS class demonstrates a higher throughput than that of rtPS class.

Basically, it seems that cross-layer optimization has become a necessity for wireless broadband systems, where performance of the overall system is vital, can be adjusted accordingly, while achieving a reasonable amount of fairness among the users.

This paper is organized as follows. In section II, the related research on cross layer design for LTE system is discussed. In section III, we develop the simulation methodology for LTE single cell baseline model and the proposed content-aware RRM model. The method to compare the performance of both models is also described here. The simulation results of comparing the proposed model with the baseline model are presented in section IV. Finally, our work of this paper is summarized in the last section.

II. CROSS LAYER DESIGN FOR LTE

Cross layer design for achieving the desired performance in wireless networks is not a new research area. It all started when wireless communication becomes more and more attractive for implementation especially in remote areas where wireline communication has become costly for deployment. Although, it may seems that the concept itself is violating the philosophy of layering concept in networking, the complex issues that are related to wireless environment such as the time-varying channels and propagation loss, somehow calls for the need for cross layer design to be taken into consideration.

Most of the cross layer design for the purpose of RRM involves the interaction between the MAC layer and the PHY layer [9,10] where in the MAC layer, proper scheduling techniques are taken place based on specific QoS requirements for each user or data bearers whilst depending on the channel state information (CSI) feedback from the PHY layer. One interesting technique is proposed by Wu et al [11] where the cross layer optimization technique does not require channel quality indicator (CQI) information to be fed back from the user side. The realtime video packet transmission is done by adapting the sent bit rate automatically according to the estimated packet loss due to expiration of packet delay deadline based on queueing analysis by taking into account both packet queueing delay and transmission delay.

In the recent years, researchers and network engineers feel the need to further increase the performance of their system due to the ever growing demand for data services especially video-related applications by the public which leads to higher volume of traffic at the eNodeB. Consequently, some initiatives have been taken to include the APP layer as part of the CLO techniques for radio resource decision making in LTE networks [12-15]. By having this type of cross-layer design architecture, the LTE/LTE-Advanced can achieve multitude objectives of improving spectrum efficiency, multi-layer diversity gain, adapting to wireless channel and satisfy users with different traffic classes [16].

Most of the APP and MAC/PHY cross layer architectures are targeted for data hungry services such as video streaming applications where high quality video frames will be adjusted which are then scheduled appropriately to particular user(s) whilst taking into account the CSI for each individual user [12-15]. In these methods, the video frames or the video encoding parameters are dynamically adjusted to suit the channel conditions for all users. However, the study on the performance parameters such as system throughput, packet loss ratio and delay for a certain time are not clearly stated in those papers. Furthermore, compatibility with the legacy systems and standards was not considered as one of the major criteria design. In this paper, we are proposing a new technique which employs the CLO concept, namely “CONTENT AWARE RADIO RESOURCE MANAGEMENT”. This CLO concept is to be expanded from the PHY layer up to the APP layer and will utilize specific properties of data and overhead transmitted over the network to ensure backward compatibility with the legacy standards and systems.

III. SIMULATION METHODOLOGY

Firstly, we establish a baseline LTE communication simulation model in which only basic RRM is applied. This baseline simulation model is important because it is considered a normal performing LTE platform which conforms to the 3GPP Release 8 standard and, hence, it will be used to compare with our proposed content-aware RRM model. The LTE topology is designed to have a Remote Host connected to a SGW/PGW Gateway which is then linked together with an eNodeB before finally having wireless interface with four UEs as shown in Figure 1.

All the four UEs are placed in a square position at the edge of a cell which is the farthest distance from one and another whilst the eNodeB is located at the centre of the cell. The communication link between the eNodeB and the UEs were implemented for both uplink and downlink transmissions in which all simulations were done in NS-3 software. The main simulation parameters were based on 3GPP specifications and each of the UEs is configured to cater for different types of application services; namely, web browsing (HTTP protocol), file transfer (FTP protocol), VoIP and video streaming.

