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Araştırma Makalesi - Research Article

LTE-A üzerinden Katmanlararası Optimize Edilmiş Çoklu Video Gönderimi

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ÖZ

3. nesil ortaklık projesi, ileri seviyede geliştirilmiş uzun vadeli evrim (LTE-A) ağları üzerinden geliştirilmiş çoklu multimedia servisleri için (eMBMS) uygulama katmanlı ileri hata düzeltme kodlarının (AL-FEC) kullanılmasını belirtmiştir. 3GPP, eMBMS'in veri iletimi için sabit bir modülasyon ve kodlama şeklinin (MCS) seçildiği tek frekanslı ağ (MBSFN) üzerinden kullanımını standardize etti. Spektral verimliliği artırmak için 3GPP tarafından tek hücreli noktadan çok noktaya (SC-PTM) iletim yöntemi tanımlandı. Bu nedenle, bu makalede LTE-A SC-PTM üzerinden AL-FEC'e dayanan çoklu video gönderimi önerilmektedir. Önerilen sistemin performansının analiz edilmesi için katmanlararası LTE-A simülatörü geliştirilmiştir. Simülasyon sonuçları önerilen sistemin LTE-A tarafından önerilen sisteme göre daha fazla kapsama alanı sağladığını (Sinyal Parazit Gürültü Oranında 4 dB gelişme sağlamaktadır) ve daha az radyo ve ağ kaynağı kullandığını (%48,7 daha fazla spektral verimlilik sağladığını) göstermektedir. MBSFN'den farklı olarak SC-PTM modu, kullanıcılardan geri bildirim alarak her bir gelişen düğümB'de (eNB) MCS gibi sistem parametrelerinin dinamik olarak seçilmesine izin verir.

Anahtar Kelimeler- ***Uygulama Katmanlı İleri Hata Düzeltme, Çoklu-ortam yayma ve gönderme servisleri, Video Gönderme, LTE-A***

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Cross-Layer Optimised Video Multicasting over LTE-A

ABSTRACT

3rd Generation Partnership Project (3GPP) specified the use of Application Layer Forward Error Correction (AL-FEC) to improve the evolved Multimedia Broadcast Multicast Services (eMBMS) over Long Term Evolution-Advanced (LTE-A) networks. 3GPP standardised the use of eMBMS over a Single Frequency Network (MBSFN) in which a fixed Modulation and Coding Scheme (MCS) is selected for the data transmission. In order to improve the spectral efficiency, a Single-Cell Point-to-Multipoint (SC-PTM) transmission scheme was introduced by 3GPP. Unlike MBSFN, SC-PTM mode allows dynamic selection of the system parameters such as MCS in each evolved NodeB (eNB) by getting feedback from users. Therefore, in this paper an AL-FEC based video multicasting for LTE-A SC-PTM is proposed. A cross-layer LTE-A simulator is developed to analyse the performance of the proposed system. Simulation results show that the proposed system provides higher service coverage, i.e., 4 dB improvement in Signal to Interference Noise Ratio (SINR) and uses less radio and networks resources (i.e., provide up to 48,7% higher spectral efficiency) compared to the legacy LTE-A system.

Keywords-Application Layer Forward Error Correction, Multimedia Broadcast and Multicast Services, Video Streaming, LTE-A

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I. INTRODUCTION

The widespread use of mobile devices such as smart phones and tablets has increased the demand for multimedia delivery over 3rd Generation Partnership Project (3GPP) Long Term Evolution-Advanced (LTE-A) [1]. There are some applications many users want to access the same application at the same time (for example news). When unicast transmissions is used, the same content must be transmitted to each user separately. However, there is limited bandwidth in LTE-A thus when the number of users increases the networks run out of bandwidth. Bandwidth shortage becomes severe when each user also wants the transmitter to retransmit the lost packets. In these scenarios, multicast transmission is one of the solutions to efficiently deliver such applications (e.g., video) over wireless channels. Therefore, 3GPP standardised the use of Multimedia Broadcast and Multicast Services (MBMS) streaming and download delivery services in Release 6 [2]. However, multicast transmission does not provide reliable delivery of the data since multicast packets are transmitted over wireless channels as a broadcast service in which there is no Medium Access Control (MAC) layer packet retransmission mechanism. Therefore, in multicast delivery, high packet losses occur. Video applications cannot tolerate higher packet losses which lead serious degradation in the video quality. To this end, in MBMS [3] Application Layer Forward Error Correction (AL-FEC) based on Raptor codes is suggested for multicast/broadcast streaming and download delivery services. AL-FEC scheme allows the receiver to recover the corrupted source data (lost packets) by using the redundant data transmitted along with the original packets.