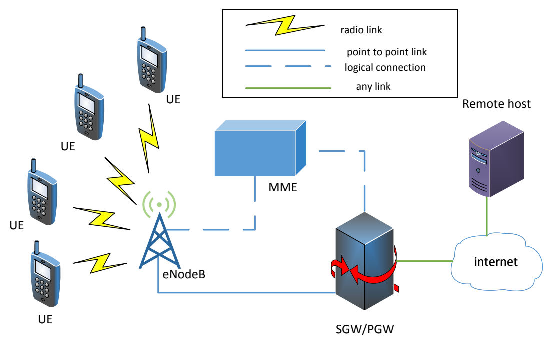

Figure 1: Single eNodeB LTE – EPC simulation topology

Table 1 summarises the implementation of the essential simulation settings and parameters used for 4 UEs in a single cell with one eNodeB whereas Table 2 shows the UEs applications test parameters. The simulation of VoIP traffic in NS-3 is based on G.711 codec and is characterized by two periods; ON and OFF. ON is for the time when the user spends on talking whereby constant packets are transmitted at regular intervals and, hence, constant bit rate traffic is generated. The OFF time is the time where the user stops from talking and packets are not transmitted [17]. ON and OFF times are given as 0.352 and 0.650 seconds respectively [17,18]. The simulation of video traffic is assumed to be coded based on H.264 or MPEG-4 Part 10 Advanced Video Coding (AVC) codec and its behavior is according to realtime services such as video conferencing or video streaming. Since the simulator does not provide appropriate video service implementations, the corresponding traffic has been modeled as a Constant Bit Rate (CBR) traffic, with the video source generates video packets at a rate of 4 Mbit/s, with packet size set to 1024

Bytes [19]. Both, VoIP and video traffics are implemented using UDP transport protocol which is the most used transport protocol especially for realtime applications.

For the best effort traffic that are represented by the web browsing and the file transfer applications, a TCPSocketFactory abstract class is used. This is because there is no NS-3 module available that provides HTTP or FTP application layer protocol. Although four application services are considered in the simulation, only the video streaming services is of particular interest to our research which represents a single-user performance for the whole system.

Table 1
Simulation Parameters for a Single eNodeB in Uplink Transmission

Parameter	Value
Bandwidth	5 MHz
Operating frequency	1.93 MHz
Duplex mode	FDD
Transmission scheme	SISO
Channel model	Friis Propagation Loss Model
Scheduler	Proportional Fair (PF)
UE velocity	20 m/s = 72 km/h
eNodeB – UE distance	7071 – 21213.2 m
Number of UEs	4
eNodeB transmit power	43 dBm
UE transmit power	21 dBm
Simulation time	60 seconds

Table 2
Test Parameters for UEs

UE	Application Type	Data Rate, R (kbps)	Packet Size (bytes)	Number of Packets
1	web browsing	32	1024	100000
2	file transfer (FTP)	32	1024	100000
3	VoIP	64	1024	100000
4	video streaming	4000	1024	1000000

A. *LTE Single Cell Baseline Simulation Model*

For the uplink transmission, only the VoIP and video streaming services were installed in the UEs whereas the web browsing and file transfer were installed in the same Remote Host but with different ports. The simulation starts off at 0 seconds with web browsing and file transfer application services are initialized at the Remote Host whereas VoIP and video streaming applications are initialized in their own respective UEs located 7071.1 m away from the eNodeB. During transmission, the existing LTE framework sets up the lower layer protocols, which includes the radio stack and the GPRS Tunneling Protocol (GTP) core network bearer, accordingly [20]. Only after 2 seconds, the UEs are allowed to move randomly following a waypoint mobility model with a constant velocity of 20 m/s or 72 km/h closing towards the eNodeB. After 60 seconds, the simulation stops and the output performance parameters were measured. The output performance parameters which are of interest such as the throughput, packet loss ratio, end-to-end delay and also UE SINR values for the video streaming application are then recorded. The simulations are then repeated for various distances between the UEs and the eNodeB as indicated in Table 1.

As for the downlink transmission, the Remote Host is now configured to be the transmitter whilst the UEs have now become the receivers. Apart from that, the same set of parameters as specified in Tables 1 and 2 are reused here, however, there are two parameters that need to be changed as well, namely, the operating frequency from 1.93GHz to 2.12GHz and the eNodeB to UE distance parameter which has to be varied from 66.5 km to 190.9 km. The reason for the changes are due to the FDD mode implementation and also due to much higher transmit power by the eNodeB as compared to that of the UEs which enables the eNodeB to transmit at a wider coverage area, respectively.

In this particular situation, all the four application services are initialized at the same Remote Host with four different ports and the UEs which act as the receivers are positioned 66.5 km away from the eNodeB, initially. The same steps taken while implementing the uplink transmission simulation is done here again for the downlink transmission until the output performance parameters specifically for the video streaming are measured. Only this time, the throughput, packet loss ratio, end-to-end delay calculations and eNodeB SINR are all measured at UE 4. The same simulation setup is then repeated with the eNodeB – UE distance incremented by 707.1 m for each simulation time until the UEs reach the distance of 190.9 km from the eNodeB. The fact that the basic channel model is used in the simulation (e.g. Friis Propagation Loss Model) which depends primarily on the eNodeB transmit power while other parameters are kept constant, enables the eNodeB to propagate its downlink signal much further away as compared to the uplink transmissions by the UEs. However, shorter coverage distances could be expected for the downlink transmission if other detailed channel models were used instead, such as the empirical COST231 Propagation Model which considers both the transmit and receive antennas' heights or the Two-Ray Ground Reflection Model which covers not only both transmit and receive antennas heights, but also higher path-loss exponent.

B. *Cross Layer Optimisation of RRM Model*

Each simulation results that was recorded specifically for UE 4 which contains the video streaming service comprises of four types of output performance data as we mentioned earlier. However, in this paper, only the throughput versus SINR graphs are plotted for both uplink and downlink transmissions as indicated in Figures 2 and 3. Those results are expected due to the link adaptation performed by the eNodeB which result in various adaptive modulation and coding (AMC) schemes in both transmissions producing staircase-like pattern. This means when the SINR is low, there will be no chance for the throughput to match its data rate and, hence, the packet loss ratio and the delay will be high. As we know that throughput is a measure of the rate of data that has been successfully delivered to a receiver for a specific simulation time, then it is pretty obvious that in order to maximize the throughput, we have to make the transmission data rate equal to the throughput itself depending also on the UE SINR. In return, we can minimize the loss of packets and, so, reducing the delays of transmission which will ultimately, prevent any wastage of bandwidth.

It is notably understandable that in every mobile data transmission, we want to maximize throughput and, at the same time, minimize the amount of packets lost and the end-to-end delay. The only issue is where and when those objectives can be met. In response to that, and by referring to both graphs for the baseline model as shown in Figures 2 and 3, we can draw the correlation between the Throughput and SINR and, thus, recommending the suitable video packet generation rate at the sources for both uplink and downlink transmissions. Ultimately, we want to introduce a new concept in RRM system which can dynamically adjust the transmitted data rate depending on the UE or eNodeB SINR performance in order to minimize the packet loss. This concept which involves cross-layer optimization approach is called content-aware RRM model or sometimes it is also called joint source and channel coding. In order to realise this, we propose a new cross-layer look-up table that sets up the matching rules between the specific UE SINR or eNodeB SINR and the assigned data rate for delivering video packets through the protocol suite for both uplink and downlink transmissions as shown in Tables 3 and 4, respectively.

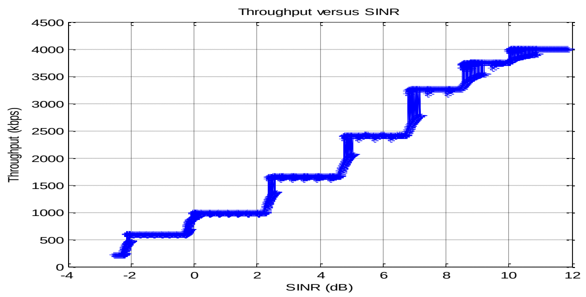

Figure 2: Throughput against SINR plot for uplink transmission (R = 4 Mbps)