In 3GPP Release 8 [4], MBMS was enhanced to become evolved MBMS (eMBMS) and its underlying transmission scheme is MBMS over the Single Frequency Network (MBSFN). In MBSFN, a time-synchronised signal is transmitted simultaneously from a number of evolved NodeBs (eNBs) or Base Station (BS) using the same Resource Blocks (RBs) to a group of users thus the Signal to Interference Noise Ratio (SINR) at the receivers are greatly enhanced. The MBSFN model provides significant improvements in Spectral Efficiency (SE), however this depends on the Modulation and Coding Scheme (MCS) selected for the data transmission since in this mode a fixed MCS is used to meet the edge-of-cell user requirements, i.e., there is no adaptive feedback from the users. To overcome this issue, in 3GPP Realise 13 [5] a Single-Cell Point-to-Multipoint (SC-PTM) transmission scheme was introduced. Unlike MBSFN, SC-PTM mode allows dynamic selection of the system parameters (e.g., MCS) in each eNB by getting feedback from users. Thus, the SE is further increased. The benefits of the SC-PTM mode over MBSFN in terms of SE is investigated in [5, 6] for mission critical communications and in [7] for public safety communications. However, these work does not consider AL-FEC mechanism. In [8], a study on performance and cost analysis of different MBSFN deployment scenarios, user populations and Packet Loss Rates (PERs) is presented for the file delivery services considering AL-FEC based on the standardised Raptor 10 codes. In [9], AL-FEC based streaming services over eMBMS are evaluated considering the PER and the required FEC overhead in a single cell scenario. As AL-FEC scheme requires additional bandwidth to provide reliable data delivery, the amount of redundant data (or overhead) needs to be determined depending on the channel conditions and the MCS mode used for the data transmission in order to efficiently use the valuable radio and network resources. In [10], a cross-layer approach which aims to optimise the number of streamed video layers, the selected MCS and the amount of the AL-FEC overhead used for each layer is studied. All these works considered the cross-layer optimisation based on the 3GPP standardised Raptor 10 codes [11], however there exists a more practical version of the Raptor codes which is called Raptor Q (RQ) [12]. The potential gain that can be achieved using RQ based AL-FEC mechanisms in SC-PTM has not been addressed yet. Thus, in this paper, a cross-layer design and optimisation based on the RQ codes is considered for delivering high data rate live video over LTE-A SC-PTM networks in outdoor environments for vehicular scenarios. The main contributions of this paper are to implement the latest RQ codes in an LTE-A deployment scenario, provide comprehensive information and guidance on the cross-layer system design parameters, and propose a cross-layer optimisation algorithm for the efficient transmission of eMBMS streaming services over LTE-A SC-PTM networks.

The paper is organized as follows. Section II gives some background information. Section III explains the details of cross-layer simulator. Section IV presents proposed cross-layer optimisation algorithm. Simulation results and analysis are given in Section V with conclusions presented in Section VI.

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II. BACKGROUND

A. 3GPP LTE-A Physical Layer Feature

The LTE-A supports two types of frames: *Type 1* is used for both full and half duplex Frequency Division Duplex (FDD), while *Type 2* is used for Time Division Duplex operation [1]. In FDD mode, the duration of a frame is 10 ms and is divided into 10 subframes (Transmission Time Interval (TTI)) of 1 ms each, and each subframe is divided into two slots of 0.5 ms each as defined in [1]. Each slot contains either six or seven Orthogonal Frequency Division Multiplexing (OFDM) symbols, depending upon the length of Cyclic Prefix (CP). The maximum number of subframes that can be used for transmitting MBMS data is six out of ten subframes per radio frame. In LTE-A, radio resources are allocated in units of Physical (PHY) RBs. PHY RB states a unit of time-frequency resources: 0.5 ms time duration (0.5TTI) and 14 OFDM symbols. PHY RBs are allocated as RB Pairs (RBPs) that is a fixed frequency-time resource allocation unit in LTE-A which consists of 14 subcarriers in frequency domain and two-time slots (1 ms) in the time domain. The allocation of the radio block is performed by scheduler at the eNB so that each incoming MAC Protocol Data Unit (PDU) is allocated a single PHY Transport Block (TB) for transmission over the eNB/UE interface. The PHY TB represents a PHY layer packet whose size T_{TB} depends on the MCS mode selected by the MAC layer scheduler. A TB may consist of N_{RBP} RBPs.