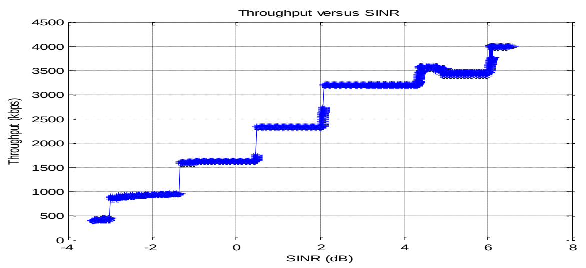

Figure 3: Throughput against SINR plot for downlink transmission (R = 4 Mbps)

Table 3 proposes that in order for the data rate of the UE to be adjusted accordingly, accurate estimates of the current channel quality of the link between the UE and its associated serving eNodeB should be done first. In the normal uplink transmission, the eNodeB has the knowledge of the SINRs on the various subcarriers by measuring and evaluating both the Sounding Reference Signal (SRS) and PUSCH signals transmitted by the respective UEs. Having estimates of the SINRs of all subcarriers allocated to a certain UE based on its unique Radio Network Temporary Identifier (RNTI) or International Mobile Subscriber Identity (IMSI), the eNodeB can determine the spectrally most efficient Modulation and Coding Scheme (MCS) for which a given target Block Error Rate (BLER) is not exceeded. For that purpose, it may choose several different modulation schemes as well as a variety of different channel coding rates [21]. Afterwards, the selected MCS is signaled as part of the scheduling grant to the corresponding UE using the PDCCH. However, in our design, the SINR values, apart from the scheduling grant, are also fed back to the targeted UE only, using the same PDCCH, every 40 ms which is also equivalent to the SRS signal periodicity. Consequently, a newly designed cross layer optimization module will use this received SINR values from the UE's PHY layer together with the information on the current data rate of its video streaming packets from the APP layer to dynamically assign the suitable data rate for its video streaming packets in the APP layer, based on the proposed look-up table in Table 3. The cross layer optimizer concept designed at the UE is shown in Figure 4. It is worthy to note that in order to make the CLO backward compatible with any previous systems (e.g. 3G and 2G), we do not involve the changing of protocols of any sort to any layers, especially, the PHY and MAC layers to ensure that the CLO is easily attached to or detached from the UE.

Table 3
Proposed Look-up Table for Uplink Content-aware RRM Model

Proposed Data Rate, R (Mbit/s)	SINR (dB)
0.250	< – 2.38
0.6	–2.38 – –0.25
1	–0.25 – 2.25
1.650	2.25 – 4.75
2.450	4.75 – 6.75
3.250	6.75 – 8.5
3.750	8.5 – 10
4	> 10

For the downlink, the channel estimation is done in the targeted UE by measuring the SINR based on the reference signal (RS) transmitted periodically by the eNodeB. This SINR information is then fed back to the eNodeB as an input for the cross layer optimizer before exhaustive search is made to decide on the most suitable data rate for video transmission from the proposed look-up table in Table 4. Once the matching data rate is found, then the CLO will instruct the Remote Host to change its current data rate to the new one.

Table 4
Proposed Look-up Table for Downlink Content-aware RRM Model

Proposed Data Rate, R (Mbit/s)	SINR (dB)
0.485	< –3.15
0.85	–3.15 – –2.5
0.9	–2.5 – –1.3
1.6	–1.3 – 0.5
2.4	0.5 – 2
3.2	2 – 4.25
3.4	4.25 – 6
4	> 6

In the normal LTE downlink transmission, the eNodeB will, based on the channel quality, allocate the available resource blocks (RBs) to different users and choose proper MCS for multiple users. The channel quality is estimated by the UEs at the receiver side in terms of SINR, however, instead of transmitting back the SINR values to the eNodeB using the PUCCH, the receiver feeds back the channel quality information to the eNodeB in terms of CQI values. Each CQI value corresponds to one MCS, and the better channel quality is, the better MCS the channel can support and, thus, the CQI value can reflect the channel quality [22].