B. Application Layer Forward Error Correction Codes

Raptor codes are a form of application layer forward error correction codes implemented at the application layer across packets to protect the multimedia data against packet losses. To this end, the encoder gathers the incoming UDP data packets, which are called source symbols, in order to build source blocks. In Raptor codes, each source block consists of k source symbols with T bytes. The encode then generates a number of encoding symbols, which are called repair symbols r, with T bytes for each source block [12]. Since the Raptor code is a systematic code, the first k symbols of n encoded symbols (packets) are the original symbols (UDP packets). The code rate of the Raptor codes is defined as $CR = k/(k + r) = k/n$.

Raptor codes are classed as fountain codes. This means they are able to generate an unlimited number of encoded redundant symbols r from a source block. A fountain code has a property that the decoder can reconstruct the original file as long as enough symbols (packets) arrive at the decoder. Raptor decoder requires slightly more symbols (received packets) than the original k symbols to successfully decode the source block, i.e., Raptor codes have a small reception overhead, ε which is given as $\varepsilon \leq (k`-k)/k$, where $k`$ is the number of received symbols at the decoder. The probability of the decoding success of Raptor codes increases with every additional received symbol. Therefore, the reception overhead of Raptor codes contingents upon k and the required probability that the source block be able to fully recover from the received symbol set [11, 12]. Raptor codes have taken attention due to low-complexity and flexibility. For instance, a Raptor code processing requirement increases linearly depending on the source block size k. Thus, Raptor codes can be implemented in software rather than hardware. Moreover, the number of source block size k and the number of encoded symbols n can be as large as required. In this work, the latest RQ AL-FEC code is considered since it has better (improved) coding performance and implementation flexibility compared to the legacy Raptor 10 [3]. The RQ codes allow implementing a quite flexible range of source block sizes. In practical implementations of RQ codes employing small block sizes is beneficial for devices which have limited power and processing capabilities such as mobile phones or tablets.

III. CROSS-LAYER SIMULATOR

A cross-layer LTE-A system simulator was developed in MATLAB in order to evaluate the end-to-end system performance. The simulator consists of four modular subsystems: 1) the H.264 video codec, 2) RQ, 3) LTE-A MAC-PHY layer, and 4) the channel simulators. Each simulator is developed separately to overcome the computational complexity. The H.264 codec simulator is able to model the transmission of any given video sequence over the MAC and PHY layers of LTE-A system. The video encoder converts video frames into Network Abstraction Layer Units (NALUs). In the simulations, it is assumed that one NALU is encapsulated into one RTP/UDP packet. The RQ encoder collects the incoming RTP/UDP packets to form source blocks each

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consists of k source symbols with T bytes and then generates n encoded symbols from each block. The LTE-A MAC-PHY layer simulator, which is based on the standard, models the packet loss trace for a sequence of NALUs. This simulation is executed for all given MCS modes and RQ code rates. At the receiver side, the RQ decoder waits to receive all the UDP packets belong to a given source block. At the decoder, when the total number of received packets (source and repair symbols) for a given source block is $k` \geq (\varepsilon + 1)k$, then the RQ decoder is able to decode the source block, i.e., all source packets of the source block are reconstructed and passed to application layer. But, if the decoding process fails (the decoder could not successfully decode the source block), then only received original source UDP packets are delivered to the video codec.

A. Channel Model

The LTE-A MBMS performance is evaluated for a vehicular user in an urban environment. The ITU channel model [13] for macrocell scenario considered in this work. The cellular network topology consists of seven LTE-A BSs. Interference from adjacent cells are considered in the evaluation of the system performance. The UE is served by the main BS, which is located at the centre cell, while the surrounding six BSs are the sources of interference. All (main and interfering) BSs operate in the 5.9 GHz band and implement a transmit power to the antenna port of 43 dBm. The BSs and MS heights are 25 m and 1.5 m respectively. The parameters used in the channel model is summarised in Table 1.