For our design, apart from the CQI values, the SINR values are also fed back to the eNodeB using the same PUCCH which will be further used as an input to our newly designed cross layer optimizer at the transmitter side. The eNodeB can easily identify the SINR values for a particular UE by its unique RNTI or IMSI. The reasons for both the CQI and SINR values feedback are because CQI values are used for the link adaptation purpose whilst SINR values provide a more accurate estimation of the channel condition before the cross layer optimizer can make the important decision in adjusting the video data rates accordingly. For adapting to fast channel quality variations, periodic CQI and SINR reporting schemes are used with a reporting interval of 1 ms or 1 transmit-time-interval (TTI). The PHY layer in the eNodeB provides the SINR information to the APP layer in the Remote Host. Since the APP layer of the Remote Host is not aware of RRM in the frequency spectrum, CQI values which consists of both wideband and inband CQIs are not useful for adaptations in the APP layer. The reason is adaptations for each and every CQIs is not practical and impossible to implement in realtime systems. So, only the SINR values are used for the data rate adaptation in the Remote Host.

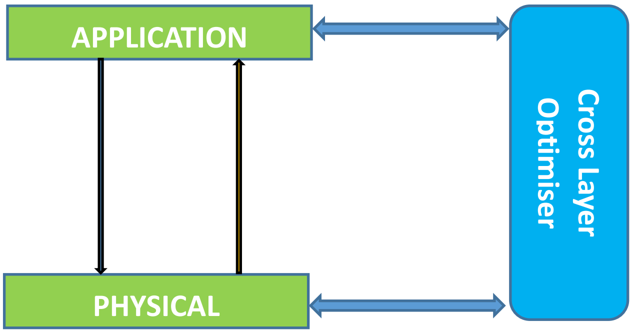

Figure 4: Cross-layer optimizer for content-aware RRM model at UE

C. Comparison Parameters

In order to evaluate the performance of our proposed content-aware RRM model, a new set of comparison parameters has been established to compare the performance of the proposed model and the baseline model. This new parameters are defined as follows:

$$\varphi_T = \int_0^t Throughput\ dt \quad (1)$$

$$\varphi_P = \int_0^t PLR\ dt \quad (2)$$

$$\varphi_D = \int_0^t Delay\ dt \quad (3)$$

where φ_T is total data received or area under the curve for throughput, φ_P is area under the curve for packet loss ratio and φ_D is area under the curve for average end-to-end delay. All the three parameters mentioned above represent areas under the curves for all the three output performance parameters; namely throughput, packet loss ratio and average delay that will be calculated with respect to total simulation time. Improvements can only take place if φ_T for one system is higher whilst φ_P and φ_D are lower than those of its counterpart

IV. RESULTS AND DISCUSSIONS

The performance of the proposed content-aware RRM model is then compared with that of the baseline model. Using the same specifications defined in Table 1 and Table 2, both models are simulated for 10 minutes with all UEs positioned at the edge of the cell which is 21213.2 m away from the cell centre before moving towards the eNodeB at 72 km/h (or 20 m/s) which represents normal vehicular speed. Only UE 4 which transmits or receives video streaming services at 4 Mbps for both models is analysed here.

Figure 5 and Figure 6 show the packet loss ratio and average end-to-end delay of both models over the course of 10 minutes when one UE 4 transmits video packets to the Remote Host via the eNodeB while moving towards the eNodeB from the edge of the cell at 20 m/s, respectively. Similarly, Figure 7 and Figure 8 display the same plot but for the downlink transmission. In Figure 5, the content-aware RRM model outperforms the baseline model in terms of packet loss ratio by a staggering 98.92% improvement. For the same amount of data transmitted in both models, the total number of packets lost during the transmission in the channel is so huge in the baseline model and thus resulting in the wastage of bandwidth. As a matter of fact, the content-aware RRM model also experiences much less average delay with 23.06% improvement as shown in Figure 6 and, this means QoS for the video streaming application can be preserved.