B. Link-level Abstraction

An Effective SINR Mapping (ESM) PHY abstraction model which is called as the Received Bit Mutual Information (RBIR) [14], is implemented to generate the PER *Pe*. In this ESM model, a block of OFDM sub-carrier SINRs is translated into a single Effective SINR (ESINR) value. This ESINR value is used to calculate the PER values for any given MCS by using a non-faded PER versus SINR look up table. This SINR look up table is created by performing bit level LTE-A simulations for an AWGN channel. The MCSs for an LTE-A 10 MHz channel profile are used in the RBIR simulator for a Single Input Single Output (SISO) system that can be seen in Table 2.

Table 1. Cross-layer simulation parameters.

Parameter	Value
LTE Advanced Bandwidth	10 MHz
Carrier Frequency	5.9 GHz
FFT size	1024
Number of subcarriers, N_C	600
Number of RB per OFDM, N_{RB}	50
BS transmit power	43 dBm
BS height	25 m
UE height	1.5 m
User mobility	50 km/h
Raptor symbol size, T	1500 bytes
Source block length, k	50
Video bit rate V_{BR}	6 Mbps
Delay constraint, D_{th}	100 ms
UPD PER threshold, Pe_{th}	0.001

Table 2. LTE-A PHY layer simulation parameters.

MCS Index	MCS Mode	Bit Rate (Mbps)	Code Rate CR_p	Bits/Symbol N_{BS}	Coded Bits/RBP N_{BRB}
1	QPSK	8.4	1/2	2	168
2	QPSK	12.6	3/4	2	252
3	16 QAM	16.8	1/2	4	336
4	16 QAM	25.2	3/4	4	504
5	64 QAM	25.21	1/2	6	504
6	64 QAM	37.8	3/4	6	756

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IV. PROPOSED CROSS-LAYER OPTIMISATION FRAMEWORK FOR SC-PTM OVER LTE-A

The proposed cross-layer optimisation algorithm aims at providing scalable and reliable multicast video streaming services over the LTE-A SC-PMT system by jointly selecting MCS mode *m* and RQ code rate *CR* for a given SINR γ under the Quality of Service (QoS) constraints of the applications which are defined in [15] (e.g., the delay and PER constraints for High Definition (HD) live video streaming are $D_{th} \leq 100$ ms and $Pe_{th} \leq 0.001$). In particular, the proposed framework aims to optimise the user Quality of Experience (QoE) and the total network resources (RBPs) required to deliver *n* encoded symbols of a source block to the video decoder. The total network resources (total RBPs) required is given as:

$$N_{TRBP}(\gamma, m, CR, k, T) = ceil\left((8.T.n) / N_{BRB}\right) \quad (1)$$

where N_{BRB} is the coded bits per RBP and given in Table 2. It is assumed that the same MCS mode *m* is used to transmit all encoded symbols belong a source block. MCS is changed for the next source block if at least one UE sends feedback message and informs the eNB that its PER *Pe* is higher than the defined UPD PER threshold of Pe_{th} which depends on the QoS requirements of the application. Therefore, the optimisation problem for given RQ parameters, SINR and video bit rate can be stated as follows:

$$(m, CR) \min N_{TRBP}(\gamma, m, CR, k, T) \quad (2)$$
$$\text{subject to } P_e(m, CR, k) \leq Pe_{th} \quad (3)$$
$$D(k, T, V_{BR}) \leq D_{th} \quad (4)$$

where $D = (k.T.8) / V_{BR}$ is the delay caused by the RQ encoding process since the RQ encoder waits to collect enough source symbols to perform encoding which depends on the RQ parameters *k* and *T*, and the video bit rate V_{BR}. In the proposed algorithm, for fixed source symbol size *T* and video bit rate V_{BR}, the source block size *k* is defined based on the maximum delay D_{th} constraint of the application and calculated as:

$$k \leq floor((V_{BR}.\, D_{th}) / 8.T) \quad (5)$$

Given the dependency of the objective function (2) to many system parameters, the optimisation problem is very complex to solve with analytical tools, especially when seeking to consider complex channel models. In related literature, solutions to similar problems have been obtained by using simulations and a heuristic strategy, e.g., [10, 16, 17]. Therefore, Monte Carlo simulations are used to provide the required data. The optimisation is performed for each mean channel SINR γ and source block size *k* values and identifies the optimal pairs (*m*, *CR*) from the available MCS set $M_s = \{m_i \in [1, 2,.., 6]\}$ and RQ code rates $C_{RQ} = \{0.5 \leq CR_j \leq 1\}$ that provides required QoS while using the least amount of radio and network resources by jointly adapting the FEC redundancy at the application layer and the MCS mode at the PHY layer. Algorithm 1 summarises the cross-layer optimisation algorithm.