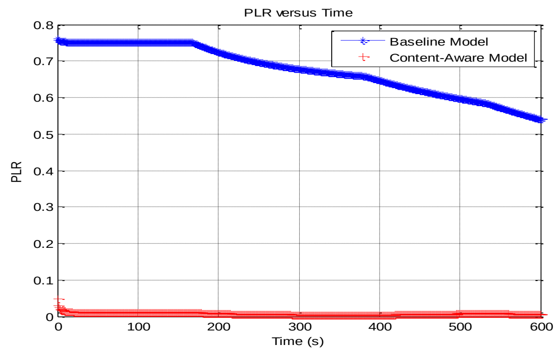

Figure 5: Packet loss ratio against time for uplink video delivery

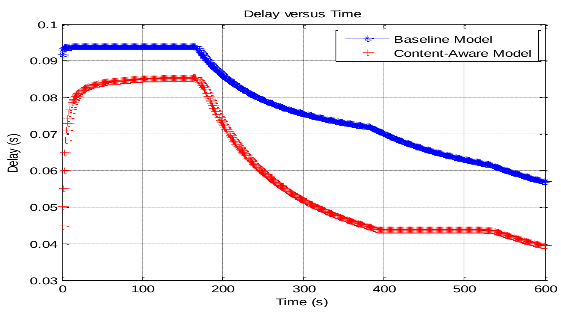

Figure 6: Average delay against time for uplink video delivery

Again, the much better performance of the content-aware RRM model in the uplink video delivery is further supported by the same content-aware RRM model in the downlink video transmission as indicated in Figures 7 and 8. Over the 10-minute simulation, the content-aware RRM model vastly outperforms its counterpart, the baseline model with a 92.1% improvement in packet loss ratio and a significant 19.52% improvement in average delay as shown in Figure 7 and Figure 8, respectively. This means by employing content-aware RRM model, we can avoid a great deal of bandwidth wastage and also preserving the QoS of the video streaming application as opposed to the baseline model where the QoS could be effectively compromised.

The Throughput against Time graphs for both models are not displayed here because the results are very much the same either for uplink or downlink transmissions. The reason is that for Baseline Model, the wireless transmission channel basically downgrades the transmitted 4 Mbps data rate whereas for the Content-aware RRM model readjusts the transmitted data rate accordingly which results in the same amount of throughput for both models.

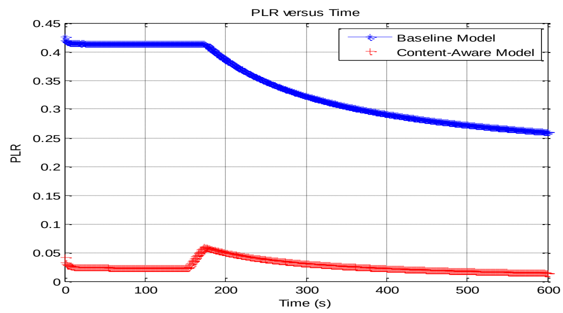

Figure 7: Packet loss ratio against time for downlink video delivery

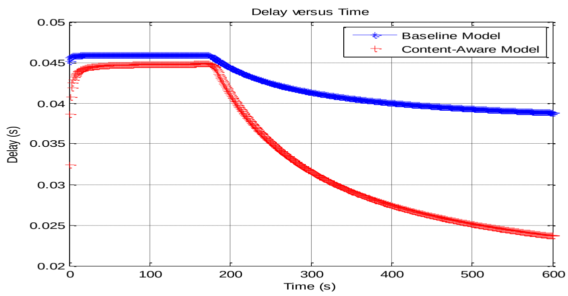

Figure 8: Average delay against time for downlink video delivery

In short, it can be summarized that the Content-Aware RRM model produces a much better performance than the Baseline model in either the uplink or downlink video transmission. In fact, for the same amount of throughput, the content-aware RRM model in all simulations proves to be extremely superior in reducing packet loss ratio and average end-to-end delay performance compared to the Baseline model.

V. CONCLUSION

In conclusion, a content-aware RRM model by employing cross layer optimization with the proposed look-up table for single cell LTE system is proposed for both uplink and downlink transmissions. The results have shown that for the same amount of throughput, the proposed model has made huge improvements in terms of packet loss ratio when compared with that of the baseline model. In effect, the proposed model can be used to further improve video delivery performance in the current LTE system without the need to modify the current standard and protocols. In the near future, we also extend this study to investigate the impact of UE mobility on the new CLO concept.

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