1) **Algorithm 1:** Cross-layer optimisation of RQ code rate *CR* and MCS mode *m*

1) For a given video bit rate V_{BR}, delay constraint D_{th} and RQ source symbol size *T* calculate the source block size *k* value as defined in (9).
2) For each mean channel SINR γ and source block size *k*:
    a) $\forall$ MCS mode $m_i \in M_s$ and $\forall$ $CR_j \in C_{RQ}$ determine a set P_c of candidate pairs of MCS mode and RQ code rate (m_i, CR_j) such that the $Pe \leq Pe_{th}$.
    b) $\forall$ (m_i, CR_j) $\in P_c$ calculate the total network resources required $N_{TRBP}(\gamma, m, CR, k, T)$ where it is defined as (5).
    c) Find the MCS and *CR* pair $p_{opt}(m_i, CR_j)$ from the available candidate pair P_c that minimises the required network resources. This is the optimum pair p_{opt} selected for the specific mean SINR and RQ parameter *k*.

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V. RESULTS AND ANALYSIS

In order to analyse the performance of the proposed system, multicast transmission of a HD video consists of 2000 UDP source packets is simulated over an LTE-A network. Details of the simulation parameters are provided in Table 1 and Table 2. First the system performance is evaluated for different source block sizes (k). Figure 1 compares the UDP PER depending on the SINR for different RQ code rates (ranges $C_{RQ} = \{0.5 \leq CR_j \leq 1\}$) and source block sizes (k=50 and k=200). It is seen that using RQ codes can significantly improve the system performance: 4-6 dB SINR improvements compared to the legacy system. Using lower RQ codes provides services (UDP PER, $Pe_{th} \leq 0.001$) even at very low SINR ranges in which the legacy system does not provide any services. Further that higher source block size provides better performance, i.e., for given SINR it requires less resources since RQ code rate is higher or for the same UDP PER it requires lower SINR values. Although higher source block size provides better performance (lower UDP PER at low SINR values), it causes longer delay. Therefore, the source block size must be selected depending upon the applications QoS requirements in order to avoid the excessive delay which cannot be tolerated by video applications. In this work, k=50 is used which is calculated from equation (9) for the parameters defined in Table 1. Therefore, rest of the results are presented for k=50.

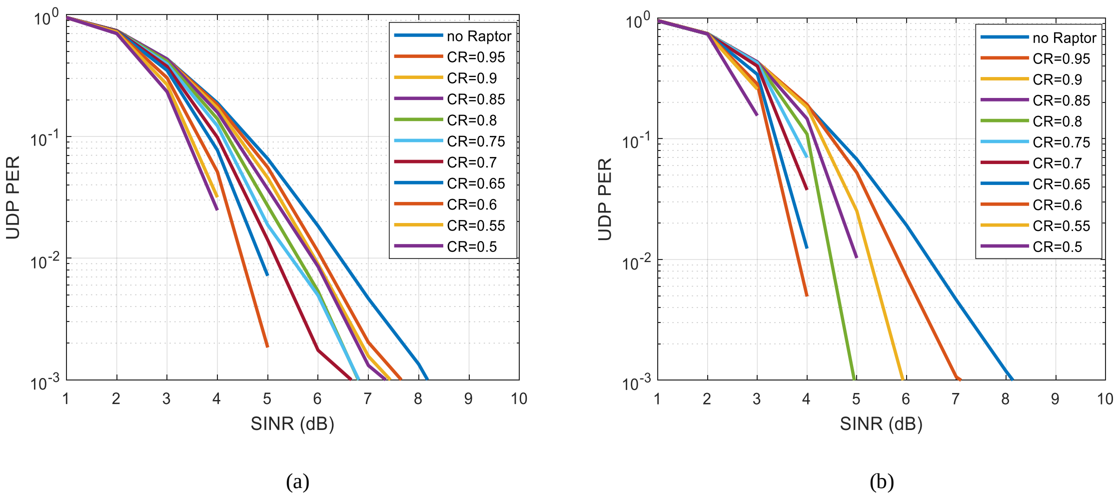

Figure 1. UDP PER versus SINR depending on the RQ code rates (CR) for MCS 1: (a) k=50 and (b) k=200.

In order to analyse how much redundancy is need for RQ codes to successfully decode a source block, the system performance over a range of mean channel SNR values is evaluated for each MCS mode. As an example, Figure 2 shows the UDP PER before and after RQ decoding depending on MCS mode for a mean channel SINR value of 8 dB and 12dB. It is seen that for MCS modes higher than 3 there is no RQ code rate in the simulated range, which is able to deliver UDP PER, $Pe_{th} \leq 0.001$. For MCS mode 2 and 3, RQ code rate of 0.5 provides a UDP PER, $Pe_{th} \leq 0.001$ and for MCS 1 all RQ codes provide a UDP PER $Pe_{th} \leq 0.001$. Similar observations can be seen for SINR of 12 dB. As seen that there are a number of MCS and CR pairs that provide successful decoding at SINR = 8 dB and 12 dB. The proposed cross-layer optimization algorithm selects the optimum combination of MCS mode and CR pair which uses less valuable radio and network resources for given QoS constraints of the application.

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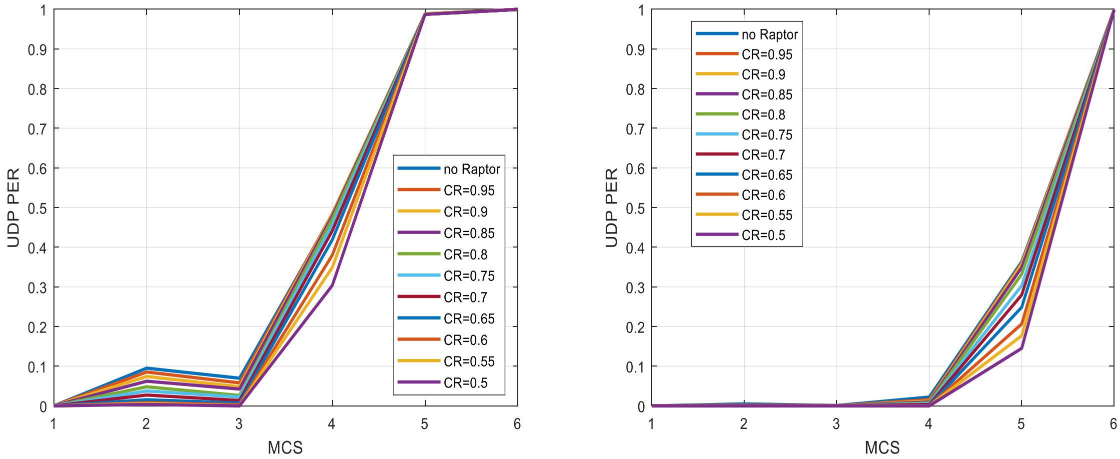

(a) (b)

Figure 2. UDP PER versus SINR for MCS 1: (a) SINR=8 dB and (b) SINR=12 dB.

To this end, Figure 3 shows the required to total RBPs (N_{TRBP}) for a source block depending on MCS modes and RQ code rates. As expected, the required to total RBPs (N_{TRBP}) increases when lower MCS modes and RQ code rate are used (low RQ code rate means more repair symbols are transmitted). Table 3 shows the optimum MCS and *CR* pair computed depending on the SINR values. It is observed that increasing SINR allows higher MCS to be selected for transmission therefore the required total RBPs significantly reduces. Note that MCS 5 is never selected because MCS 4 and 5 use the same amount of resources as seen in Figure 3. However, MCS 4 provides better PER performance than MCS 5. Therefore, it is better to select MCS 4.

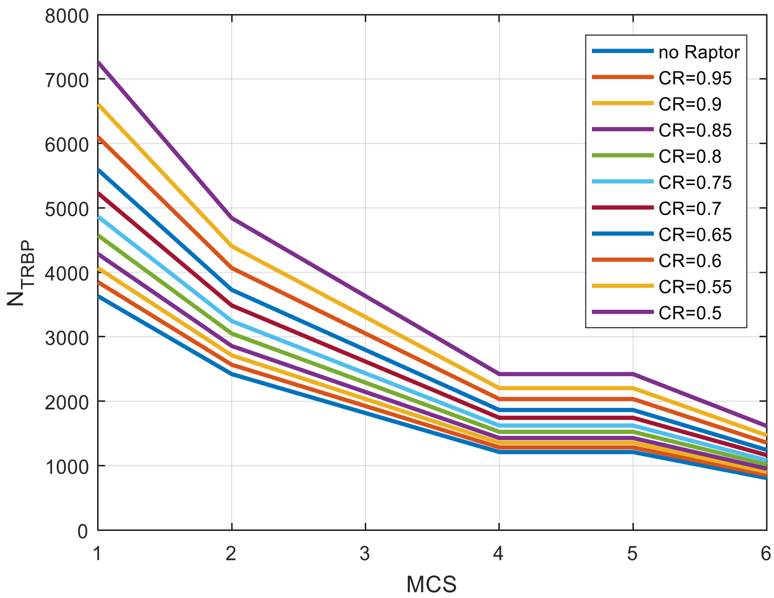

Figure 3. Required total RBPs (N_{TRBP}) depending on MCS modes and RQ code rates.

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Table 3. Optimum MCS and *CR* pair depending upon the SINR.

SINR (dB)	MCS Index (*m*)	RQ code rate (*CR*)	Required RBP N_{TRBP}
3	-	-	-
4	1	0.55	6609
5	1	0.65	5592
6	1	0.8	4575
7	1	0.95	3849
8	3	0.5	3631
9	3	0.65	2796
10	3	0.85	2143
11	4	0.65	1864
12	4	0.75	1622
13	4	0.95	1283
14	4	1	1211
15	4	1	1211
16	4	1	1211
17	6	0.7	1162
18	6	75	1082
19	6	0.8	1017
20	6	1	807

Furthermore, implementing RQ codes allows higher MCS to be selected for transmission even at very challenging channel conditions (lower SINR values) as shown in Figure 4 which compared the required total RBPs (N_{TRBP}) depending on SINR for the legacy and proposed systems. It is seen that without using RQ the system cannot provide any services until the SINR of 8 dB. However, with the use of RQ codes, the service can be provided for SINR ≥ 4 dB. In additions, the proposed cross-layer system provides more spectrally efficient transmission then legacy system for multicast transmission, i.e., up to 48,7% improvement in terms of using less RBP.

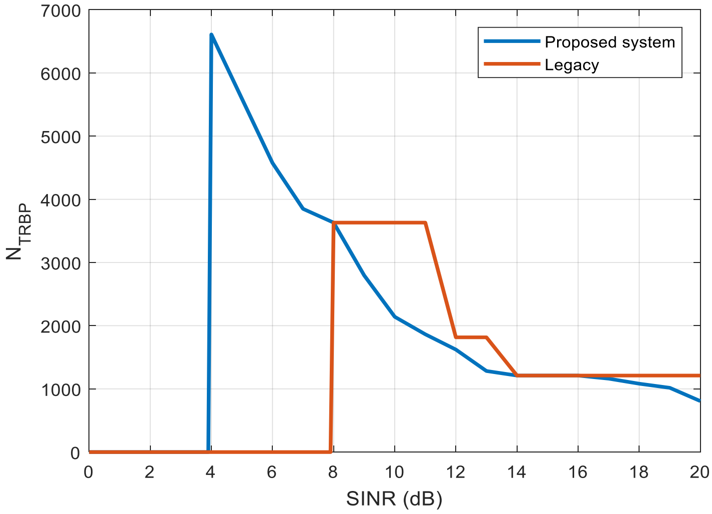

Figure 4. Required total RBPs (N_{TRBP}) versus SINR for the legacy and proposed systems.

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VI. CONCLUSION

This paper proposed a cross-layer optimisation algorithm which aims at providing scalable and reliable multicast video streaming services over the LTE-A SC-PMT system by jointly selecting MCS mode and RQ code rate for a given SINR under the QoS constraints of the applications. A cross-layer simulator was developed to explore optimum systems parameters (cross-layer trade-offs) under different channel conditions and evaluate the performance in an NLOS environment for a vehicular user. The results showed that the proposed system can provide services under harsh channel conditions (low SINR values) compared to the legacy system and 4 dB improvement in SINR. Further that it provides up to 48,7% more spectrally efficient multicast transmission than the legacy system. Overall, it was shown that in order to provide successful multicast transmission using SC-PTM scheme in LTE-A, AL-FEC codes and hence a cross-layer optimisation framework are needed to meet the stringent QoS requirements of applications using limited network and radio resources.

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